1994 PROCEEDINGS

48TH ANNUAL BROADCASTING ENGINEERING CONFERENCE PROCEEDINGS

NAB BROADCASTERS

Bob Moore
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1994 PROCEEDINGS

48th Annual
Broadcast Engineering
Conference Proceedings

Las Vegas, Nevada
March 20-24, 1994

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March 1994

Dear Industry Engineer:

On behalf of NAB's Department of Science & Technology, I am pleased to present the *1994 NAB Engineering Conference Proceedings*.

The 48th Broadcast Engineering Conference features 31 sessions with more than 163 useful and informative presentations, many of which are contained in this publication. The starting point for this conference is a crucial keynote address by Jules Cohen, P.E., contemplating the future of broadcasting and the challenges facing our industry.

The Conference continues with a broad spectrum of technology and policy issues. These include the first reports on industry testing of in-band AM and FM DAB, radio and television data broadcasting technologies, and full coverage of HDTV transmission issues. We are particularly pleased that these *Proceedings* include a special section featuring the draft, *Grand Alliance HDTV System Specification*, submitted to the Advisory Committee on Advanced Television Systems (ACATS) last month.

For the third year, the Society of Broadcast Engineers has programmed a full day of sessions for the Broadcast Engineering Conference. In addition, two other organizations, the Society of Motion Picture and Television Engineers and the IEEE Broadcast Technology Society, have joined us to present two special programs, the SMPTE Post Experience and the Digital Transmission Tutorial. We hope that you had an opportunity to attend these programs.

The best engineers in the world have worked to create the most dynamic and informative broadcast engineering conference in the world. NAB sincerely hopes that you have found the 48th Broadcast Engineering Conference to be educational and enjoyable. I encourage your comments.

Best regards,

Michael Rau

Michael C. Rau

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Sunday, March 20, 1994

Moderator:
Charles Dages, CBS Television Network, New York, NY

THE 1994 BROADCAST ENGINEERING CONFERENCE OPENING CEREMONY

KEYNOTE ADDRESS,
BROADCASTING: PAST, PRESENT AND ?
Jules Cohen
Jules Cohen and Associates, P.C.
Washington, DC

BROADCASTING: PAST, PRESENT AND ?

Jules Cohen
Jules Cohen and Associates, P.C.
Washington, District of Columbia

I should like to extend my thanks to the NAB, and especially Michael Rau, for granting me this opportunity to express a few thoughts relating to the past, present and future of broadcasting. The title I have assigned to these remarks was engendered by the fuzziness of the image in my crystal ball. We hear from many prognosticators who appear quite positive about their vision of the future; however, my ego is not so great that I can state, categorically, how our world of broadcasting will entwine with the multimedia world of the future.

My earliest memory of broadcasting occurred in about 1924, near the infancy of broadcasting, when my father brought to our home an Atwater Kent radio receiver. Without any knowledge of how radio worked, it seemed quite magical.

I know now that the receiver had three, separately tuned, stages of RF. In these days of simple tuning, particularly digital tuning, the primitive process used then is now hard to envision. Finding a station was a real adventure. The calibrations on the three capacitor dials tracked approximately, but not exactly. A great many back and forth adjustments were necessary to achieve satisfactory reception. Once the settings for a station were puzzled out, they had to be logged to make possible a relatively easy return to the station desired.

Unfortunately, on that first night, we could not enjoy the wonders of radio. Dad had brought along a lead acid storage battery for the filament supply, but Mother would not allow a lead acid storage battery to sit in her living room. So we had to wait until the next evening when Dad brought home a "battery eliminator" --- a mercury arc rectifier that gave off a fearsome glow and buzz. In any event, that radio receiver with its horn loudspeaker and multi-turn loop antenna brought to us a wide world of music and dramatic performances. Since I am using no visuals, you can pretend that you are sharing our living room and providing images limited only by your imaginations. Perhaps we lost something when radio dramas disappeared as a major contributor to our leisure activity. With the possible exception of science fiction offerings, most of us can create mental images from an ingeniously wrought aural program, with sound effects, that can rival the best of the settings devised by the experts.

Although ganged capacitors with trimming devices soon eliminated the need for multiple dials, Major Edwin Armstrong's invention of the superheterodyne receiver, permitting the principal amplification of the signal to be achieved in a fixed-tuned intermediate-frequency (IF) stage, greatly improved our access to radio programming.

Single knob tuning, calibrated by frequency, became more reliable. Elimination of the arbitrary logging scale was a great boon. That, and other innovations, such as built in rectifiers, where DC was a must, as for the vacuum tube plate supply, and speakers incorporated in the same cabinet as the receiver, encouraged the growth of radio receiver distribution to virtually every household.

Some of you may be surprised to hear that I saw my first television presentation (it could hardly be described as a program) in 1928 or '29. An appliance store in our Chicago neighborhood put on the demonstration. I assume that what was used to scan the image was a Nipkow disk. I don't think that anything more sophisticated was available at that time. As I recall, we looked through a peephole into this darkened box and actually saw moving images. The images were not well defined, but they were recognizable and they moved.

Except for a few experimental operations, television remained relatively dormant until the end of World War II. AM broadcasting dominated the scene and was profitable despite the ongoing Great Depression. It was an era when broadcast operations could not only build magnificent facilities by our present standards for aural broadcasting, but could afford to have some of its engineers, such as Jim Rockwell at WLW and Jack DeWitt at WSM, conduct experiments on radio propagation. They also developed new designs for transmitters and other equipment. George Brown and other industry engineers, particularly at RCA, wrote seminal articles that appeared in the RCA Review and in the Proceedings of the Institute of Radio Engineers, now incorporated in the American Institute of Electrical and Electronic Engineers after merging with the American Institute of Electrical Engineers.

World War II brought a burst of invention in the field of electronics that served us well, particularly in the field of television that began in earnest after the War. The slot antenna was developed by Henry Riblett and his group at the MIT Radiation Laboratory. The cathode ray oscilloscope was improved from a rather primitive device to something far more sophisticated, leading to better display devices and test equipment. High power UHF and microwave generators were improved or invented. The list goes on.

Television broadcasting started slowly in the late 1940s and expansion stopped during the 1948 to 1952 FCC "freeze" on requests for permission to build new television facilities. A stop to processing applications for television facilities was deemed necessary to permit the Commission to take a fresh look at its rules for the service. New rules were adopted, and applications were again accepted starting July 1, 1952.

Television expanded rapidly once the FCC resumed processing applications for construction permits. New stations were built, and the few stations that had begun before 1948 improved facilities. However, AM broadcasting continued to rule the roost. Many a television station was built and maintained during its break-in period with profits from associated AM stations. During that same period, FM, too, was a weak sister of AM. The major manufacturers of broadcast systems preferred to concentrate on the more profitable television equipment than on FM.

I need not remind this audience of where we stand today, with FM dominating the world of aural broadcasting and television broadcasting competing with myriad alternate ways of delivering moving images to the home.

Now we stand on, or perhaps we can say that we have just crossed, the threshold of a new world of digital electronics. The use of digital techniques has been with us for some time in the production of both sound and images, but our systems have been basically analog with entry and exit from digital devices. Digital audio broadcasting (DAB), or as some would have it "digital audio radio" (DAR), will provide digital paths from microphone to loudspeaker. The advent of digital television is no more than two or three years away. (However, our ears and eyes will remain analog devices.)

What about the multimedia world that we hear so much of, and which will be strongly in evidence at this 1994 NAB Convention? My personal belief is that considerable interchange of broadcast aural and visual material with what we now identify as computer-oriented material will occur, but my imagination will not accept a world where a major segment of the public spends its lives before a single display device used for entertainment, news, shopping, banking, investing, game playing, and all that has been suggested. A world without the personal contacts of the marketplace would be a cold world indeed.

I do believe that digital broadcasting will provide the opportunity for new services involving an economical means of delivering data wherever point-to-multipoint service is desired. Such services are likely to be the additional revenue streams much sought by broadcasters beset by competition from cable, both wired and wireless, satellite, telephone and recordings.

So what is the likely future of broadcasting, and what about the role of the engineers? As to both, I am upbeat. So long as broadcasters maintain their programming skills and remember their obligations to be outlets for local needs, the public will continue to demand the universal availability of programming without a price tag. From the poorest to richest homes, all have access to at least the offerings of the broadcast networks, public broadcasting and the independents. These sources, in sum, open a window to news, drama and most sporting events for everyone. The social and political implications of the disappearance of free, over-the-air local television are serious. Our republic cannot tolerate the creation of a society of "haves" and "have nots" with respect to what is available now to all.

I do not believe that even those predicting a wired nation and the death of broadcast television suggest that aural broadcasting will disappear. No wired system can match the ubiquity of radio. Interestingly, the evolutionary processes in the delivery of both voice and data are moving us from wire to wireless.

Engineers are used to change. In fact, they are the principal drivers of change. We went from simple, single tower AM broadcast facilities to elaborate antennas with as many as twelve towers to provide new services in an increasingly crowded spectrum. The solid state revolution required the learning of an entirely new technology for those who had learned their profession in the days of the vacuum tube. We learned to use the computer, not only for a more efficient way to perform engineering calculations, but also as a control device. Engineers were the inventors of advanced technology. Additionally, they were charged with the means for implementing that technology. So long as we are willing to adjust to the demands of change, we shall continue our useful role. If we resist change, we shall be crushed by the juggernaut of new technology.

While recognizing the inevitability of change and our need to adapt, we should not forget that our primary role is the improvement of broadcasting and the expansion of its availability. We must maintain a sensitivity to the desirability of enlarging the role that broadcasters can play through the provision of ancillary services, but broadcasting remains the core of our existence. No enlargement of our capability to expand the service offerings of the broadcast community should be allowed to conflict with our providing a continually improved broadcast service. I hope that this association continues to be the National Association of Broadcasters, and does not become the National Association of Multimedia.

THE "GRAND ALLIANCE" HDTV SYSTEM

Sunday, March 20, 1994

Moderator:
Jae Lim, MIT, Cambridge, MA

***SYSTEM OVERVIEW**
Woo Paik
General Instrument
San Diego, CA

Wayne Luplow
Zenith Electronics
Glenview, IL

***VIDEO FORMATS**
Robert Keeler
AT&T Bell Laboratories
Holmdel, NJ

***VIDEO COMPRESSION**
Carlo Basile
N.A. Philips Laboratories
Briarcliff Manor, NY

***TRANSPORT/INTEROPERABILITY**
Glenn Reitmeier
David Sarnoff Research Center
Princeton, NJ

***TRANSMISSION**
Woo Paik
General Instrument
Glenview, IL

Wayne Luplow
Zenith Electronics
Glenview, IL

***AUDIO**
Craig Todd
Dolby Laboratories
San Francisco, CA

*See Special Report: "Grand Alliance" HDTV System Specification draft document at the end of these Proceedings, pg. 433.

DATA BROADCASTING: RADIO

Sunday, March 20, 1994

Moderator:
James Ary, Great American Communications, WTVN/WLVQ, Columbus, OH

***PROFITABLE IVHS SUBCARRIER OPPORTUNITIES**
David Kelley
Terrapin Corporation
Garden Grove, CA

HIGH SPEED SUBCARRIER DATA SYSTEM (HSDS)
Gary Gaskill
Seiko Telecommunication Systems, Inc.
Beaverton, OR

THE DEVELOPMENT AND STRATEGY OF AN FM MULTIPLEX DATA BROADCASTING SYSTEM FOR MOBILE RECEPTION
Tetsuro Miyazaki
NHK
Tokyo, Japan

***DIFFERENTIAL GPS USING FM-SUBCARRIERS: THE ACC-Q-POINT SYSTEM**
Paul Galyean
Magnavox Electronics Systems Company
Torrance, CA

*Papers not available at the time of publication

HIGH SPEED SUBCARRIER DATA SYSTEM (HSDS)

Gary Gaskill and Ken Gray
SEIKO Telecommunication Systems, Inc.
Beaverton, Oregon

Abstract

This paper summarizes some of the key features of a High Speed FM Subcarrier Data System (HSDS); with emphasis on the subcarrier data channel. The modulation and demodulation techniques used by HSDS are described. HSDS receiver sensitivity and its relationship to coverage are discussed, along with compatibility of the subcarrier with the main channel audio and RDS. The packet and frame structures and multi-station methods are defined and the reliability techniques are explained. In addition, general specifications the modulator and available receiver and demodulator integrated circuits are described. Results from lab and field tests are provided including error correcting effectiveness. Results of the Portland, Oregon coverage tests are shown. Finally, the services currently provided in the Northwest USA are described.

INTRODUCTION

Wireless personal communication is becoming an increasingly important aspect of the communications industry. One facet of this rapidly evolving technology is the development of one-way systems providing messaging and information services including: personal messaging, traffic information, time of day, news, sports, weather, business and emergency information.

SEIKO Telecommunication Systems Inc. (STS) has developed such a system. It is a flexible High Speed FM Broadcasting Subcarrier Data System (known as HSDS) with capability for world wide operation. HSDS is fully developed and has been deployed by STS under the RECEPTOR trademark in a number of cities around the world. Its commercial introduction was in Portland, Oregon in July of 1990. Seattle - Tacoma, Washington began commercial operation in October of 1992. Currently marketed applications include paging and information services to numeric display wrist-watch receivers (SEIKO® Receptor™ MessageWatches™) and pocket pagers.

Two highly integrated circuits -- a receiver and demodulator, operating with 3 volt lithium cells, are available for small, inexpensive low power designs. These ICs are the basis for the wrist-watch receiver. Queuer/encoders and modulators are also available for the system. Utilization of existing FM radio broadcast infrastructure, currently available integrated circuits and transmission equipment make this system very inexpensive to deploy.

SUMMARY OF HSDS SYSTEM

The HSDS protocol is a very flexible, one-way, communications protocol. The system design permits very small receivers. Receivers, with duty cycles varying from continuously on to duty cycles of less than 0.01%, provide flexibility to select message delay, data throughput and battery life. HSDS can operate as a stand alone single station (channel) system, or as multiple systems operating independently in a geographical area with each system including multiple stations. Multiple stations are accommodated by frequency agile receivers, time offset message transmission on each station, and transmitted lists of stations surrounding each station. Reliability can be enhanced via multiple transmissions of messages. Duplicate messages are rejected via comparison with the transmitted message number.

The system is time division multiplexed. Time is subdivided into a system of master frames, subframes and time slots. Each slot contains a packet of information. In multiple station systems, each station's transmissions are synchronized to UTC ensuring synchronization between stations. Each receiver is assigned a subset of slots as times for monitoring transmissions. Multiple receivers may share time slots, due to the random nature of expected communications. Each slot is numbered and each packet contains the slot number that permits rapid location of assigned time slots.

The error correction scheme is very flexible. The methods used vary with the applications. The method used in the watches is designed to correct a short burst of errors associated with random noise or automotive ignition noise. The standard CCITT 16 bit CRC is typically added as a component of each packet and minimizes the chance of false messages.

HSDS modulation and encoding provide a high data rate, a narrow bandwidth with high spectral efficiency and negligible impact on the main channel. HSDS modulation is AM-PSK with duobinary encoding. The HSDS data rate is 19,000 bits per second in a bandwidth of 19khz, centered at 66.5khz. The HSDS signal is modulated as a subcarrier ranging from 5% to 20% injection but typically at 10% on a commercial FM radio station's carriers in the frequency range of 87.5 to 108Mhz. Sharp transmission filter skirts cause extremely little impact on the main channel in no multipath situations; and generation of a pseudo-randomized data stream reduces impact on the audio even in multipath situations. The narrow bandwidth of HSDS allows for compatibility with RDS operation world wide. HSDS allows for use of subcarriers above 76khz in the USA and compatibility with European spectrum allocation.

PHYSICAL LAYER

This section describes the characteristics of the subcarrier data channel for the HSDS messaging system. Included in this discussion are the modulation and demodulation techniques used to achieve communications through the subcarrier channel at an acceptable bit-error-rate (BER), and the performance of the subcarrier channel in the FM broadcast channel. The ultimate performance or reliability of the HSDS messaging system is determined not only by the physical layer but also by the characteristics of the messaging protocol described in later sections. Additional considerations of the physical layer included in this section are the issues of subcarrier interference with the main channel audio, and the compatibility of the HSDS subcarrier signal with an RDS signal.

Modulation and Demodulation

The HSDS modulation / demodulation scheme was chosen to satisfy the following criteria:

- Non interference with commercial FM radio receivers.
- Compatibility with world wide broadcast FM regulations.
- Simplicity in IC implementation of the demodulator.
- Low-cost mobile receiver with a small form factor.
- Adequate BER performance in the presence of noise.
- Commercially satisfactory coverage area.
- High data rate.

Modulation

The HSDS subcarrier channel is centered at 66.5kHz. The choice of 66.5kHz as a subcarrier frequency is compatible with international (CCIR) standards of 53 to 75kHz and the US standard of 53 to 99kHz. The 66.5kHz center frequency was also chosen because it is 3.5 times the 19 kHz pilot which allows phase locking to the pilot. Additional advantages include: reduced receiver parabolic noise in relation to higher frequency subcarriers and better audio performance than other subcarrier frequencies because 67kHz subcarriers have been common for many years and receivers typically have 67kHz reject filters.

For ease of receiver implementation the subcarrier frequency and transmitted data are phase-locked to the main channel 19kHz stereo pilot. The subcarrier data to pilot phase relationship is approximately 63 degrees. The raw data rate of the system is 19kbps. The 66.5kHz subcarrier is a double-sideband suppressed-carrier amplitude modulated signal using the duobinary technique described by Lender (1963). The duobinary technique uses controlled intersymbol interference to achieve 1bits/sec/Hz efficiency with realizable filters while 2-level DSB PAM system cannot achieve this efficiency. The duobinary encoding technique achieves this result by using a filter to create intersymbol interference that combines the current and previous data bit, creating a three level output signal in the demodulator. Normally intersymbol interference degrades the performance of a demodulator; but for duobinary, the use of a cosine filter allows the receiver to decode the current bit in the presence of the intersymbol interference using a simple symbol by symbol decoding scheme. In addition to the improved spectral efficiency obtained by the duobinary modulation technique, the cosine filter required for optimum performance is simple to implement.

The transfer function $H(\omega)$ of the duobinary filter is given by

$H(\omega) = \cos(\omega T/2), \ |\omega| \le \pi/T$
= 0, elsewhere, where T is the symbol period = 1/19kHz.

In the usual situation where the channel symbol power is limited, the optimum filtering required to maximize the signal-noise ratio at the receiver would involve equally splitting the filtering between the transmitter and the receiver as $\sqrt{H(\omega)}$. However, the FM subcarrier situation is different in that the limiting factor for the channel is not symbol power but the frequency deviation of the carrier caused by the modulation. This frequency deviation constraint implies a voltage-limiting constraint for this modulation rather than a power-limiting constraint. As a result the usual matched-filter theorem for maximizing signal-noise ratio does not apply. An improved filtering split between transmitter and receiver when a voltage constraint is applied has been found through simulation to be:

$^{3/4}\sqrt[4]{H(\omega)}$ at the transmitter, and

$^{1/4}\sqrt[4]{H(\omega)}$ at the receiver

Demodulation
Synchronous demodulation with sub-Nyquist sampling is achieved by phase locking the sampling clock to the 19kHz pilot. The receiver filter is implemented as an FIR with symmetrical coefficients. The time domain waveform of the modulated subcarrier is shown in Figure 1.

HSDS Subcarrier Receiver Performance

It is of particular interest to determine the signal-to-noise ratio in the subcarrier channel SNR_{sc} , since this determines BER performance for a given modulation technique. For an FM receiver system in the absence of multipath effects, SNR_{sc} depends primarily on the following parameters:

- Carrier power C in dBm in the main FM channel receiver IF.
- Percentage of power available to the subcarrier, typically specified as an injection percentage.
- Effective subcarrier channel bandwidth B_{sc} in Hz.
- Receiver noise figure NF_{rcvr}.

The relationship that contains all the above factors then becomes a key system design equation for coverage tradeoff studies. Assuming narrowband FM and an ideal FM demodulator, the general relationship for the SNR in any subcarrier channel is given by:

$$SNR_{sc}=C*\beta^2/(F_{rcvr}kTB_{sc}),$$

where

β = ratio of peak frequency deviation of the subcarrier to the subcarrier modulating frequency

$k = 1.38*10^{-23}$Joules/K Boltzmann's constant,

$T = 290^{\circ}$K Thermal noise temperature.

Expressed in logarithmic units the result becomes:

$SNR_{sc}=C+20*Log(\beta)-10*Log(kT)-10*Log(B_{sc})-10*Log(F_{rcvr})$

As an example of the use of the above formula we will consider the HSDS case:

$$kT=-174dBm/Hz$$

NF_{rcvr}=5dB (worst case for current receiver IC)

B_{sc}=25kHz (10Log(B_{sc})=44dB)

SNR_{sc}=12dB,

20Log(β)=-19dB (10% injection),

where an SNR_{sc} of 12dB is assumed to be the minimum required for an acceptable BER performance, and B_{sc}, the noise bandwidth at RF is twice the effective receive filter bandwidth at baseband.

Under the above conditions with a subcarrier injection of 10%, the weakest signal at the receiver input that can be demodulated to give an adequate BER performance can be calculated as receiver sensitivity:

$$\begin{aligned} C_{min} &= SNR_{sc}-20*Log(\beta)+10*Log(kT) \\ &\quad +10*Log(B_{sc})+NF_{rcvr} \\ &= 12+19-174+44+5 \\ &= -94 \text{ dBm} \end{aligned}$$

The above result can now be used to consider coverage. To this end a relationship is needed connecting receiver sensitivity with an electromagnetic field strength value. This is obtained using the standard free-space half-wave

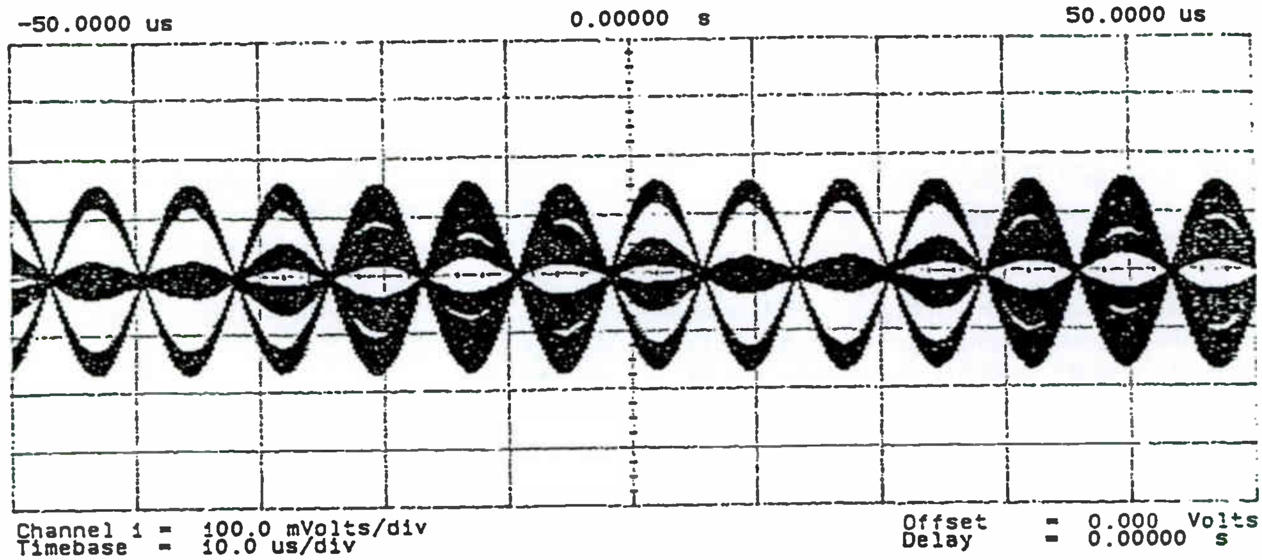

Figure 1

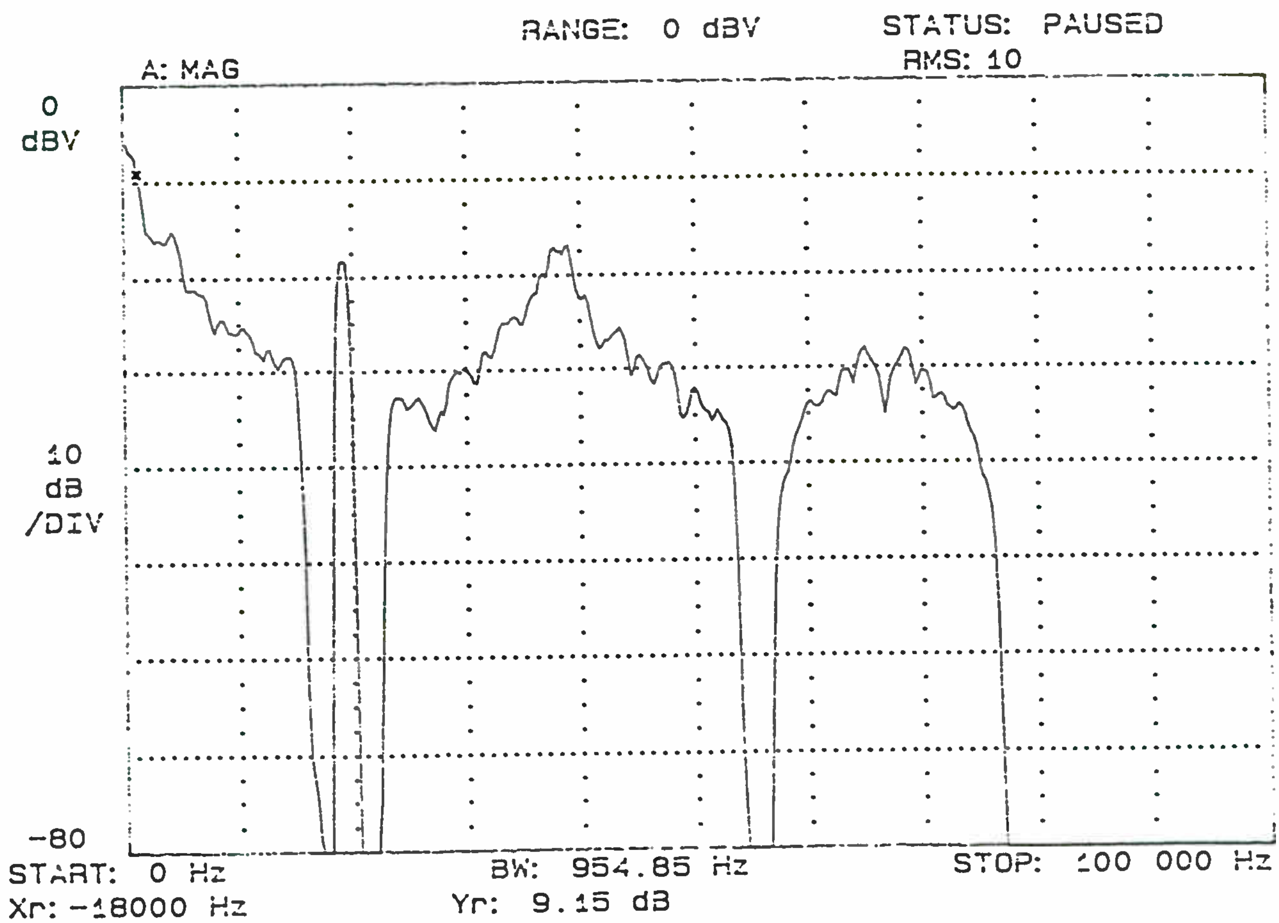

Figure 2

dipole equation relating field strength E (volts/m) to power P_{dipole} (watts) developed at the dipole terminals.

$$P_{dipole}=(E^2/120\pi)*(\lambda^2/4\pi)*1.64$$

λ=wavelength of the radiation and is assumed = 3 meters (f = 100Mhz) for illustration purposes.

For a given antenna gain, relative to a dipole, the receiver sensitivity computed above can be related to the required field strength E. Two cases are considered:

1. An automobile whip antenna with a gain of 0 dBd with an appropriate receiver results in a field strength requirement of 23.5 dBuV/m.

2. A wrist sized loop antenna with a gain of -29dBd and the current HSDS MessageWatch™ module results in a sensitivity of 52.5 dBuV/m.

The nominal sensitivity of the HSDS MessageWatch™ receiver and a HSDS automobile receiver sensitivity as computed above can then be related back to the standard coverage charts for any given FM station.

FM Broadcast Environment

The FM radio base band signal is discussed in this section. HSDS is described and shown with a typical stereo station's base band signal. Compatibility of HSDS with the audio and RDS is discussed.

HSDS in the FM Broadcast Channel

The HSDS subcarrier is summed onto the FM station's baseband signal before being FM modulated onto the RF carrier. The spectrum of an audio stereo baseband signal with HSDS signal is pictured in Figure 2.

Compatibility with main channel audio

In subcarrier applications it is critical that the subcarrier not interfere with the audio channel in a way that would affect a listener's perception of audio quality. There are two principal considerations in this regard: the first relates to the transmission filter and the out of band filter attenuation. HSDS filtering at the transmitter is implemented digitally with a finite impulse response filter. The subcarrier is more than 60dB below the pilot at the band edges.

The second interference source is potential nonlinear mixing of the subcarrier with the audio due to multipath. As a countermeasure to interference from multipath, the HSDS protocol (described in later sections) uses data randomization to 'whiten' the signal and in particular avoid the generation of tones in the audio portion of the band. Extensive field and laboratory testing, along with significant operational experience in Seattle and Portland have demonstrated HSDS has no noticeable interference with the main channel audio.

Compatibility with the RDS subcarrier

RDS is a subcarrier standard first available in Europe and now gaining acceptance in the USA. The standard occupies the band of 54.6kHz to 59.4kHz and provides a data rate of 1187.5 bps. Lab tests were performed to evaluate the effects of the HSDS signal on RDS reception. A highly compressed audio was used for these tests. Seven commercial RDS automobile radios were tested. RDS receiver sensitivity was defined as .01 BER using 2.66% injection. Results from a representative receiver are summarized in Table 1.

RDS receiver sensitivity	No Audio	With L-R Audio
No HSDS	-93.5 dBm	-92.5 dBm
With HSDS	-93.0 dBm	-91.75 dBm

Table 1

As can be seen from the table, impact of HSDS signal is less than .75 dB with or without audio. The impact of L-R audio on RDS sensitivity is 1 dB or more. Considering L-R audio has a greater impact on RDS than HSDS, the impact of HSDS on RDS is entirely acceptable.

Link Layer

This section describes the link layer. The layer includes features required to make a reliable single station data link. The link layer includes the frame structure and packet structure (size, word synchronization, error detection and correction).

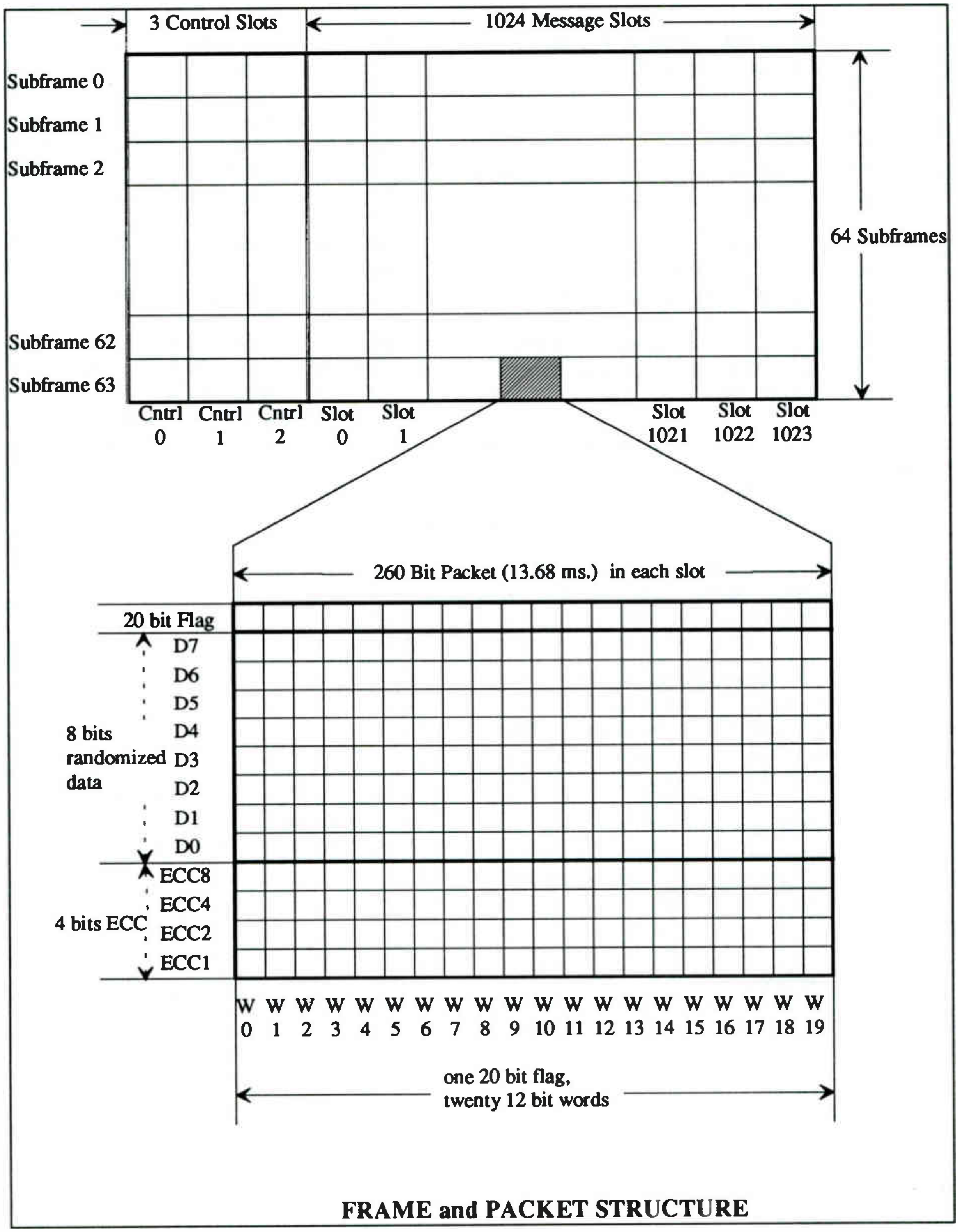
3 Control Slots
1024 Message Slots
Subframe 0
Subframe 1
Subframe 2
Subframe 62
Subframe 63
64 Subframes
Cntrl 0
Cntrl 1
Cntrl 2
Slot 0
Slot 1
Slot 1021
Slot 1022
Slot 1023
260 Bit Packet (13.68 ms.) in each slot
20 bit Flag
D7
D6
D5
D4
D3
D2
D1
D0
8 bits randomized data
ECC8
ECC4
ECC2
ECC1
4 bits ECC
W 0 W 1 W 2 W 3 W 4 W 5 W 6 W 7 W 8 W 9 W 10 W 11 W 12 W 13 W 14 W 15 W 16 W 17 W 18 W 19
one 20 bit flag, twenty 12 bit words
FRAME and PACKET STRUCTURE

Figure 3

Packet Structure

HSDS Data is transmitted in fixed length packets. The bottom part of Figure 3 illustrates the packet structure used in the HSDS Protocol. Each packet is 260 bits long. Packet format bits in each packet control the packet's structure. A typical packet format consists of a word synchronization flag, error correction code (ECC), information bits and error detection code (CRC). The following description assumes the packet format used in currently operating HSDS systems.

The information bits from higher layers consist of 18 octets (8 bits per octet) per packet. A two octet CCITT standard 16 bit CRC is generated from the 18 octets and appended thus creating a 20 octet link data unit. The first octet (the incrementing slot number) of the 20 octet data unit is XOR'd with each of the following 19 octets that results in pseudo-randomized data. (In the event of multipath or other distortions to the signal during reception data randomization tend to 'whiten' the sounds in the audio channel.) Appended to each octet of randomized data are 4 bits of Hamming ECC, the resulting 20 - twelve bit words are referred to as ECC'd data. This error correction method provides single bit error detection and correction in 12 bits, or 8.3% correction capability. This method is quite easy to decode and reasonably efficient.

To increase burst error correction capability, the ECC'd data is interleaved on transmission providing immunity to 20 bit error bursts. Word synchronization is established by a 20 bit flag sequence that is added at the beginning of the CRC'd, randomized, ECC'd, interleaved data. Table 2 shows the steps in the link layer transmitter encoder and the reverse steps performed by the receiver decoder.

A double error correction on a stream of packets is being planned for applications with less severe power constraints, and requirements for higher data reliability.

Step	Transmit Encoder	Receive Decoder
1	Compute and add CRC	Find flag
2	Randomize data	Deinterleave data
3	Add error correction	Apply error correction
4	Interleave data	Derandomize data
5	Add flag	Compute and compare CRC

Packet Structure Encode and Decode steps
Table 2

Frame Structure

The HSDS Protocol is based on a packet oriented time division multiplexed (TDM) scheme. The top section of Figure 2 illustrates the frame structure used in the HSDS Protocol. The largest structure used by the protocol is a master frame. Each master frame contains 64 subframes. Each subframe is divided into 1027 units called time slots. Each slot contains a packet. The first three times slots in each subframe are control slots, and the remaining 1024 are regular message time slots. Control slot packets include time of day & date and lists of surrounding stations using the HSDS Protocol. Regular message time slot packets typically include a slot number, receiver address, data format, packet format and the message data.

The 19khz stereo pilot signal is used as the clock for the data. Since the stereo pilot may not be exactly 19khz, at the time of transmission, a single bit may be added (pad bit) between packets as required to maintain exactly 19,000 bits per second data rate. This occasional addition of a pad bit maintains proper synchronization with UTC.

Network Layer

This section describes the network layer. The network layer includes features required to make a number of individual stations act as a single system. The network layer includes receiver addresses, multiple application multiplexing, multiple station concepts such as local station lists, station time offsets and synchronization between stations via UTC, and network reliability concepts.

Multiple Station Systems

When multiple station systems are required, master frames are synchronized to UTC and begin at the start of each quarter hour (plus an individual transmitter's time offset). The synchronized and time offset stations provide an opportunity to change the tuned frequency and make subsequent attempts on other frequencies to receive packets.

System Reliability

Robust wireless systems require methods to address multipath and shadowing issues. The effects of multipath, which are not included in the analysis of receiver performance earlier, play a significant role in determining system performance. Multipath can be viewed as a time-varying nonlinearity that can distort or reduce the received signal to a point that reliable reception is no longer possible. Shadows behind hills and mountains or due to man made obstacles to radio frequencies can reduce signal strengths below sensitivity levels.

Some systems attempt to address these issues with extensive error correction schemes. While these techniques are useful for the moving receiver, they become ineffective when the receiver is stopped in an extremely low signal strength area or moving very slowly through multipath nulls. A car stopped at a traffic light or a person at a desk wearing a receiver are two examples of the breakdown of even the most robust error correction methods. Diversity techniques are frequently used to combat multipath effects. HSDS addresses multipath and barrier issues with a combination of frequency, space and time diversity and message numbering.

Diversity Techniques possible with HSDS

Frequency diversity can be achieved through the capability to tune any frequency in the range from 87.5 to 108Mhz. By transmitting on multiple frequencies a receiver in a multipath null at one frequency is not likely to be in a multipath null on another frequency. In the event a receiver is behind a hill or other such obstacle to reception, space diversity can be used as required by the application.

Space diversity (transmitters at different locations) provides paths from two or more directions reducing the size of shadowed areas, and reduce the possibility of missed messages.

Time diversity can be provided in two ways -- multiple transmission on the same station and delayed transmission between stations. Multiple transmissions of information several minutes apart is utilized for wrist mobile applications where a receiver may be passing through a radio frequency shielded area, such as a tunnel or deep basement. The second method for time diversity includes a time offset for data transmission. This time offset between stations provides an opportunity to change the tuned frequency and make subsequent attempts to receive a packet for low data rate receivers. When much greater throughput is required, receivers operate on the best station selected from the stations available.

Message Numbers

In addition to diversity techniques, each transmission includes a transmitted message number that eliminates duplicate messages and permits detection of missed messages. When there are multiple transmissions of a message, it is possible to receive the same message several times. By rejecting duplicated message numbers, multiple transmissions do no appear as duplicate messages. A receiver can keep track of the received message numbers, and if there is a skip in the message number sequence, it can be concluded that a message was missed. The input system can log each message by message number and permit retrieval of missed messages by the owner of the receiver.

Implementation

Receiver Integrated Circuit

The receiver integrated circuit's primary function is an FM receiver that takes the RF signal from the antenna and generates the base band signal from 0 to 100khz. General Specifications for the IC are as follows:

- sensitivity < -91.5 dBm at 10% subcarrier injection
- 12 dB subcarrier to noise ratio output at sensitivity
- frequency range from 87.4 to 108Mhz
- power required 2.1 to 3.3 Volts, 18 mA maximum
- operating temperature 0 - 50° C
- number of external components - about 29
- number of adjustments required - none

Demodulator Integrated Circuit

The demodulator integrated circuit's primary function is to demodulate the base band signal into a 19 kbs synchronous binary data stream. Other functions include a synthesizer for the receiver IC (providing 100khz steps over the FM band) and loop antenna tuner control, and a 256 bit EE-PROM. General specifications for the IC are as follows:

- Demodulator
    - data filter - 32 point FIR square root cosine filter, 66.5khz center
    - variable level data detection from 5% to 20% injection levels
    - clock - 19khz digital phase locked loop
- power required - 2.1 to 3.3 Volts
  3.5 mA maximum - data demodulator
  1.5 mA maximum - synthesizer
- operating temperature 0 - 50° C
- number of external components -10
- number of adjustments required - none

Complete Receiver / Demodulator Module

The above integrated circuits have been combined into a complete receiver / demodulator module. This module, when attached to an external antenna, provides a 19,000 bit per second synchronous data to external decoder chips and processors. Its approximate size is 30 x 27 x 6 mm.

Subcarrier Generator

An HSDS subcarrier generator using DSP technology is available. Features include:

- Data is phase locked to 19khz pilot with adjustable phase angle.
- Full redundant operation including automatic switch over to secondary subcarrier generator with separate power supply.
- Out of band energy detection and auto shut down for fail-safe operation of audio.
- Internal watch dog timer for automatic reset
- Remote control operation

Protocol Queuer / Encoder

An HSDS Protocol queuer and encoder is also available.

Lab Tests

HSDS with a 10% injection and the described integrated circuits provide a receiver with a -91.5 dBm sensitivity, 12 dB subcarrier to noise ratio, $2x10^{-2}$ BER, a 0.55 packet completion rate (PCR) and, with 3 transmissions on a single station, a 0.91 message completion rate (MCR). This is illustrated graphically with Figure 4, which shows the various components and their sensitivity specifications

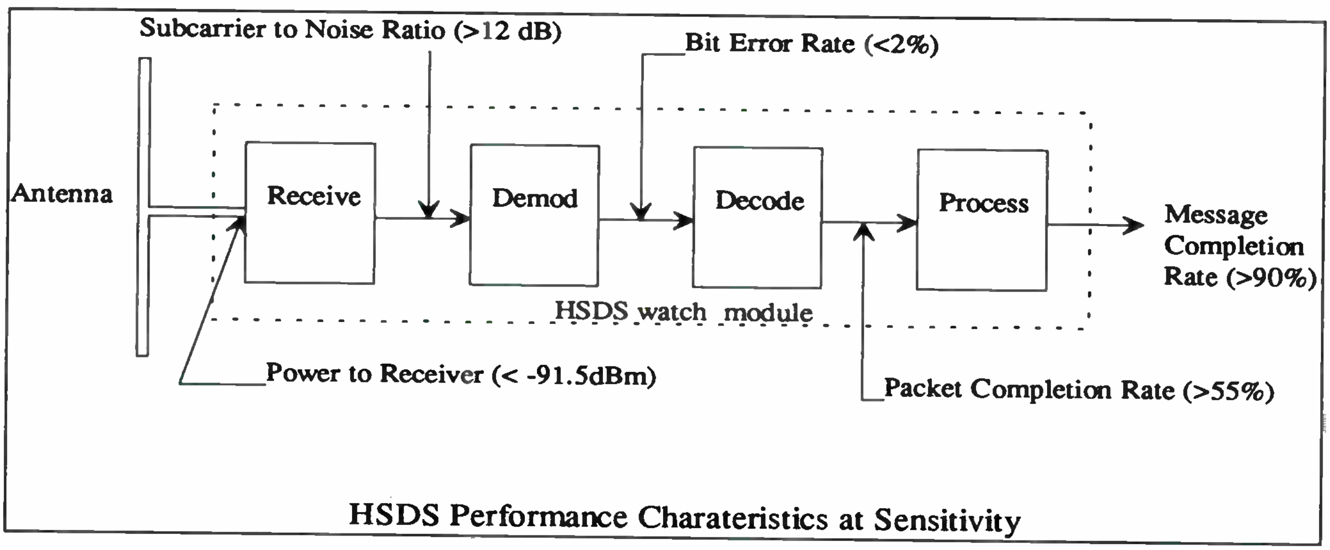

Figure 4

A plot of a current MessageWatch™ module's performance PCR versus input power in the lab is shown in Figure 5.

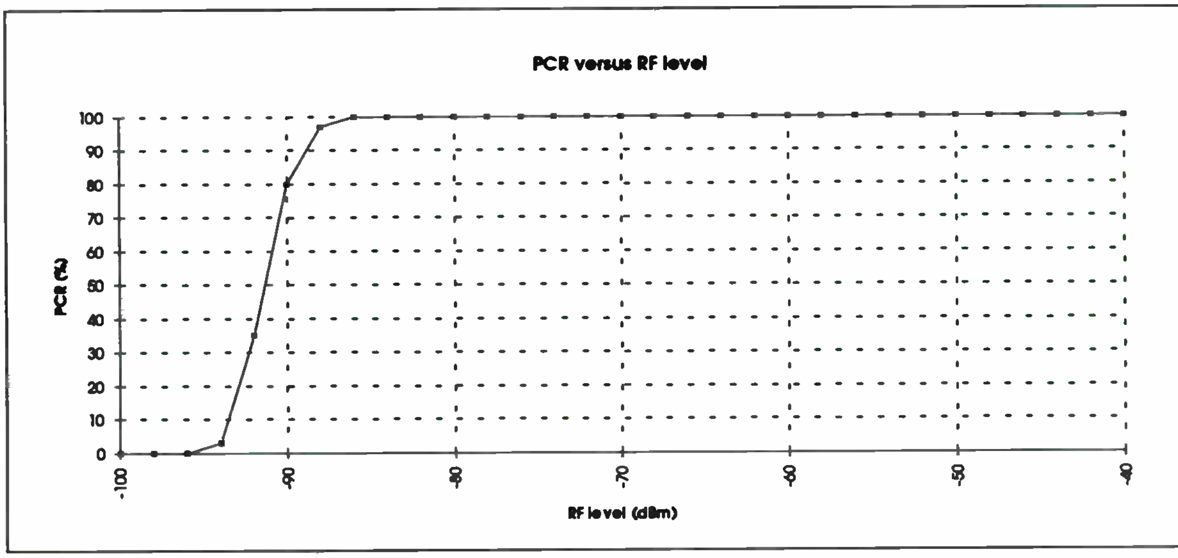

Figure 5

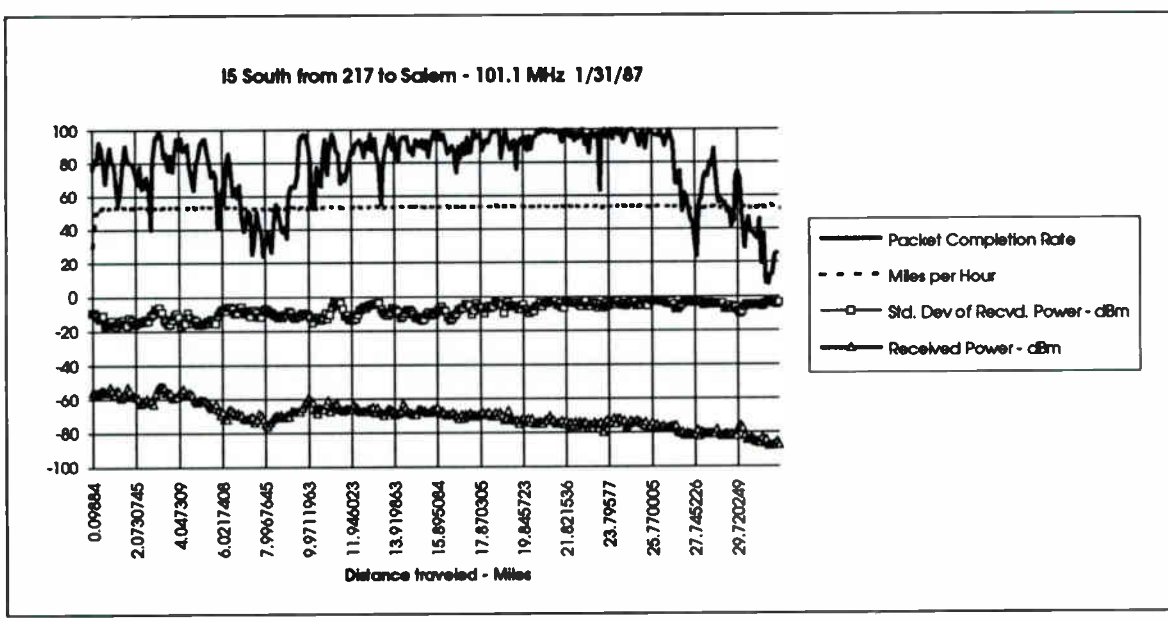

Figure 6

Field Tests

Results from two field tests in Portland, Oregon are shown. The first tests, in 1987, were used for initial range and performance evaluation including error correction techniques. The second tests, in 1989, were used to determine coverage of the then planned commercial system.

1987 Field Test -- Error Correction

Results shown are from field tests in Portland, Oregon in 1987 on a single station. The transmitter is 100 kW ERP on a tower 830 feet above average terrain on frequency 101.1Mhz located 45-30-55 N and 122-43-50 W.

A sample from a single run from the intersection of Oregon Highway 217 and Interstate 5 (7 miles from the tower) going South toward Salem on I5 (41.5 miles from the tower) is used for all the following analysis. This data is referred to as the I5 to Salem run.

Figure 6 shows basic results from the I5 to Salem run -- the packet completion rate (PCR), mean received power in dBm, standard deviation of the power in dBm, and travel speeds in miles per hour are plotted versus the distance traveled. The signal to the receiver was reduced to reflect a -30 dBd wrist band antenna. Each plotted point represents a distance of about 500 feet and the average of 250 samples of PCR and received power. The standard deviation is plotted as a negative value to help distinguish it from PCR data. Standard deviation of the received power reflects the degree of multipath -- the larger the deviation, the greater the multipath. Travel speed was maintained at about 55 miles per hour. Received power level is clearly declining as one moves away from the tower. The highest PCR appears to be in areas with adequate signal and minimal multipath as would be expected.

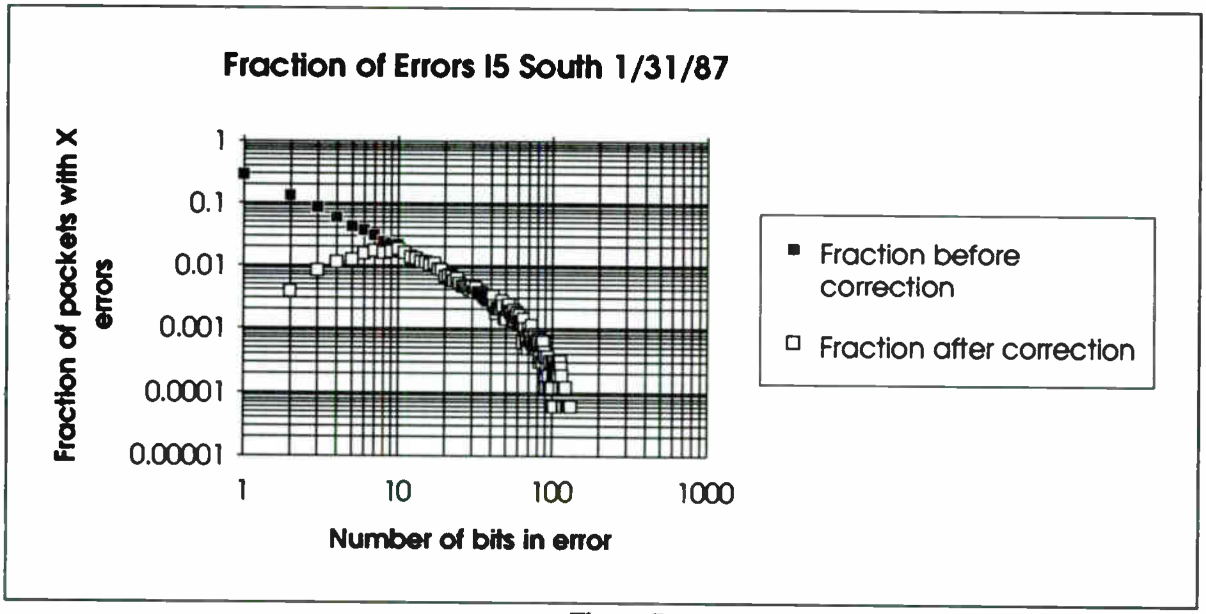

Figure 7

A study of bit errors and the effectiveness of the Error Correction method was performed. An analysis of the I5 to Salem run shows the packets sorted into four categories -- error free packets requiring no error correction, packets that could be corrected, packets that could not be corrected, and packets missed entirely due to failure to obtain word synchronization. Fully 64.1% were error free prior to error correction, 11.9 % were correctable, 7.2% were in error after error correction and 16.7% were missed due to lack of synchronization flag. (A later modification of the system decreased the percentage of packets with missed word synchronization by looking for either the leading flag of the packet of interest or the flag of the following packet.)

Figure 7 shows the correctable and not correctable packets sorted for the number of bits initially in error. This shows a nearly log log linear decline in the percentage and the number of the errors in the packets. All packets with 1 bit in error were corrected while the majority of packets with less than 10 bits in error were corrected. Again from figure 7, the correction scheme performed poorly with greater than 10 bits in error but there are few packets with such errors.

No error correction method can be 100% successful. Of the packets needing error correction, more than 62% of the packets were correctable, while 38% were not correctable. These results lead to the conclusion that repeating packets is necessary to provide satisfactory completion rates for receivers with very small mobile antennas and the error correction scheme should not be too extensive.

commercial operation: 95.5Mhz, 98.5Mhz, and 101.1Mhz. The recorded signal strength was taken from a quarter wave vertical monopole mounted on a vehicle's roof with its center height 2 meters above ground through a splitter to a spectrum analyzer measuring the power. The recorded Packet Completion Rate results were obtained using the same antenna used for signal strength and a -90 dBm receiver padded down to simulate a -30 $dBd_{vertical}$ antenna at a center height of 1 meter. Current MessageWatch™ receiver sensitivity of -91.5 dBm and antenna gain of - 29 $dBd_{vertical}$ specifications

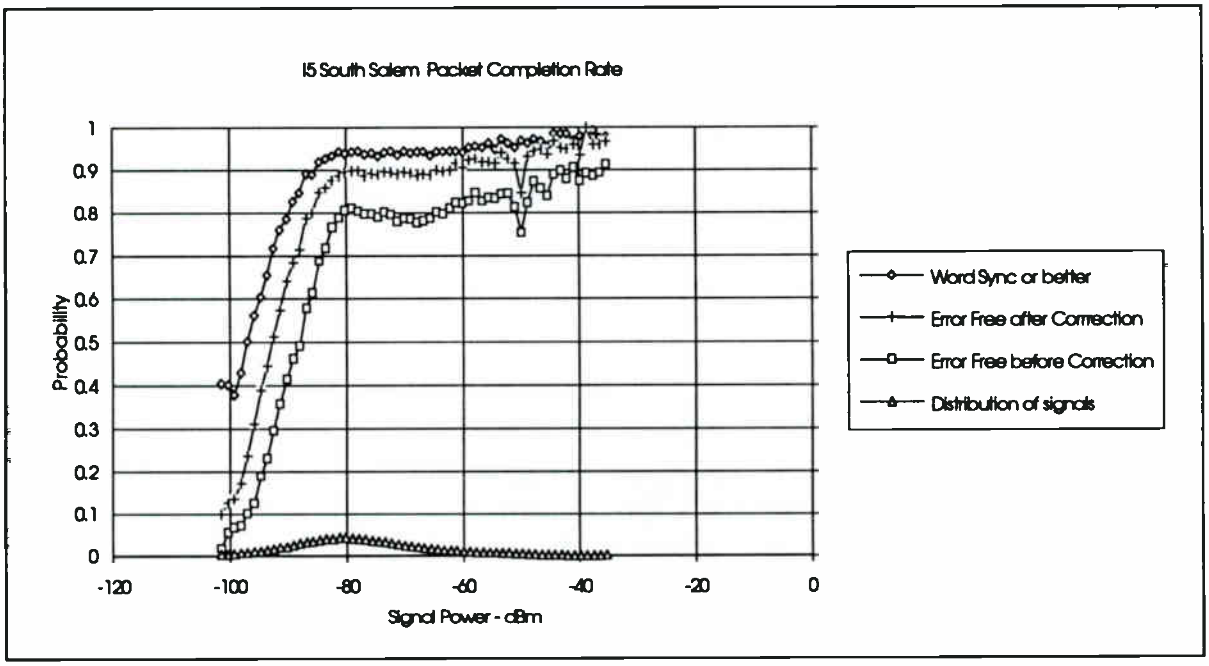

Figure 8

are 2.5 dB better than simulated in this testing.

The same data was then sorted based on received signal power and is plotted in Figure 8.

The distribution of samples from that run are also included. The effectiveness of error correction is clearly significant and is even very helpful at relatively high RF power levels. More extensive error correction schemes cannot improve these results significantly and the capacity of the system used by extensive correction schemes is better used for repeating transmissions.

1989 Field Tests -- Portland Range / Coverage

The following set up was used for 1989 Portland range / coverage testing. Three stations were selected for

A map of the Portland Area is provided as Figure 9 showing major towns. Thin lines or dots on the chart show the data points taken in the testing. Figure 10 shows signal strength contours of 60dBuV/m and 80dBuV/m for station 101.1Mhz. Portland is in a valley surrounded by mountains to the East and West that limit coverage in those directions. The Packet Completion Rate Maps - Figures 11, 12, 13 & 14 show areas of >90%, >69%, >55%, >44% PCR respectively for station 101.1 Mhz . These maps represent 90% or better Message Completion Rate areas for 1, 2, 3 and 4 transmissions respectively. The dramatic increase in coverage associated with multiple transmissions is clear.

Commercial Operations

Currently commercial service under the Receptor trademark is offered in two metropolitan areas: Portland, Oregon, and Seattle/Tacoma, Washington. Portland utilizes three FM radio stations to provide a coverage footprint with approximately 1.2 million people living within its boundaries. In Seattle and Tacoma, seven FM radio stations are used to provide a coverage area with 2.6 million people.

Beside numeric and standardized messages (such as Call Home, Call Office, etc.), more than 10,000 customers receive various combinations of the following informational services:

- Professional and collegiate football, basketball, and baseball scores
- Daily weather forecasts
- Local ski reports during winter months
- Daily Dow Jones Industrial Average closing information
- Winning daily and weekly lottery numbers

Summary

A high speed data system (HSDS) delivered over commercial FM radio subcarriers has been shown. It is a flexible system, providing inexpensive one-way wireless data communications that will be used in many applications.

References

Lender, A.(1963), "The Duobinary Technique for High Speed Data Transmission," AIEE Trans. Commun. Electronics, vol. 82, pp. 214-218.

Portland Data Points 4/28/89

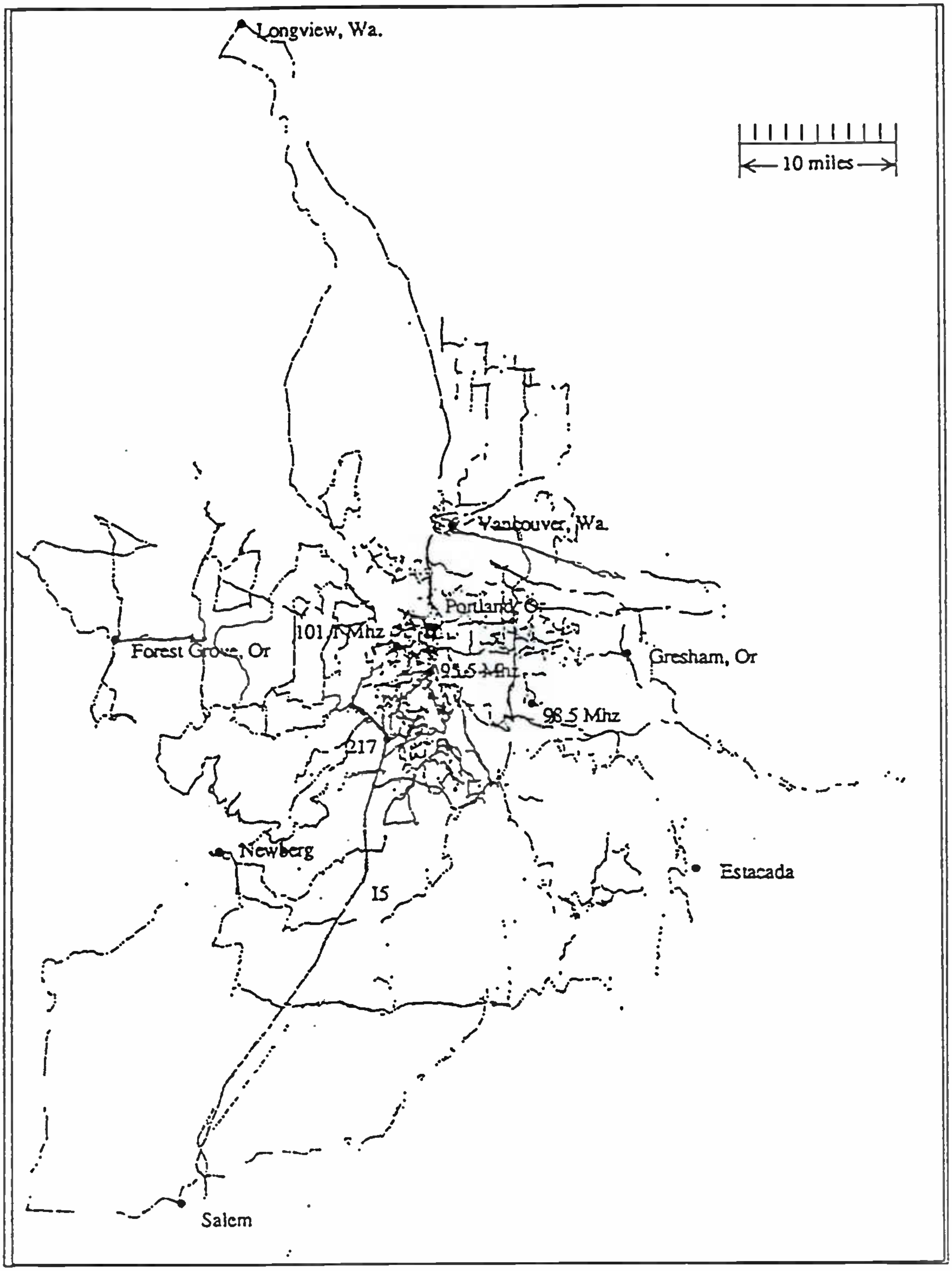

Figure 9

Portland Field Strength 101.1 Mhz 4/28/89

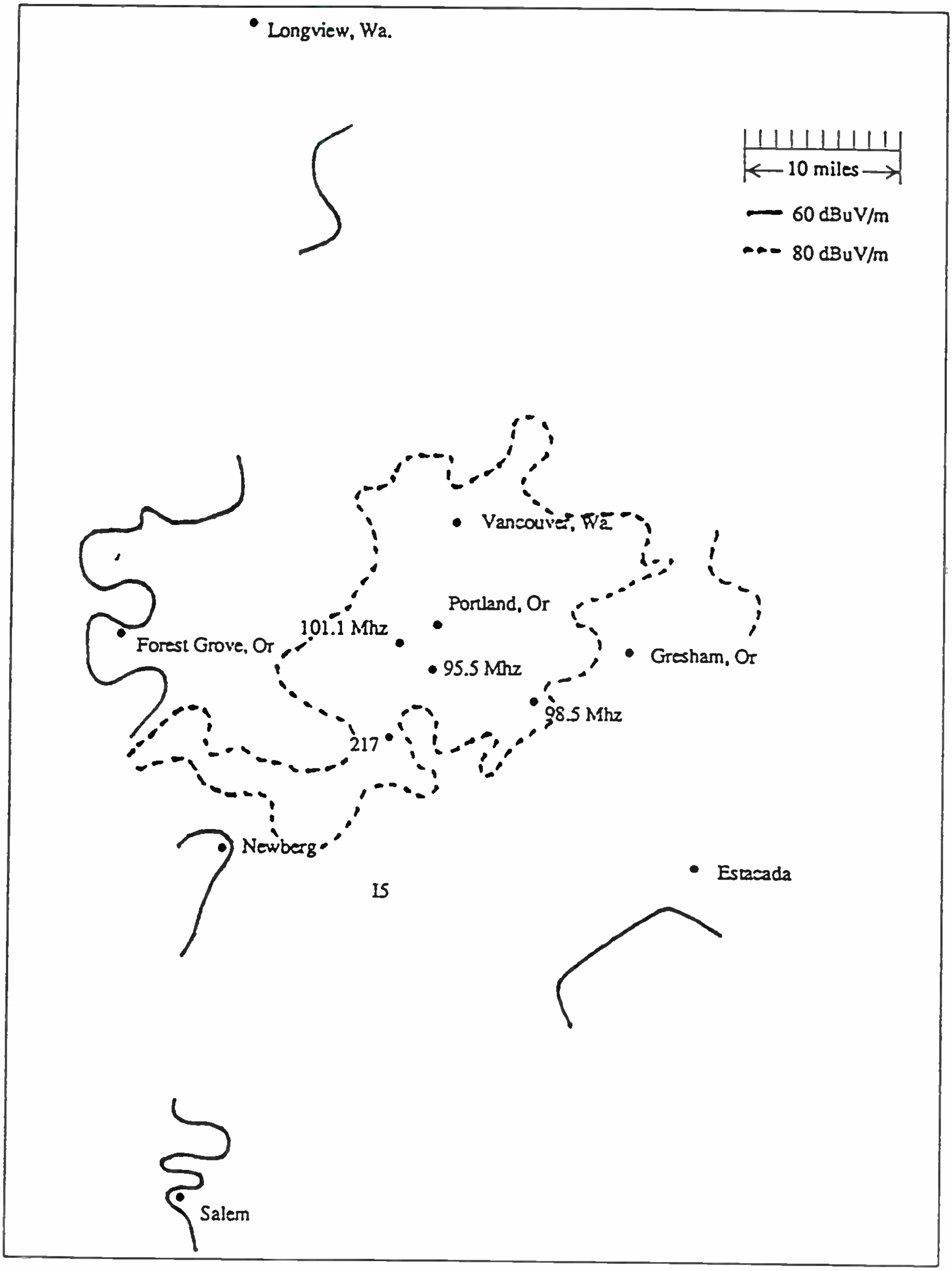

Figure 10

Portland >90% PCR 101.1 Mhz 4/28/89

Figure 11

Portland >69% PCR 101.1 Mhz 4/28/89

Figure 12

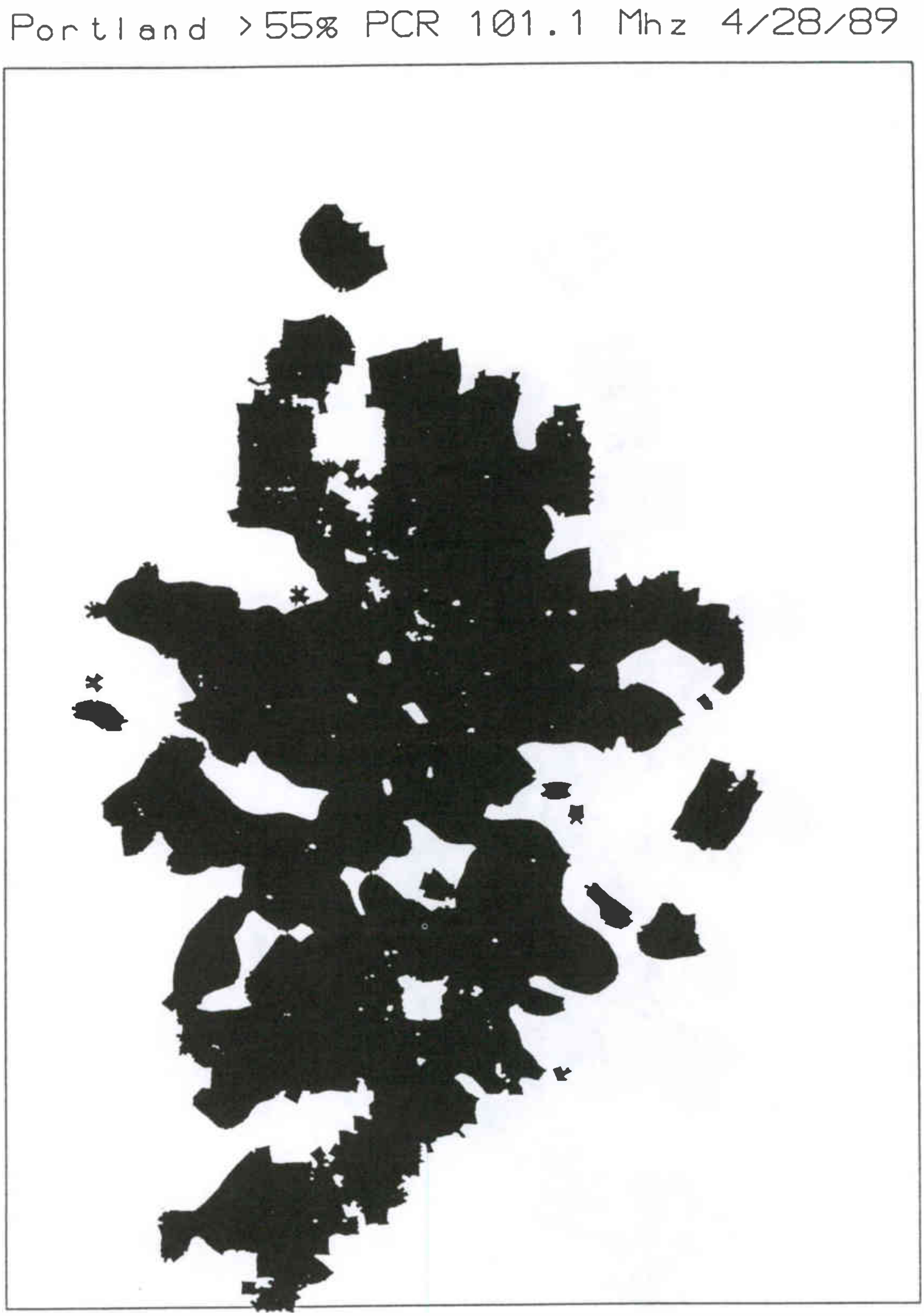

Figure 13

Portland >44% PCR 101.1 Mhz 4/28/89

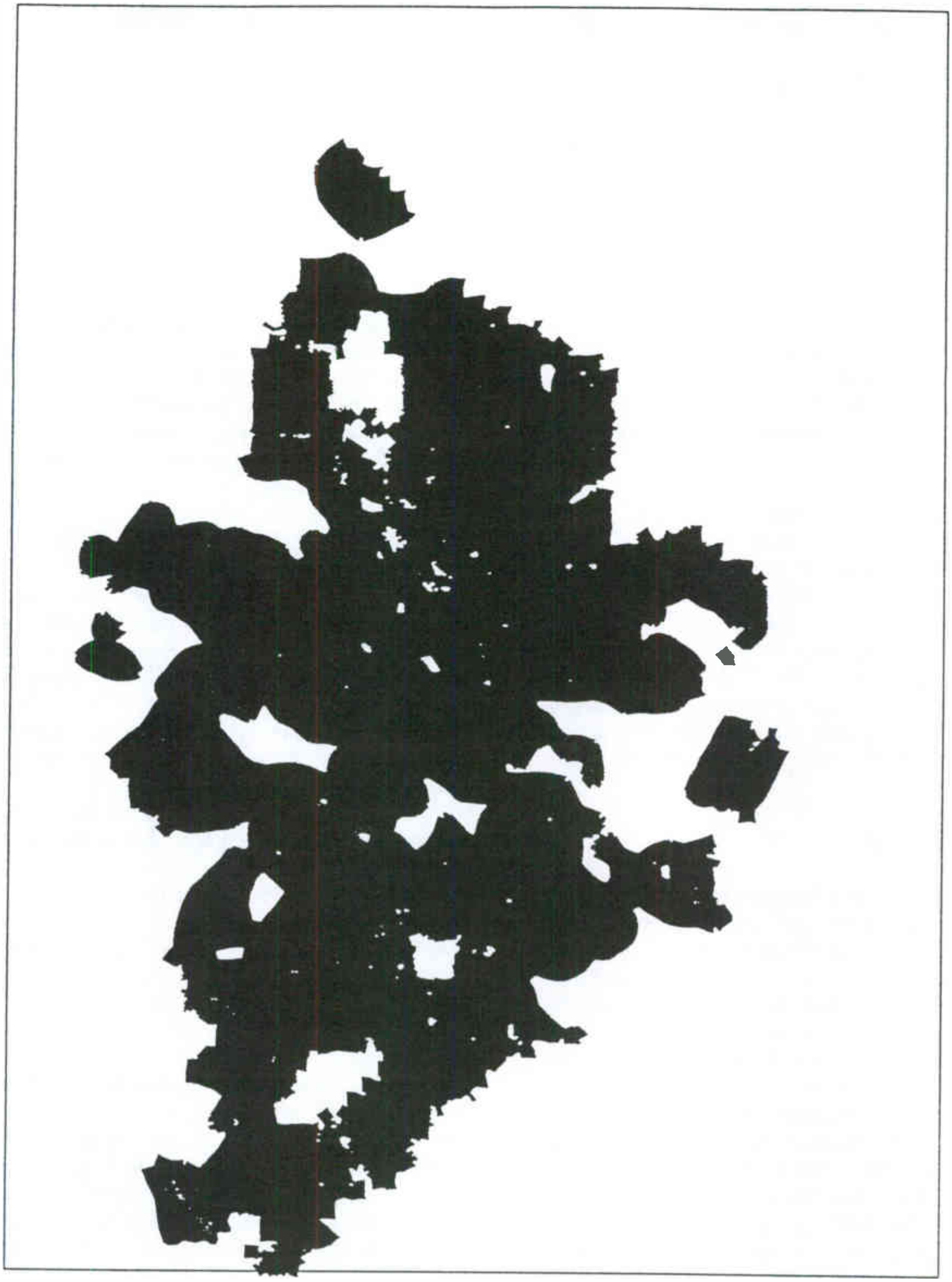

Figure 14

THE DEVELOPMENT AND STRATEGY OF AN FM MULTIPLEX BROADCASTING SYSTEM FOR MOBILE RECEPTION

Tetsuro Miyazaki and Tadaaki Tanaka
NHK Planning Division of the Engineering Administration Department and
Osamu Yamada and Tadashi Isobe
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ABSTRACT

NHK has developed an FM multiplex broadcasting system for mobile reception using a new modulation method called LMSK (Level controlled Minimum Shift Keying). The system is called DARC(DAta Radio Channel).
DARC has a transmission capacity of 16 kbps, more than 10 times that of the RDS (Radio Data System) of Europe. The system's specifications have been established.

For application of DARC, traffic information services are expected to be a key in Japan. In Tokyo in November of 1993, a large scale public demonstration was carried out to build a public opinion desiring early implementation of traffic information services.

1. INTRODUCTION

An FM multiplex broadcasting can provide new services such as traffic information, text and graphic information (news, weather forecasting, etc.) and data to control receivers (tuning and switching on and off, program identification, etc.).

In FM multiplex broadcasting, digital signals are combined with stereo sound signals using a frequency division technique. Therefore, new spectra are not needed and the cost of installation is very low.

In Japan, we have been studying FM multiplex broadcasting systems since 1985, at the Telecommunications Technology Council authorized by the Ministry of Posts and Telecommunications. The council set the standards of character and graphic broadcasting for mobile reception in May in 1993. The standards are based on the technique that was developed by NHK.

On November 9 and 10 of 1993, Vehicle Information & Communication System (VICS) Promotion Council held the public demonst-ration in Tokyo. The demonstration gave us the great opportunity to appeal the performance of DARC.

2. NHK'S FM MULTIPLEX BROADCASTING

NHK has developed DARC using a new digital modulation method called LMSK(Level controlled Minimum Shift Keying) which is a kind of MSK. DARC is multipath-proof, and has a transmission capacity of 16kbps. Moreover, it has a compatibility with the RDS system because they use different frequency bands.

DARC has many merits for traffic information services. The important merits are as follows:

(1) Proper reception in a moving car
(2) A wide coverage area
(Almost all of the stereo sound service area)
(3) A large transmission capacity of 16kbps

Figure 1 shows the baseband spectrum of the DARC signal. The multiplexing level of the digital signals is controlled by the L-R signal level. MSK with an amplitude level which is controlled as mentioned above is referred to as LMSK. By using LMSK, which increases the multiplexing level only during an increase of the modulation level of the stereo sound signals, the digital transmission with the best energy efficiency under multipath interferences is possible.

The product code of the (272,190) code and an interleaving technique have been adopted for correcting errors caused by fading. The error correction capability of the (272,190) code is so powerful that it is adopted for other digital systems such as teletext in Japan.

The LSI for demodulating LMSK signals and accomplishing error collection of (272,190) code has been developed.

3. TRANSMISSION TEST

Transmission tests were carried out to measure the performance of the 16kbps LMSK system in the service area of the NHK Tokyo FM station. Table 1 shows the transmission parameters of the test, All expressways and main national roads in the area were selected for the tests, and the measured distance totaled more than 1500km. Transmission data could be received without error during the 90% of all measured time by carrying out error correction with the product code of (272,190) code.

Generally in cars, we can't get 100% correct reception at all places due to intense multipath interference among high-rise buildings or cut-off in tunnels. It is necessary to transmit same data several times.

Figure 2 shows the roads where more than 95% correct reception rates can be obtained in the FM service area of the NHK Tokyo FM station. Here a correct reception rate is derived from the number of received correct pages out of the number of all transmitted pages, and one page is assumed to consist of 15 packets. (1 packet = 22 bytes)

A correct reception rate of more than 95% can be secured with one reception within approximately 20km of Tokyo Tower where the transmission antenna is installed. In case of three times iterative transmission, more than 95% correct reception rates can be obtained in almost all service areas.

4. PUBLIC DEMONSTRATION OF VEHICLE INFORMATION & COMMUNICATION SYSTEM

4.1 Vehicle Information and Communication System (VICS) Promotion Council

VICS Promotion Council was established in October of 1991. The Council consists of over 200 companies and organizations. Establishment of the Council had been planned by the group of promoters, which was based on mainly private companies such as car manufacturers and electric/communication manufacturers. The establishment had been supported by the three government offices - the National Police Agency, the Ministry of Posts and Telecommunications and the Ministry of Construction -.

The purpose of the Council is to promote the adoption of the VICS in to practical use within a reasonably short period of time.

The activities are as follows:

(1) Conducting investigations, research and development aimed at putting the VICS into practical use.
(2) Activities to spread the use of the VICS.
(3) Other activities necessary to achieve the purpose of the organization.

A number of interested organizations have been conducting researches on an information distribution system in response to various high-level needs of drivers.

Through the efforts of the National Police Agency, the Ministry of Posts and Telecommunications, and the Ministry of Construction, those researches were integrated into VICS. Consequently various activities were brought up to the national level.

VICS is similar to the Advanced Traveler Information System(ATIS) in the Intelligent Vehicle Highway System(IVHS), which is being developed for the United States by the Federal Highway Administration.

4.2 Public Demonstration of VICS

Public Demonstration of VICS was held in Tokyo, on November 9 and 10 in 1993. The demonstration aimed for the promotion of public understanding and recognition of VICS, in consequence, building a public opinion desiring early implementation and commercialization of VICS. In the demonstration about 600 volunteers experienced VICS by using 45 cars provided by 27 member manufacturers .

Figure 3 shows the total configuration of the demonstration experiment system. The information distribution media for communicating with drivers were radio beacons, optical beacons, and FM Multiplex Broadcasting. Road traffic information was sent from Tokyo Metropolitan Police Department and road administrators.

Symposium and exhibition were held on the same dates.

5. THE STRATEGY OF AN FM MULTIPLEX BROADCASTING

5.1 Traffic information services

FM broadcasting networks have already covered almost areas all over the country. Therefore drivers will be able to get traffic information everywhere and at all times if it is broadcasted on the FM broadcasting networks.

For VICS information services, three types of display are under consideration. DARC can transmit traffic data to all types of display as realtime traffic information. The three types of display are as follows:

Type 1: Map display
: To display traffic jams, accidents, etc. on the in-vehicle road map of navigation systems.

Type 2: Simplified graphic display
: Text and simplified graphics presented via an in-vehicle TV-type display that shows VICS information.

Type 3: Text display
: VICS information displayed in text on the liquid crystal display(LCD) of an in-vehicle radio set and audio set.

DARC has high transmission capacity. So even when traffic congestion is severe in Tokyo, it is estimated that traffic information can be sent twice every five minutes using a half transmission capacity. Traffic information is designed to be renewed every five minutes at present.

In practical application, it is effective to display road congestion information or traffic accident spots on a detail map made by navigation systems. In Japan, approximately 400,000 vehicles already have navigation systems installed, and a national-wide digital road map database of CD-ROM is on the market. DARC can send a large amount of real-time traffic information to those navigation systems as previously mentioned. Using real-time traffic information on the navigation systems, drivers can easily choose optimal routes to avoid traffic congestion. This will result in the reduction of driving time and traffic jams as a whole.

5.2 Other new services

Because of the mountainous terrain of Japan, many relay stations are constructed to eliminate blind areas from key stations. Drivers may have to tune their radio frequently while driving according to their location. Therefore, an automatic tuning function could be attractive for drivers when the DARC is introduced. It is also convenient for drivers to display the accurate time or the name of broadcasters.

In Japan, musical programs such as classical music, opera, and popular songs are mainly broadcasted in FM broadcasting. So it is considered to broadcast information relating to music programs, for example names of its composer and player, and song'texts.

A portable receiver using a special LSI is scheduled to appear on the market in the near future in Japan. The receiver is designed to have a two lines LCD display.

5.3 Cooperation with Digital DJ in the United States

At the NAB Radio show in Dallas on September of 1993, Digital DJ presented a new service using an FM Multiplex Broadcasting. Digital DJ adopted DARC for an FM Multiplex Broadcasting.

The Digital DJ receiver has an LCD screen. When a listener hears a song he likes and wants more information, he can look at the screen to learn the artist's name, CD name and other relevant information. Similarly, if a listener hears an interesting advertisement, he can not only receive additional information relating to the advertisement such as addresses and phone numbers, but also he can store the information on the Digital DJ memory and recall it later.

6. CONCLUSION

For application of DARC, traffic information services are expected to be a key in Japan. One of the reasons is that DARC is specially suitable medium for mobile reception. Another reason is that many drivers tend to demand more detail and more accurate traffic information especially in big cities like Tokyo.

DARC has many merits. It needs no new radio waves, and offers wide service areas, and low installation costs. Portable receivers with an LCD screen and in-vehicle receivers are to be on the market in the near future. Those receivers will soon prevail among people of all ages as highly intelligent receivers.

DARC is now under consideration in Sweden and Norway.

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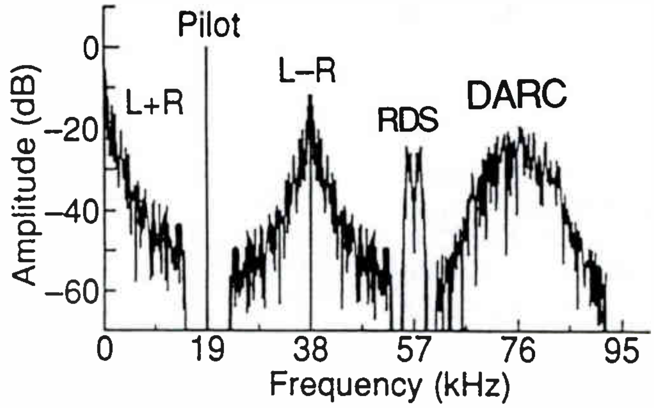
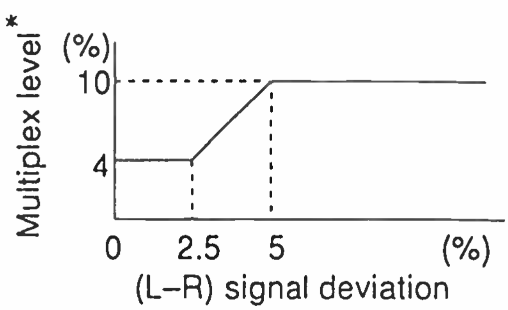


Figure 1 Baseband spectrum of FM multiplex broadcasting

Table 1 Transmission parameters

Frequency	82.5MHz
Tx Location	Tokyo Tower
Rx Location	Same area as FM stereo broadcasting
ERP(Watts) in the Direction of Rx	44,000
Antenna Height Above Sea Level(m)	212
Modulation method	LMSK
Subcarrier frequency	76kHz
Bit rate	16kbps
Multiplex level*	4-10%



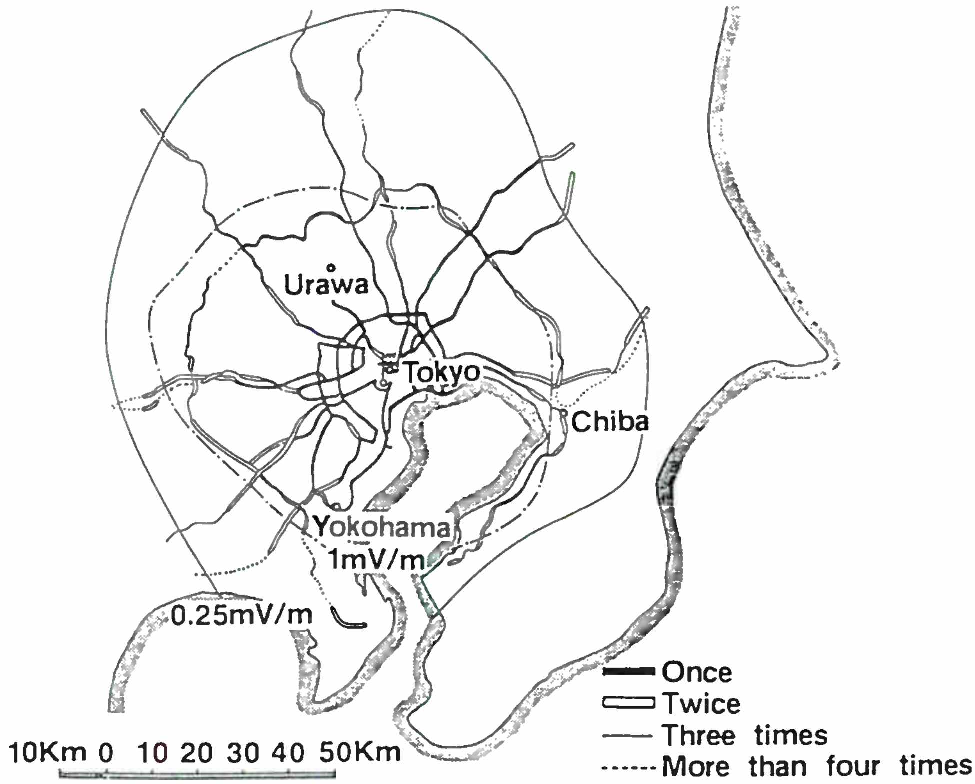
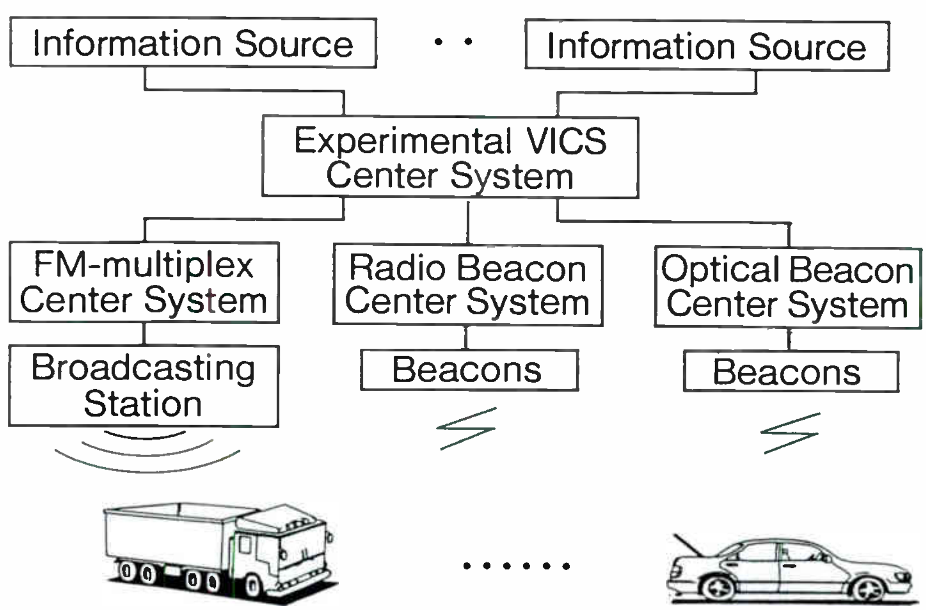


Figure 2 Routes having more than 95% correct reception rate



Vehicle and On-board Equipment

Figure 3 Outline of the Experimental System

ISDN AND T1 TRANSMISSION

Sunday, March 20, 1994

Moderator:
Margaret Bryant, WMAQ-AM, Chicago, IL

***OVERVIEW OF INDUSTRY COMMERCIAL DELIVERY OVER ISDN**
Angela DePascale
Global Digital Datacom Services, Inc.
E. Farmingdale, NY

NEW BROADCAST DECISIONS: DIGITAL NETWORK COSTS VERSUS REQUIRED BIT RATE FOR DESIRED AUDIO BANDWIDTH AND AUDIO QUALITY
James Switzer
RE America, Inc.
Westlake, OH

***ISO/MPEG LAYER III AUDIO CODING AND ISDN**
Steve Church
Telos Systems
Cleveland, OH

SELECTING A BRI ISDN TERMINAL ADAPTER
Lynn Distler
Comrex Corporation
Acton, MA

***THE NBA, CHANGING FROM SWITCHED 56 TO ISDN**
Doug Lane
National Basketball Association
North Reading, MA

THE GROWING IMPORTANCE OF DIGITAL TRANSMISSION FOR BROADCAST APPLICATIONS
John Kelly
Intraplex, Inc.
Westford, MA

DIGITAL TRANSMISSION ALTERNATIVES AND APPLICATIONS FOR AUDIO
David Anderson
IDB Communications Group, Inc.
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*Papers not available at the time of publication

NEW BROADCAST DECISIONS: DIGITAL NETWORK COSTS VERSUS REQUIRED BIT RATE FOR DESIRED AUDIO BANDWIDTH AND AUDIO QUALITY

James M. Switzer
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ABSTRACT

Availability of digital telephone network circuits such as ISDN, T-1, Fraction T-1 and Switched 56 is rapidly growing in the United States. These digital network services are fast becoming the broadcast standard for digital audio transportation. As broadcasters embrace digital audio codec technology they need to address a new set of issues related to bandwidth and quality of the desired audio. These issues are directly related to the available network bit rate speeds and network costs. This paper can serve as a "layman's" guide to broadcasters in achieving the proper mix.

INTRODUCTION

Transportation of audio program material for broadcast or studio use is in the final stages of completing an "evolutionary circle". In the early days of radio and television, programs were transported to studios or transmitter sites by the local telephone company or through AT&T long distance phone lines and coaxial cable. Analog lines and mechanical equalization were the method of choice, and in fact are still in use today. Since the 1960's, alternative transportation methods evolved, such as microwave, RF radios and satellite transmission. These alternatives brought down the cost of point to point and point to multi-point transmission of program material over long distances. These technologies replaced most of the program audio lines supplied by telephone company services.

Today, the digital "super highway" is growing, and with it brings the opportunity for broadcasters to return to the telephone company for program distribution. The "circular migration" back to the phone company is made possible with the advent of high speed microprocessors which, among numerous other uses, can be utilized for real time digitization of analog audio or video material.

Digital audio is being implemented in upgrade and new studio construction at an incredible pace. Several manufacturers have embraced popular international standards, or have implemented proprietary algorithms (the mathematical equations used for bit rate reduction of digital audio or video).

The broadcaster has many choices in traveling down the road towards full digital program implementation. Budgetary constraints often drive the decision making process for implementation of new technology. Of greatest concern is that a capital investment must be proven to support a need that returns a value consistent with the goals of the business.

In the case of broadcasting, the consumer is the focus, and the product is the program format. In order for new technology to fit in to the focus of broadcasting, it has to provide a value that affects the goals. In the case of digital audio transportation, the desired audio quality, and the cost of delivery is of primary concern. Hidden within these concerns is the assumption that the audio quality is maximized while the cost of delivery is minimized. In order to implement digital audio technology, the investment must provide short and long term benefits.

Identifying The Present And Future Needs

When planning the implementation of digital audio codecs for audio transportation on the public telco networks (or microwave and satellite), the short and long term benefits are not always obvious. Before a cost analysis of the proper digital network can be performed, the decision for the right equipment relative to the needs has to be made.

Identifying The Present And Future Needs, Continued...

The following considerations are recommended when evaluating a transmission system based on short term and long term needs:

A) :Present Needs: Define the current need that is driving the evaluation and selection of a digital audio transportation system:

1) What is the task?
2) What is the desired bandwidth?
3) Is it mono or stereo?
4) What level of subjective quality is needed ?
5) Is it a one way transmission or two way (duplex)?
6) Identify the digital bit rate transmission speed and type of network needed.
7) Select the proper digital audio codec.
8) Draft a relative capital and operating cost outline for present needs.

B): Future Needs: Define known future uses, anticipated potential uses, or create new potential uses for the equipment identified for the present needs:

1) What other opportunities exist that might extend the needs identified for the present?
2) What are the potential future bandwidths desired?
3) Is it mono or stereo?
4) What is the potential level of subjective quality needed to satisfy future potential?
5) Is it a one way transmission or two way (duplex)?
6) Identify the digital bit rate transmission speed and type of network needed.
7) Select the proper digital audio codec.
8) Draft a relative capital and operating cost outline for future potential needs.

C): Analyze the results:

1) Establish the incremental costs for the present and future needs of the audio equipment.
2) Determine if the incremental costs add incremental value (revenue).

A) Present Needs Discussion: Usually there is a planned need or event that drives the search process. It may be that a station needs a new STL or RPU system. Perhaps there is a need for a back-up system for redundancy or emergency operation. Whatever the case may be, it is best to define the need in writing to make sure that it is clearly identified.

A fictitious example for a Cleveland, Ohio station might respond to the recommended outline as follows:

Present Need Example:

Question 1: What is the task?: The station can obtain a contract for the broadcast of the local pro football franchise as long as it can supply audio quality greater than provided by standard telephone dial up lines (300 Hz to 3.4 kHz single line). In fact, the franchise recommends G.722 type 7.5 kHz quality or better. The RPU system can be mono since it is voice only.

2) What is the desired bandwidth?: The task above states a minimum of 7.5 kHz.

3) Is it mono or stereo?: The task above states a minimum for mono transmission.

4) What level of subjective quality is the needed ? The task above is for sports commentary, indicating a voice only need. Since it is a live feed situation, there will be little or no editing of the bit rate reduced audio. This implies that G.722 quality will suffice.

5) Is it a one way transmission or two way (duplex)? In this example, one way transmission is satisfactory if cues are performed from the off air signal. Additional cues can be accommodated on a standard phone line if necessary.

6) Identify the digital bit rate transmission speed and type of network needed. Since the identified G.722 technology is only capable of 56 or 64 kbit/s mono transmission, a switched 56 service would work. An ISDN (Integrated Service Digital Network), while providing up to 128 kbit/s on a standard BRI (Basic Rate Interface) will also suffice, even though it has twice the bit rate required.

7) Select the proper digital audio codec: Several manufacturers provide G.722 equipment. This type of equipment requires a network interface device as well. A DSU/CSU is needed for switched 56 service, or alternatively, a terminal adapter and an NT-1 is needed for ISDN. Equipment decisions for product selection will be discussed later in this paper.

Identifying The Present And Future Needs, Continued...

8) Draft a relative capital and operating cost outline for present needs: For the present needs, it would appear that the broadcaster should buy a G.722 capable audio codec and an appropriate DSU for switched 56 service, or a terminal adapter for ISDN service. For this Cleveland, Ohio example, switched 56 service is available, but so is ISDN. ISDN is half the cost of switched 56, making this the network of choice in Cleveland. (It is common across the U.S. for ISDN to be at least half the cost of switched 56, however, many broadcasters may only have access to switched 56 service, as ISDN is in the early stages of deployment).

The key present needs have now been identified for this example. The following cost outline in Fig.1 for both capital and operating expenses for the identified digital audio equipment can be drafted:

QTY	Capital Equipment	Cost:	Notes:
2	G.722 codecs, cables	$4,500	(based on average market pricing)
2	ISDN TA's w/NT-1's	$3,000	(based on average market pricing)
	TOTAL:	$7,500	

QTY	Operating Cost Function	Cost:	Notes:
2	ISDN Install (Ameritech)	$ 260	($130 per line, one line at each end)
2	Season Phone Bill (Ameritech)	$ 400	($40 per line for 5 month season)
	TOTAL:	$ 660	

Fig. 1

The total capital outlay is $7,500 and the total operating cost for the season is $660. This is assumed reasonable considering the expected market share and advertising revenues for eight home football games.

B: Future Needs:

The scenario for the football broadcast venue in the present needs example above identifies enough equipment to do the job. When identifying the needs of today, it is best to search into the magic "crystal ball" to identify the possibilities of its use in the future. The questions posed for future needs as outlined previously are now answered (relative to the fictitious example currently being evaluated):

1) What other opportunities exist that might extend the needs identified for the present? The way the broadcast of the football games is being presented for present use could be done another way. The present needs example identifies a mono 7.5 kHz transmission. Even though the 7.5 kHz bandwidth of the broadcast is adequate for the football franchise minimum requirements, neither the AM or FM station is broadcasting the game at its own full transmitter bandwidth potential. Stereo broadcasting is a possibility as well. The broadcaster might place the main commentary on one channel and color commentary on the other, while crowd noise is common to both channels. Local station inserts during the game broadcast would have an obvious greater bandwidth and a noticeable switch between mono and stereo sound if the mono 7.5 kHz equipment identified in the present needs scenario is followed.

For these reasons, a stereo 15 kHz remote transmission could be the preferred choice.

Additionally, the equipment investment identified in the present needs scenario is underutilized during the time available between game broadcasts and after the football season has ended. Present needs show the equipment is in use for a single "season", and if not used for other events, will sit idle during the non-seasonal use.

The same equipment identified for use during the football season may also be used for other events. It is important to identify the type of events that the equipment can be used for.

In the Cleveland example, there are many other remote broadcast event opportunities. As the home of the "Rock and Roll" hall of fame, there will be musical event opportunities. Other venues include the Cleveland Convention Center and the International Exposition Center which offer a variety of events such as The Flower Show, The Car Show, Sports Shows and many more. Several area shopping malls offer seasonal events or guest appearances of famous people. Cleveland also offers The Cleveland Air Show, The Cleveland Grand Prix auto race, and others. In the "Flats" area of downtown Cleveland there are several popular comedy clubs, restaurants and Nautica, an outdoor sound stage on the Cuyahoga River.

B: Future Needs Continued...:

These venues may require up to 15 kHz stereo as well as mono 7.5, or 10 kHz bandwidth. The G.722 equipment identified in the present needs description above would not suffice.

2) What are the potential future bandwidths desired? The answers to question 1 above identifies up to 15 kHz.

3) Is it mono or stereo? The answers to question 1 above have identified stereo.

4) What is the potential level of subjective quality needed to satisfy future potential? The answers to question 1 above have identified musical events as well as greater bandwidth for commentary, which require a higher degree of subjective quality and bandwidth than that of G.722.

5) Is it a one way transmission or two way (duplex)? In the above examples, one way will suffice.

6) Identify the digital bit rate transmission speed and type of network available. For 15 kHz stereo, the digital audio codec will need a minimum of 112 or 128 kbit/s.

7) Select the proper digital audio codec. Several manufacturers provide audio codecs which meet the needs of 15 kHz stereo transmission (or more). This type of equipment requires a network interface device as well. A DSU/CSU is needed for switched 56 service, or alternatively, a terminal adapter and an NT-1 is needed for ISDN. Some codecs can provide built in ISDN terminal adapters. For this example, an RE 660/661 ISO Layer II digital audio codec is used.

8) Draft a relative capital and operating cost outline for future opportunities. To accommodate the future needs costs to meet the above scenarios, the following chart in Fig. 2 applies:

QTY	Capital Equipment	Cost:	Notes:
1 1	RE 660 Encoder RE 661 Decoder cables	$6400 $100	(based on current list price)
2	ISDN TA's w/NT-1's	$3,000	(based on average market pricing)
	TOTAL:	$9,500	

Fig. 2

Note: For full duplex operation at full bandwidth, add a second digital audio codec ($6,400).

QTY	Operating Cost Function	Cost:	Notes:
2	ISDN Install (Ameritech)	$1,040	($130 per line, estimating eight total lines for the year)
2	Season Phone Bill (Ameritech)	$ 960	($40 per line for 2 average full time lines per year)
	TOTAL:	$2,000	

Fig 2, Continued...

C): Analyze the results:

1) Establish the incremental costs for the present and future needs of the audio equipment. The above cost chart in the future potential scenario assumes an ISDN line at the football stadium, Broadcast station and 6 major remote broadcast function locations utilizing a flexible digital audio codec such as the RE 660/661 for a total capital investment of $9,500 and an operating cost of $2,000. The incremental capital cost over the present needs example to meet the future needs example is $2,000, with an incremental increase of $1,340 in operating costs.

2) Determine if the incremental costs add incremental value (revenue). If the equipment is purchased to satisfy the present needs outlined in the example above, the opportunity cost of lost events due to lack of equipment, or the cost of replacing or purchasing additional equipment to meet the future needs becomes obvious. By planning future needs into the present budget, a significant cost savings will be realized for the long term, while adding only a marginal cost at present.

While this example does not apply the actual revenue and value of the additional venue potential for the broadcaster, it is assumed that there is significant market appeal and advertising revenues in favor of recommending the equipment identified in the future opportunity scenarios.

Conclusion to the example evaluated above:

With a minimum incremental cost for capital and operating expenses, significant additional event coverage as well as improved transmission performance for the original task has been proven. By planning the present needs with direct respect to the future potential, significant future cost savings of replacement or upgrade equipment has been realized.

Supporting The Assumptions

The audio quality, bandwidth and mode (stereo or mono) requirements are used to determine the minimum bit rate requirements for the desired transmission capability. The minimum bit rate needed is initially defined by the application (such as demonstrated in the present and future analysis example earlier). The network bit rate may be limited to what is currently available from the local and long distance telephone service provider, either by physical presence or monthly recurring network cost to the broadcaster.

The outline used to demonstrate a hypothetical example of present and future needs address issues such as subjective audio quality, required bit rates, audio algorithms, network terminal equipment and other issues. The following further clarifies these issues.

Subjective Audio Quality:

Audio quality can be measured electrically, according to accepted standards, or subjectively, according to the perception of the listener. Since the subject of this paper deals with the use of digital audio codecs which utilize bit rate reduction techniques, it is proper to focus on the subjective quality of the delivered audio program.

It is not the intent of this paper to rank the subjective quality of the available digital audio algorithms on the market today. This paper should allow the non-technical management decision maker to make educated decisions without an in depth technical background, or the need to understand the intricacies of how digital audio algorithms work.

For further studies related to digital audio, a list of articles and publications appear at the end of this paper.

International Standards:

It is important for the broadcaster to recognize the value of international standards. These standards are under the scrutiny of international committees and are subject to improvements and modifications, with the intent that manufacturers adhere to these standards. Standards provide a degree of interoperability between different manufacturers of equipment. While this is the ideal goal of a standard, it is not always the case. For the most part, however, interoperability does exist at common levels and modes of operation.

Existing international standards for digital audio bit rate reduction systems include CCITT J.41, CCITT G.722, and ISO MPEG 1, Audio Layers I, II, and III. ISO MPEG 2 is currently in development and is expected to be adopted by November, 1994. [1]

J.41 is among the oldest standards, and is a 2:1 fixed system commonly found in telephone company use for 15 kHz telco carrier audio. The audio is "companded" to produce a 384 kbit/s output per channel. While the quality is very good, the high bit rate requirements of 768 kbit/s for a stereo transmission force this standard in to expensive telco services such as half or full T-1 service.

CCITT G.722[2] is a relatively old standard (about 20 years) that utilizes two bands of 4 kHz audio with 8 bit resolution, assigning 6 bits resolution to the lower band and two bits resolution to the upper band. This means that frequencies over 4 kHz are subject to significant impairments in subjective audio quality. Subjectively, G.722 sounds good for commentary, but is bandwidth limited to 7.5 kHz. This standard has been commonly embraced by broadcasters simply because it has been relatively inexpensive to use.

ISO MPEG 1, Audio layers I, II, and III was adopted in November, 1992 in the audio section of International Standard ISO/IEC 11172-3 (Coding of Motion Pictures And Associated Audio For Digital Storage Media At Up To About 1.5 Mbit/s).

The Layer II algorithm, at 16 bit resolution, has received the greatest attention around the world, as its use is intended for digital audio broadcast, motion pictures, contribution and distribution networks.

The Layer III algorithm currently offers improved audio quality over Layer II at lower bit rates such as 56, 64, 96 and 112 kbit/s. Layer III is significantly more complex to implement, and is generally more expensive to purchase.

Because of the complexity of Layer III, most of the improvements to the ISO MPEG 1 algorithms occur for Layers I and II. In fact, nearly all of the attention is to Layer II. This is due in part to the fact that the Layer II decoder is being implemented in low cost chip form for consumer product usage. Other factors include the use of Layer II for DAB. The decoder follows a "generic" design that receives decode instructions in the incoming bitstream. Improvements to the encoding algorithm will be automatically accepted by the generic Layer II decoder.

International Standards, Continued...:

This allows improvements to the encoder without the need, in most cases, to modify the decoder. Finally, greater attention is given to Layer II based on the progress of the Eureka standard as well as organizations such as CCIR who are recommending or in the process of preparing a recommendation to use Layer II for worldwide broadcast applications.[3]

New standards proposals to Layer II also include multi-channel sound and Lower Sampling Frequencies (Layer II LSF).

Layer II LSF further increases the subjective low bit rate (56 to 112 kbit/s) audio quality to a point which easily meets the CCIR defined requirements for low bit rate commentary codecs (comparable to the subjective quality of Layer III)[4].

Fig. 3.0 below lists the author's recommended applications for use with the CCITT G.722, and ISO Layer II digital audio algorithms. The intention of this chart is to demonstrate the acceptable bit rates of digital networks as they apply to specific venue usage.

Recommended Applications Relative To Network Bit Rates

Algorithm	Bit rates Supported (kbit/s)	Typical BW (kHz)* mono/2 channel	Audio Mode: Mono	Dual Mono	Stereo	Joint Stereo	Recommended Applications
G.722	56/64	7.5 kHz	✓				Voice Only
ISO Layer II	56	11 kHz	✓				Commentary, Sports ⇩
ISO Layer II	64	11/6 kHz	✓	✓	✓	✓	" " ⇩
ISO Layer II	96	20/6 kHz	✓	✓	✓	✓	" " ⇩
ISO Layer II	112	20/12 kHz	✓	✓	✓	✓	RPU, Advertising, Music ⇩
ISO Layer II	128	20/17 kHz	✓	✓	✓	✓	" " ⇩
ISO Layer II	192	20/20 kHz	✓	✓	✓	✓	STL, TSL, STS ⇩
ISO Layer II	256	20/20 kHz		✓	✓	✓	Concerts, Post Production ⇩
ISO Layer II	320	20/20 kHz		✓	✓	✓	High quality ⇩
ISO Layer II	384	20/20 kHz		✓	✓	✓	Contribution networks

Fig. 3

* Typical Bandwidth estimates an average bandwidth assumed to be usable based on sample rate selection and algorithm limitations at the available bit rate

Selection Of Equipment

There are two types of equipment associated with the transmission of digital audio:

1) The digital audio encoder and decoder (or codec)
2) The telco network interface device.

The Digital Audio Codec: Technology seems to be advancing rapidly in the digital world of microprocessors. Personal computer obsolescence is the best example. Last year's 386 is replaced by this year's 486, which will be surpassed by the 586 Pentium. Such rapid obsolescence has created a fear in buying "current" technology. However, when you need a computer to do a job, you end up buying what is available anyway.

The computer analogy can be applied to digital audio codecs. If you buy G.722 technology today, you may wish you had bought something more powerful in a very short period of time.

Knowing that modern audio algorithms will improve as enhancements and modifications are made, it is essential to select a product that offers easy upgrade features. Upgrades are generally done by changing DSP firmware chips. This process requires access inside the codec, attention to electrostatic sensitivity, removal and insertion of the new DSP firmware chips. This can be cumbersome when the digital audio equipment is rack mounted in a studio or portable remote set-up.

The Digital Audio Codec, Continued...:

The easiest upgrade method is from those manufacturers who utilize "flash prom" technology. This allows firmware changes to be completed from a PC using a simple connection to the RS-232 data port on the digital audio encoder or decoder. This process takes less than ten minutes to upgrade, and can be performed by the less technical person.

A product which offers more features than needed at the present may prove to be quite appropriate in the long term value of the product. Look for digital audio inputs capable of handling AES/EBU and SPDIF digital formats. Many manufactures offer this as a cost upgrade feature. Digital audio codecs from RE provide these inputs as a standard feature. By having the digital audio input and output capability built in to the product today, it will not require a future upgrade (or replacement) as the broadcaster moves towards full digital implementation.

Other features to consider should include remote control capability, multiple sample rate selection, auxiliary RS-232 data capability, audio monitor points, and ease of operation. A feature found in RE audio codecs include built in sample rate conversion and studio reference clock synchronization. This may be a key feature for those broadcasters who a currently planning a digital studio upgrade.

The Network Terminal Equipment:

Switched 56 (SW-56) service requires a DSU/CSU network interface device. This device converts the bit rate reduced digital audio information into a bitstream formatted for transport across a SW-56 service. When using a bit rate of 56 kbit/s for mono broadcasts, such as when using G.722 equipment, one DSU/CSU and one SW-56 network connection is required. When using a stereo codec or greater bandwidth in the mono mode, two SW-56 lines and two DSU/CSU's are required. The digital audio encoder or decoder then performs a function called "inverse multiplexing", which sums the two lines for a virtual 112 kbit/s network.

When using ISDN, the network interface device is referred to as a terminal adapter (TA). Some TA's require an outboard device known as the network termination or "NT-1". The TA is capable of supporting two "B" bearer channels of up to 64 kbit/s and a data (D) channel for dialing.

The NT-1 electrically conditions these digital channels in to the final ISDN two wire output, known as the "U" interface. Some brands of TA's have the NT-1 built in, and offer a form of inverse multiplexing called "BONDING". When summing two "B" channels, either BONDING or the inverse multiplexing in the digital audio equipment can be used, however, the method has to be the same at both ends of the transmission.

There are several popular brands of DSU/CSU's and terminal adapters. Discussions on this equipment exceed the space allowed in this paper. For additional information on network terminal equipment, please refer to or request The Digital Network Access Guide, available from RE America, Inc. and its distributors.[5]

Pricing the Network

By answering the questions which identify the present and future needs, a minimum bit rate for digital audio transportation can be determined. The application will determine whether or not the network needs to be "switched", or "dedicated'. STL's or permanent remote pick up venues which originate a large number of program hours per month are generally better served by dedicated services. Dedicated networks are point to point fixed locations, with fixed monthly recurring service charges. Backup STL's, or limited usage venue locations are better served by "switched" networks. Switched networks cost less per month than dedicated networks, but have a per minute usage charge for local and long distance connections.

Switched 56 is generally available throughout the United States, most of Canada, and even a limited number of locations in Europe. ISDN is widely available in Europe, generally available in Canada, and just now becoming available in the major markets of the United States.

The following chart (next page) shows approximate current rates and services available from the largest service providers. A phone number is provided for each listed company. These numbers can help the reader acquire further information in the respective telephone company coverage areas. This information is subject to change without notice. It is advisable to call for current rates, as tariffs are re-filed and approved at different times for each company listed.

Pricing the Network, Continued...

Digital Network Cost Reference Chart:
Bold pricing indicates Centrex ISDN rates

Major Telco	SW-56 Install	SW-56 Monthly	ISDN Install	ISDN Monthly	For More Information, Call:
Ameritech	$550-850	$107	$120-150 **$124-337**	$30-40 **$44-68**	1-800-TEAM DATA
Bell Atlantic	$725	$150	$120-150 **$123-207**	$28-37 **$38-56**	1-800-570-ISDN
Bell South	$630-755	$90	**$163-222**	**$19-46**	1-800-428-ISDN
GTE-Contel	$125	$50	$30-50	$10-18	214- 718-5608
NYNEX	$412-533	$90-100	$150-195 **$105-220**	$29-89 **$35-102**	1-800-GET-ISDN
Pacific Telesis	$500-725	$45-197	**$70/$220**	**$27/$30**	1-800-622-0735
S. New England Tel.	$600	$80	March 94	March 94	203- 771-5111
Southwestern Bell	$480-700	$70-105	$154	$55.50	314- 235-9553 Texas Only
Sprint (Centel, United)	$20-150	$30-95	Pending	Pending	913-624-3972
US West	$150-200	$50-75	$67 **$350**	$85 **$98**	303-965-7013 (IA/NE) 404-422-8238 (OR)

Long Dist. Carrier Mileage sensitive rates	SW-56 1st 30 sec Add'l 6 sec	ISDN (per 64 kbit/s) 1st 30 sec Add'l 6 sec	For More Information, Call:
AT&T peak off peak	$.135-.191 $.01-.022 $.125-.185 $.009-.02	$.135-.191 $.01-.022 $.125-.185 $.009-.02	1-800-222-SW56
MCI	$.018-.044 $.006-.02	Call For Pricing →	(214) 918-5850
US Sprint	$.06-.127 Per Min	Call For Pricing →	Call local offices

Fig. 4

The digital network cost reference chart demonstrates the price difference for installation and monthly usage costs between switched 56 and ISDN networks. It is important to note that calls placed from an ISDN connection can communicate to switched 56 services and vice versa as long as it is done at 56 or 112 kbit/s.

The cost of dedicated digital network services must be priced individually by each telephone company. The supplied telephone contact numbers are valid for dedicated digital network service pricing as well.

The switched 56 service pricing usually has a per minute usage fee for local calls (plus any long distance per minute fees if applicable). In additional to the monthly ISDN service charge, additional per minute or per call usage sensitive rates may apply as well.

For more information on switched and dedicated networks, and for further discussion on other digital networks such as T-1, please refer to the RE America, Inc. Digital Network Access Guide.[6]

Conclusion

When evaluating a digital audio compression (bit rate reduction) system as a solution for a current audio transportation need, it is best to look to the long term needs as well. By identifying future potential needs now, a significant cost savings can be realized. By answering only a few questions based on present and future needs as outlined in this paper, the broadcaster can add significant flexibility, with only a nominal additional expense, to obtain a digital transportation system which meets the needs of today and tomorrow.

References:

[1]Kerkhof, L.v.d.,Nielsen, S., Stoll, G.,: "Generic Architecture of the ISO/MPEG Audio Layer I and II:Compatible Developments to Improve the Quality and Addition of New Features". 95th AES Convention, New York, Oct. 1993, preprint #3741, p.9.

[2]CCITT recomendation G.722 (3291) AP IX-142-E, pp 231-318. See also, Bellcore Technical Reference TR-TSY-001044, Issue 1, December, 1989, "Generic Requirements For 7 kHz Audio ISDN and Private Line Point To Point Applications".

[3]Kerkhof, L.v.d.,Nielsen, S., Stoll, G.,: "Generic Architecture of the ISO/MPEG Audio Layer I and II:Compatible Developments to Improve the Quality and Addition of New Features". 95th AES Convention, New York, Oct. 1993, preprint #3741, p.1.

[4]Kerkhof, L.v.d.,Nielsen, S., Stoll, G.,: "Generic Architecture of the ISO/MPEG Audio Layer I and II:Compatible Developments to Improve the Quality and Addition of New Features". 95th AES Convention, New York, Oct. 1993, preprint #3741, p.11-12.

[5]Switzer, James M., "RE America, Inc. Digital Network Access Guide", 2nd Edition, January, 1994.

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Recommended Readings On Digital Audio Bit Rate Reduction Systems:

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SELECTING A BRI ISDN TERMINAL ADAPTER

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Abstract

Terminal adapters (TA) are required to connect Basic Rate Interface (BRI) ISDN lines to data terminal equipment (e.g. digital audio codecs). Selection of a TA can be confusing, since there are many TAs available with a wide range of features. Different broadcast situations will use the flexible facilities of BRI ISDN in different ways and, unfortunately, different manufacturers and versions of ISDN telephone switches can require different TAs. This paper gives a background on the basic functions of a TA and then gives an overview of issues that should be considered in selecting a TA for connecting digital audio codecs to BRI ISDN.

ISDN BACKGROUND

The Integrated Services Digital Network (ISDN) was designed as the telephone system of the future. In the early-to-mid eighties, when the system was being standardized, it was believed that ISDN would be the solution to all voice and data problems. It was assumed that our homes and offices would all be connected at the incredibly fast (at that time) data speeds afforded by the network. Because it encompassed so much, the ISDN standards grew to many complex volumes.

Because an investment had already been made by the public in lower speed data equipment, it was felt that some type of conversion would be required to allow this equipment to work on ISDN. The solution was called the terminal adapter, a device which would incorporate the complexities of ISDN on one end, and on the other, provide a simple, standard user interface. It was felt at the time that a terminal adapter would be a temporary measure until the deployment of inexpensive, ISDN-ready equipment.

Since the sluggish introduction and deployment of ISDN, other technologies have forged ahead and, in some cases, surpassed it. LANs, a major application for ISDN, became faster (10 MB/s or more) and made the 64 KB/s data channels offered by ISDN obsolete for that application. Users of high speed LANs could not be bottlenecked by an ISDN WAN, so work began (and continues) on technology to support LAN like rates across the telephone network. (Broadband ISDN and ATM are examples).

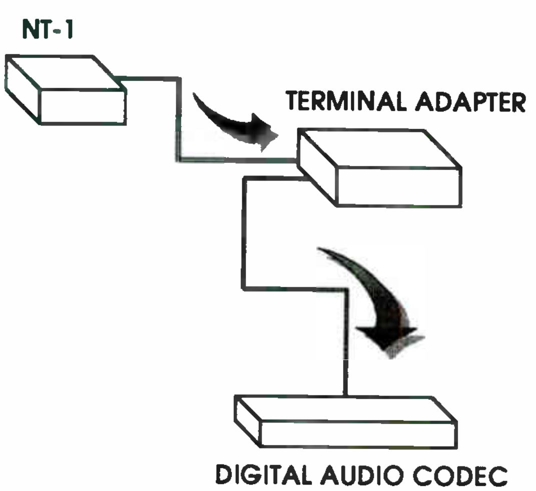
ISDN has found its place, however, in videoconferencing, T1 backup, and high quality audio transfer. Because it simply has not become the universal solution it was planned to be, few manufacturers have found it attractive to implement a complete ISDN interface to their data equipment and telephones. Instead, they have taken the logical approach of allowing the terminal adapter to function as the permanent ISDN interface and focus their talents on their own equipment. Because the ISDN specifications have grown so complex and implementation of these specifications is so varied, the need for an external device to deal with these problems makes the terminal adapter a device which is very likely here to stay.

THE SPECIFICS OF BRI ISDN

The Basic Rate Interface of the Integrated Services Digital Network (BRI ISDN) is the simplest and most commonly available type of ISDN service. It provides user access to two 64 KB/s Bearer (B) Channels and one 8 KB/s Delta (D) Channel. Each B Channel can be specified to carry full-duplex data (Circuit Switched Data) or analog, voice-grade audio (Circuit Switched Voice) - or alternate voice and data. (Packet switched data is also an option for the B channel, but would not normally be useful in the applications discussed here.) The D Channel is used by telephone companies for supervisory functions and is also available to the user for packet switched data. The terminal adapter you select will need to support the specific BRI ISDN line configuration you order from the phone company.

THE EQUIPMENT REQUIRED

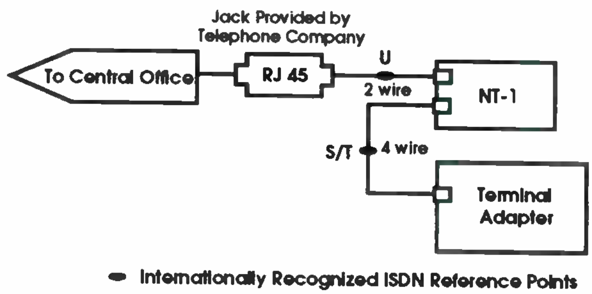
Figure 1
Customer Premises Equipment (CPE)



Three pieces of equipment are required for sending high quality audio via ISDN lines: the NT1, the terminal adapter, and the digital audio codec. (see Figure 1) To digress into some "telephonese," the NT1 and TA are referred to by the phone company as the Data Computer Equipment (DCE.) The digital audio codec is called the Data Terminal Equipment (DTE.) And all three pieces of equipment are referred to by the phone company as Customer Premises Equipment (CPE) - in other words, you own it!

THE NT1

Figure 2
BRI ISDN Installation



Before we look at the TA, you should be aware that you will probably also have to provide a Network Termination Unit (NT1). The NT1 is usually customer provided equipment in North America because line connection is made directly into the "U" reference point, before the NT1 (See Figure 2.) In Europe, for example, connection is made at the "S" or "T" reference point - so the NT1 remains on the telco side of the circuit. The NT1's function is to convert the BRI-U two wire RJ-11 type connector provided by the phone company into a BRI-S/T four wire RJ-45 type connector used by the terminal adapter.

Some currently marketed terminal adapters include a built-in NT1, eliminating the need for purchasing one separately. However there could be an advantage in having a stand-alone NT1 because it typically includes two ports. Since the terminal adapter only requires one port, the second port may be used with an ISDN type digital telephone for separate voice communications, although this can be used only during times when there is a free B channel.

THE TERMINAL ADAPTER

ISDN terminal adapters provide essentially two functions:

- They adapt data to a form that can use the ISDN Network.

- They handle communication with the network, including details such as call setup and dialing.

COMPATIBLITY ISSUES

It is important to be aware of what is required in a terminal adapter in order to be compatible with local ISDN lines and with the requirements of codec applications. There are three main compatibility issues that one should consider.

I Switch Types

All ISDN service in the United States is not alike. Different telephone companies use different switches. This causes great confusion and has initiated the institution of a national standard for ISDN services. This standard is known as NI-1 or National ISDN Standard. Not all service is entirely conforming to this standard at this time so the differences in switches must be considered. The two most commonly used switches are AT&T's 5ESS and Northern Telecom's DMS100, and there are a variety of software versions of each. There are other switch types but they are not nearly as common and usually conform to NI-1 standard.

It is important to know what kind and version of switch your phone company will be using and make sure that your terminal adapter is compatible with that switch. Most terminal adapters are compatible with both 5ESS and DMS100 but a few are not.

The type of switch used also significantly impacts the way the terminal adapter will be programmed, so you will need to know what type switch you will be connecting to. The terminal adapters which are dedicated to one switch are often easier to set up in terms of programming so, in cases where less versatility is required and ease of initial set up is important, these terminal adapters may prove to be the best choice.

II Data Connect Type

Another potentially confusing area regarding terminal adapters is exactly how the TA will transfer the data from the ISDN line to your terminal equipment (digital audio codec). Various electrical protocols and connectors exist to achieve this.

V.11 (also known as RS-422) is a common protocol which specifies electrical levels. In North America, this is usually located on a 25 pin "D" connector. In Europe, these levels are often incorporated onto a 15 Pin "D" connector and, in conjunction with some handshaking signals, implemented in what is called an X.21 interface. Under most circumstances, an appropriate cable will convert these two protocols.

The most common interface in North America is V.35. This protocol specifies levels which are different from V.11, and is usually implemented on a larger, 34 pin Winchester-type connector. To make things even more confusing, some manufacturers have put V.35 levels on 25 pin "D" connectors.

All these protocols are similar in that they allow balanced, synchronous transfer of data. It is important, however, to be sure your codec and TA speak the same language in terms of protocol, or that they can be easily adapted.

III Data Rate and Rate Adaptation

Data Rate and its adaptation must be considered. A terminal adapter capable of at least 64 kb/s of synchronous data exchange is necessary for use of a digital audio codec. Frequently in North America, telephone carrier systems allow only 56 Kb/s clear channel transmission. For this reason, ISDN Terminal Adapters must have the correct protocol to adapt to 56 Kb/s. There are many rate adaptation standards but V.110 is the most commonly used for the 64 KB/s to 56 KB/s adaption.

FUNCTIONALITY FEATURES

There are several features which may be offered in terminal adapters which may affect its suitability to a particular broadcast application.

I Number of Ports

The most significant feature which can affect a terminal adapter's suitability for broadcast applications is the number of ports it contains. There are many terminal adapters which include only a single port, and this may be sufficient for a simple bidirectional audio feed. If expanded facilities are desired, however, you may need separate ports for each B channel, plus maybe an access port on the D channel for packet switched data. We've given some examples of multiport uses at the end of this paper.

Note: A BRI ISDN line allows separate calls to be dialed on each of its B channels. With some switches and TAs, it is possible to dial two calls to the same location by programming only one telephone number and one Service Profile Identifier (SPID) into the TA. (The SPID is an identification number which the telephone company gives you that is specific to the service they are providing). However, the Northern Telecom DMS100 switch requires two separate SPIDs and telephone numbers to place two calls and your TA must allow for the duality. This means that not only does the TA need two physical ports, but that each port is fully internally supported, allowing programming of two separate SPIDs and two telephone numbers.

II Type of Ports

We have already stated that the B channel may be ordered with circuit switched voice, circuit switched data or alternate voice/data. Some terminal adapters provide an analog port for one of the B channels, allowing connection of a standard telephone. There are also TAs which allow you to externally program whether the B channel port will be set up for analog (voice) or data. The most common TA configuration, however, is for two data ports.

III Call Control

Another significant feature which may make a difference in ease of use is call control. There are many ways that calls can be initiated.

a) Keypad dialing - The simplest method is via a keypad. On some models, numbers may be dialed on the keypad, stored to memory and then dialed at the touch of a single button. A keypad can also facilitate the initial programming of the TA. (It is surprising how many good Terminal Adapters do not include a keypad.)

b) RS232 - This method of dialing involves use of a separate PC for dialing. We've seen some very nice uses of this method of call control in larger scale applications.

c) Front panel programming - On models which do not have a keypad, dialing is accomplished through the use of an up/down switch and an accompanying LCD display. The number is stored by using a switch which indicates whether the number on the LCD display should be higher or lower. After the number is correctly brought to the display, it may be stored to memory and dialed by pressing a single button. Initial setup parameters are programmed using the same method. Call information and progress may also be shown on the LCD display. This process can be cumbersome and, if it is anticipated that numbers will need to be changed with any frequency, a keypad may be a better option.

d) Preset autodial - This is an option which allows the Terminal Adapter to automatically dial a pre-programmed number whenever a data terminal is placed on line and issues an active Data Terminal Ready (DTR) signal.

g) Auto-Answer - This option allows the terminal adapter to automatically answer an incoming call and to disconnect the line when the calling party hangs up.

IV Inverse Multiplexing

In applications requiring data rates higher than 56/64 KB/s, it is necessary to combine a sufficient number of 64 KB/s B channels to form the desired data rate. Inverse multiplexing is the generic term for combining multiple lower bit rate channels into one high bit rate channel and there a several terminal adapters on the market that provide this capability. BONDING is one protocol for inverse multiplexers (IMUX). Another is H.221. Several IMUX manufacturers implement their own proprietary protocol, but also include a BONDING option for compatibility with other manufacturers. BONDING stands for Bandwidth on Demand Interoperability Group.

Several codecs have a built-in IMUX to combine two B channels into 128 KB/s, but there are applications where it may be useful to have this function occur in the terminal adapter. One such case would be if codecs from different manufacturers were being used that incorporated proprietary IMUXes that were not compatible. In this instance, BONDING compatible TAs could be place in front of the codecs to make them work with each other.

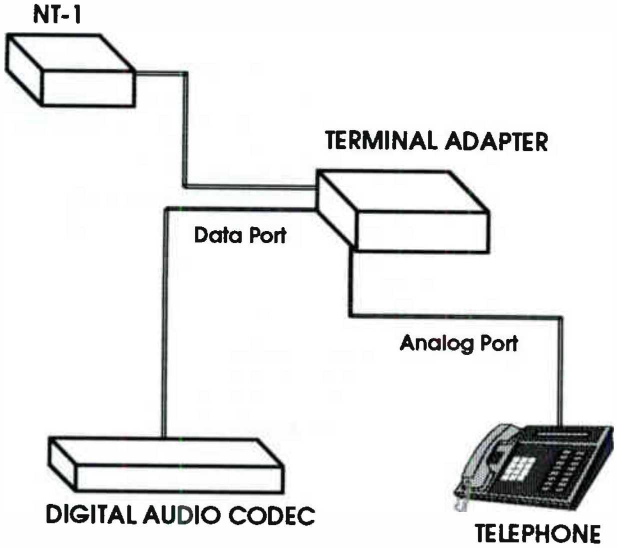
There is also at least one manufacturer with a TA that provides interface for four BRI ISDN lines to produce data rates up to 512 KB/s. This can be very useful in situations where lower data reduction codecs are used to transmit full bandwidth audio.

V Programmability

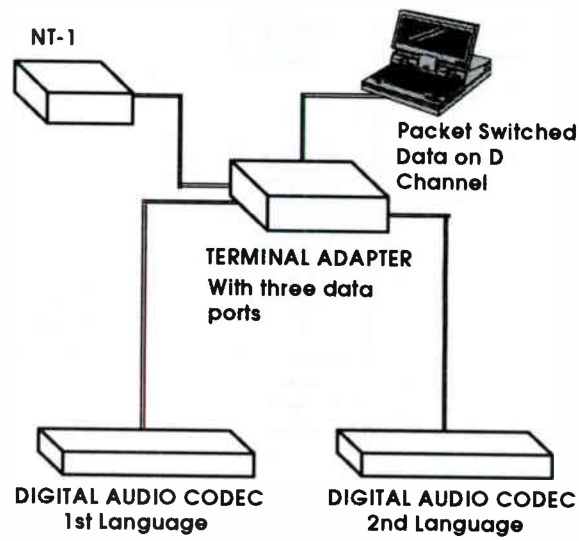
One other factor to consider is ease of set up. On this front, terminal adapters are not created equal. Some TAs frequently operate on their default settings and others may take hours to program. Terminal adapter salespeople seem to be very candid about discussing setup, so you may want to "pick their brains" before purchase.

SOME REAL LIFE APPLICATIONS

Scenario 1
Sports Remote



Scenario 2
Dual Language Commentary



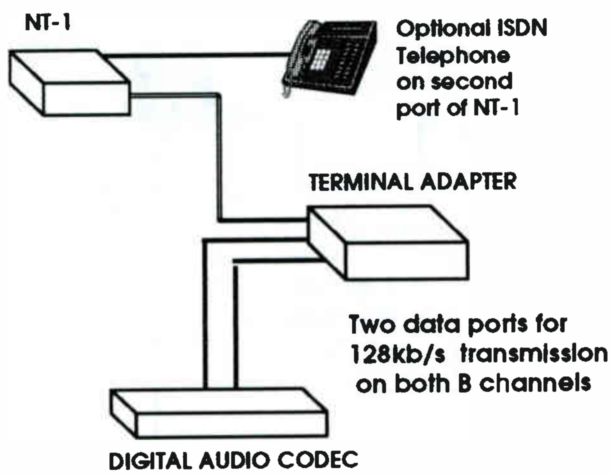
Requirements: High quality voice (7.5 KHz) program transmission. Return audio for "mix minus." Additional, two way communications channels for voice coordination, modem, or fax machine.

Solution: Install one BRI ISDN phone line. This line, having two independent B channels, can be configured so that one B channel carries circuit switched data (for the digital audio codec) and the other carries voice. Likewise, your terminal adapter should have one data port (probably V.35 or X.21) and one voice port (with an RJ-11 jack). Since the codec works in full duplex, return audio can be sent back down the codec to the talent in the field.

Requirements: Two high quality voice (7.5 KHz) channels which may be independently dialed. Return audio on each channel. X.25 packet switched data for E-mail messaging.

Solution: Install one BRI ISDN phone line. This time, configure both B channels for circuit switched data and order packet switched data for the D channel. A TA with three data ports is required. 7.5 KHz digital audio codecs provide program transmission and return audio for each language and a laptop computer can be connected to the D channel for messaging to and from the remote site.

Scenario 3
Remote Broadcast of Concert

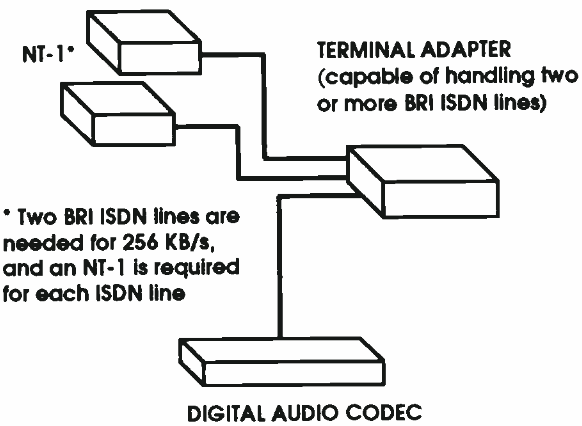


Requirements: Inexpensive access to wideband (15 KHz or better) stereo program channel Optional dial-up voice communications when B channels are not in use.

Solution: Install one BRI ISDN phone line. Both B channels on the line should be configured for circuit switched data, and a dual data port terminal adapter will be required. An inverse multiplexer (built into either the codec or the TA) is needed to sum the data capability of each B channel to provide enough data so that a wideband codec (with extremely high data reduction) can pass 15 KHz or more stereo audio.

An ISDN telephone can be connected to the second port of the NT-1, allowing voice communications during times when a B channel is not in use.

Scenario 4
Dial-up Studio-to-Studio Link
(at 256 KB/s)



Requirements: Wideband (15 KHz or better) discrete stereo transmission with low delay and high transcoding immunity.

Solution : Install two or more BRI ISDN phone lines. The TA should provide interface to all phone lines used and have the ability to inverse multiplex them to the desired data rate (256 KB/s or more). Use a low data reduction codec which can work at these rates.

TO SUMMARIZE

1. Does your TA have the correct number and types of ports for what you want to do? In other words, does your audio codec use one or two data B channel ports? If it uses one, do you want voice capability on the second B channel, or would you prefer alternate voice/data use? Do you require access to the D channel for packet switched data?

2. Does your TA support the type of dialing you want? Do you want to dial each number via a keypad on the TA? Would you like to enter numbers into memory and "speed dial" via the front panel. Or, for more complicated call management, perhaps you need external control via an RS232 port.

3. Does your TA support the specific BRI ISDN service configuration you have ordered through the telephone company and can your TA work adequately with the CO switch version that will be providing your service? There is a wide variety of switch types and versions within these types that support ISDN. And these are constantly being revised by the telephone companies. TAs require setup parameters unique to these types and versions and TA manufacturers are constantly providing software updates to maintain compatibility.

In fact, one of the main factors in selecting a TA might be finding a manufacturer who is committed to helping you through the setup process and can help you down the road when switch software revisions come along. We note that some TAs have a supervisory port to allow remote setup of the TA by the manufacturer. Considering the war stories we've heard from people going through the initial ISDN/TA setup, that sounds like a very attractive idea.

This paper is not to warn you off ISDN, but rather to caution you that it is not "plug and play." The high quality audio and cost savings are fantastic. Just don't plan to carry your TA to live remote an hour before air time and expect to be up and running when you need to be.

To make your job easier, you may want to work with an authorized service vendor of your telephone company. These companies can order the ISDN lines and provide the terminal adapter... and help you make it all work. The authorized service vendors are paid commissions by the telephone companies, so there is no extra cost to you.

Finally, an international note of caution. Unfortunately, although ISDN is meant to be a global standard, it certainly is not there yet. TA's that work in one country are likely not work in another. Check to see that your TA has been "homologated" in the country in which you want to operate. Many of the manufacturers we've contacted sell their terminal adapters worldwide and can advise you on TAs which are suitable for specific countries.

The following is a list of terminal adpater manufacturers and authorized service vendors (*) we contacted while gathering information for this paper and we thank them for their input. This, however, is not meant by any means to be an exhaustive list of either category.

Adtran
901 Explorer Boulevard
Huntsville, Alabama 35806
205-971-8000 Fax 205-971-8699

* Allcom, Inc.
5621 West Howard Street
Niles, IL 60714
312 763-6100 Fax 312 763-6115

Controlware
1325 Campus Parkway
Neptune, NJ 07753
908-919-0400 Fax 908-919-7673

Develcon
4470 Chamblee-Dunwoody Road
Suite 200
Atlanta, GA 30338
800-423-9210 Fax 404-936-0711

Fujitsu
ISDN Systems Division
3055 Orchard Drive
San Jose, CA 95134
408-954-1088

Gandalf
1 City Place Drive
Suite 420
St. Louis, MO 63141
800-677-0303 Fax 314-567-3275

* Global Digital Datacom Services, Inc.
100 Broadhollow Road, Suite 310
East Farmdale, NY 11735
516 694-6805 Fax 516 694-6906

Integrated Network Corporation
757 Route 202/206
Bridgewater, NJ 08807
908 218-1600 Fax 908 218-0804

* OSI Networks, Inc.
9430 Topanga Canyon Blvd., Suite 103
Chatsworth, CA 91311
818 700-6200

Northern Telecom
800-667-8437

TelePower
6451 Independence Ave.
Woodland Hills, CA 91367
818-587-5540 Fax 818-587-5546

Transtream, Inc
28025 Dorothy Drive
Agoura Hills, CA 91301
818-706-3252 Fax 818-706-1509

UDS Motorola
5000 Bradford Drive
Huntsville, AL 35805
800-451-2369 Fax 205-830-5657

THE GROWING IMPORTANCE OF DIGITAL TRANSMISSION FOR BROADCAST APPLICATIONS

John Kelly
Intraplex, Inc.
Westford, Massachusetts

Abstract

High Speed digital transmission has been available to broadcasters since the divestiture of AT&T in 1983, but its use for broadcast applications has been limited, until recently. A number of influences, particularly compression and falling costs, are accelerating the deployment of digital services for broadcast applications. This paper analyzes the movement toward digital transmission for broadcast applications and provides guidelines to determine the applicability of digital transmission to specific applications.

THE EVOLUTION OF THE DIGITAL NETWORK

Digital networks that are available to the broadcaster include T1, 56 kilobit per second dedicated leased lines, Switched 56K, and ISDN. Each of these services is a derivative of T1 service which was introduced into the AT&T network in the early 1960s. T1 originally provided an analog conversion of pulse code modulated voice signals into a 64 kilobit per second data stream, and combined twenty four of these data streams into a single, four wire transmission facility. T1 operates at a transmission rate of about one and one half million bits per second. (This number will become important later.) T1 was useful to the telephone monopoly at that time because it saved wire plant (four wires instead of forty eight) and because digital transmission was inherently more reliable than analog transmission, particularly over long distances.

In 1983, AT&T divested itself of the local operating companies and was ordered to offer services on a cost provisioning basis. As a result, T1 which was used for intertoll traffic (traffic between central offices) became available for commercial customers. In the ensuing 20 years, T1 has become enormously popular, and its use is no longer limited to the multiplexing of voice signals. In fact, T1 today carries a wide variety of different traffic, including high quality audio signals generated by and for broadcasters.

The T1 "product" itself has evolved over the years, and telephone companies have created a number of service offerings founded upon the T1 architecture. While many of these services have usefulness for the general business needs of the broadcaster, it is the transport of high quality audio that is the concern of this paper. Services useful to the broadcaster for the transmission of high quality audio include T1, Switched 56K, and ISDN.

THE BENEFIT OF DIGITAL TRANSMISSION

There are many factors that explain why T1 has increasing acceptance in the broadcast industry. At the heart of this acceptance is the acknowledgment by the industry that digital transmission is superior than analog transmission, especially over long distances. Why is this?

We don't need to be physicists to understand that a signal degrades when transmitted over a medium - any medium. This degradation takes on many forms, but is generally perceived as unwanted noise that is added to the original signal. I say unwanted because the broadcaster's objective is to deliver a signal to his listener with the same clarity as the signal at its origination.

During analog transmissions, the signal strength weakens over distance and is amplified to restore the signal to its original strength. Unfortunately, additive noise is also amplified.

During digital transmissions, the signal strength also weakens, but an entirely different process is used to restore the signal. Since the "restoring equipment" only needs to distinguish between "zeroes" and "ones", on the transmission path, the signal can be regenerated instead of being amplified. And, at the point of regeneration, unwanted noise can be eliminated.

As a result, a signal can be delivered to a remote location in substantially its original form, regardless of the distance between sender and receiver, when transmitted over digital facilities. Broadcasters appreciate this characteristic, and today seek to install digital equipment for their systems wherever practical.

BANDWIDTH REQUIREMENTS FOR AUDIO TRANSMISSION

We must also understand the conversion of high quality audio signals from analog format to digital format before we can explore applications using digital audio transport. The quality of a compact disk provides 20 kilohertz of stereo bandwidth. The Federal Communications Commission, however, allocates only 15kHz of spectrum bandwidth to radio stations, so we will demonstrate the bandwidth requirements using this figure.

An engineer named Nyquist stated that, if you wanted a true sample of an analog signal, you had to sample at a frequency at least twice the frequency of your signal. (Just take my word for it on this one.) 15kHz audio is often sampled at 32kHz because of this rule, and each sample is represented by a 16 bit "word". We can determine how many data bits per second are required by multiplying the 16 bits times the 32kHz frequency. The answer is 512 thousand bits per second.

And that's just for the left channel! A stereo pair needs, then, 1,024,000 bits per second under this formula. What's important about this number is that a level of audio quality acceptable to the industry can be transmitted faithfully (and uncompressed) within the 1.5 million bits per second of a T1.

Ah, compression. Modern compression techniques allow this digitized signal to be reduced significantly without offensive result. Broadcasters have adopted compression when multiple audio channels were transmitted between locations, or when the required bandwidth was not available. There are many compression flavors and scales. A popular compression scheme provides a 4:1 compression, allowing 1,024,000 bits of uncompressed linear stereo audio to be transmitted in 256,000 bits.

EARLY USE OF T1 FOR BROADCAST APPLICATIONS

With this background information about digital networks and the bandwidth required for compact disk quality audio, the following discussion of digital broadcast applications should be more easily understood.

The earliest use of digital T1 services to carry broadcast quality can loosely be titled "Terrestrial Extension of Satellite/Microwave Circuits". Events around the world, particularly sports and political events, were transmitted by satellite to large broadcast networks such as ABC, NPR, Mutual, and others. Often the satellite uplink antenna could not be co-located with the event, and so a T1 circuit was established between the event and the satellite uplink facility. In these applications, broadcast audio was digitized and compressed at the event site, and transmitted over T1 to the satellite uplink site. Depending on the satellite service, the audio was sent digitally to the broadcast network and returned to its analog state, or returned to analog at the satellite uplink and transmitted analog over the satellite.

An early example of this application occurred in 1983 at the Reagan - Gorbachev Summit Meeting in Reykjavik, Iceland. Multiple audio feeds were combined over a single T1 at the meeting site, sent via digital circuits to a satellite uplink facility, and distributed across the United States by a large broadcast network provider. In this way, reporters were able to deliver news reports and stories with the same clarity as if they were at the studio.

T1 BROADCAST NETWORKS

The example above has been repeated hundreds of times since the Iceland summit meeting and the viability of audio over T1 circuits has continually proven to be effective. Today, broadcasters continue to install T1 on a temporary basis to cover large events, whenever a link between an event site and an audio distribution network needs to be established. Political conventions, caucuses, Olympics, other sporting events, remote news coverage (examples include the coverage of events during the Waco, Texas massacre), and celebrity fundraisers and concerts (Billy Joel - Live from Moscow was carried on a digital T1 to satellite network) are all examples of the usefulness of T1 as an audio transport facility.

T1 is not limited to temporary status. Today there are hundreds of T1 circuits that carry compact disk quality audio. Large broadcast networks such as ABC, NPR, Mutual, Unistar, and Christian Broadcast Networks receive their programming from a variety of sources, including T1 connections to program generating studios. Also, T1 circuits connect these broadcast network providers to their satellite uplink facilities. Finally, some of these networks are interconnected by T1 facilities, so that audio programming can be shared among network providers.

International Networks

Internationally, the use of digital networks (called E1 and running at 2 million bits per second, but otherwise similar to T1) is also escalating. Internationally, radio is frequently state owned, and programming is created by a single or small number of state owned studios. This programming is then "contributed" to individual radio stations over circuits called "contribution lines". (Radio stations that have distance between studio and transmitter use circuits called "distribution lines", similar to STLs in the US.)

The prevalent method of establishing contribution lines in the 1980s was over satellite circuits. Just as in the United States, however,other countries' telecommunications infrastructure is being dramatically changed as a result of the deployment of fiber optic cable. As fiber is installed, the cost of T1/E1 terrestrial lines drops dramatically, and these circuits become cheaper and more reliable than satellite circuits.

International broadcast networks have two components that use high speed digital circuits. First, program creation can occur at multiple locations, and programs must be shared between sites. T1/E1 circuits connect these sites and audio is transmitted, both "real time" and delayed. Second, the programs are "contributed" to radio stations over digital T1/E1 circuits for "distribution" to the surrounding community. This topology tends to create broadcast networks that are large, integrated, and require close cooperation between the telephone and broadcast divisions of each country's government.

T1 FOR STL'S, LMA'S AND DUOPOLIES

Were this paper to close at this point, the reader would justly conclude that the use of T1 for broadcast applications was limited to a handful of large broadcasting companies or governments, building complicated network infrastructures. But the opposite is true, and the most prevalent use of T1 in broadcast applications is simple point to point connections between two radio stations, or a radio station and its transmitting tower.

When a radio station is separated from its transmitter by distance, the choices for establishing this necessary connection are limited. By far the most common method is through a special microwave link, transmitting at a frequency allocated by the Federal Communications Commission in the 900 megahertz range. This technology requires a precise "line of sight" path between the radio station and the transmitter.

Sometimes, this "line of sight" path is not easily obtained. Natural and man-made obstacles exist, especially in metropolitan areas. Also, metropolitan areas crowd the available frequencies in the FCC allocated spectrum, limiting the availability of frequencies or causing interference when frequencies are closely spaced. In these cases, a terrestrial wire path may be only alternative to establish this STL link.

Telephone companies provide low cost analog circuits for the transmission of quality audio between studio and transmitter. There are a number of problems with these circuits. Most critical is the telephone company movement away from analog technology and towards digital technology. The effect on the broadcaster can be severe. These analog circuits require constant attention from skilled central office technicians. But no new technicians are being trained in analog network maintenance, as the telephone companies focus their energies on digital network technology. As these circuits become more expensive to maintain, there is an inevitable increase in tariff charges. New circuits are exorbitant to install, and discouraged.

Contrast this with T1. Charges are dropping dramatically, the carrying capacity of a single T1 can combine voice, data and multiple audio channels, and T1 is a full duplex service, allowing transmitter to station traffic to be combined on the same circuit. Also, the end equipment for a T1 STL is much less expensive than the corresponding equipment for a 900mHz STL.

The use of T1 for station to transmitter links is growing at an accelerating pace, as the number of stations changing or installing new T1 circuits grows. While the pricing structure of T1 compared to 900mHz microwave (allocated microwave is free currently, T1 costs something!) will insure the short term viability of microwave, T1 remains an effective alternative for 1990s STL requirements.

T1 has gained new popularity from the consolidation of the radio industry. Radio stations are being bought, integrated into a parent's station administration, and operated remotely over - you guessed it - T1 lines. Duopoly and LMA (Local Marketing Agreement) situations have advanced the reputation of T1 as an element of cost savings. Programming is created at a single site and distributed to multiple remote transmitters over T1 circuits. The cost of T1 is minuscule compared to the staff requirements of the radio station.

ON - DEMAND BROADCAST APPLICATIONS

Broadcast use of T1 digital service has received minor press coverage in recent years. In contrast, new applications that use Switched digital services have received considerable attention. The availability of new network services, combined with significant advances in compression technology, have spurred great interest in applications that can be optimized using this technology.

Earlier we stated that all digital services were outgrowths of T1 technology. Switched digital services that are important to the broadcaster include Switched 56 kilobit service and Basic Rate ISDN.

Switched 56K service uses a single 64 kilobit "timeslot" which is the basis of T1, described earlier. It is a derivative of T1 and, in many cases, travels over actual T1 circuits in the network. The 8 kilobit difference between 56 and 64 is used for dialing and control information. Switched 56K is widely deployed and available in the United States.

Basic Rate ISDN combines two 64 kilobit timeslots with a 16 kilobit dialing and control channel. So, dial - up 128 kilobits is possible over a single circuit. Basic Rate ISDN is widely available in Europe and Asia, but not as prevalent in the United States. These two technologies are different, but serve the same purpose for the broadcaster. Switched Digital Services, as referenced in the paper, refers to both of these services.

Compression of audio signals can provide quite good fidelity at rates of 56K, 112K (two 56K circuits) and 128K. At 56K, compression provides acceptable transmission of 7.5kHz audio, and a number of "talk radio" programming currently occurs on a dial up basis using a compressed 7.5kHz signal over a Switched 56K circuit. Anyone listening to an NBA basketball game on the radio today is listening to an announcer at the sports event forum connected to the radio station over a Switched 56K circuit. Since the station only pays for the circuit during the transmission of the sports event (plus the cost of the Switched 56K line, like your home telephone charge), significant savings accrue compared to a dedicated circuit between sports forum and station.

CLOSING REMARKS

The transmission of quality audio over high speed digital circuits, unheard of ten years ago, is now an important tool used by the broadcast industry. Broadcast distribution networks, international public radio, STLs, Duopolies, LMAs, and dial up audio are a few of the many current uses of digital circuits transmitting audio signals. Users achieve important advantages in performance and cost compared to alternatives, and so the use of digital transmission of audio is expected to continue its dramatic growth.

DIGITAL TRANSMISSION ALTERNATIVES AND APPLICATIONS FOR AUDIO

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Techniques for digital transmission of high quality audio have been in commercial use for approximately ten years. The transmission medium itself, once predominantly on satellites, is now moving increasingly to the ground. This paper will relate the experiences of a large provider of such services, highlighting capabilities of the various emerging technologies for different service applications.

IDB Communications Group recently celebrated its ten year anniversary. Along the way, we have been at the cutting edge of numerous technologies for the digital transmission of audio. Each of these technologies has offered new service applications, but not without challenge. Like any area of broadcasting, the rewards of being one of the first with a new tool are frequently hard-earned with equal amounts of hair-pulling to make it work. Digital transmission of audio is no exception. In some respects, the job has become more difficult since we can no longer say with the same simplicity of analog "it was fine leaving here."

Our own digital experience at IDB goes back to 1985, when we joined GE as one of only two providers of Scientific Atlanta DATS service for distribution of radio programming via satellite. Using approximately 384 kb per 15 KHz channel, DATS, or Digital Audio Transmission Service, became the standard of the industry for digital program distribution.

IDB was also one of the pioneers in using specialized Intraplex digital multiplexers for audio transmission. Like DATS, this equipment was also relatively inefficient, using 384 kb per 15 KHz channel, or 192 kb per 7.5 KHz channel. We originated from the Reagan/Gorbachev Summit in Reykjavik, Iceland in 1986 using Intraplex equipment, and then again for a Billy Joel remote for HBO from Leningrad in 1987. We also utilized 16-bit Sony pulse code modulation equipment in 1987 to link up Los Angeles with New York via satellite and fiber for an interactive Stevie Wonder recording session.

The limitation with all these early digital technologies was their excessive bandwidth requirements. Satellites and fiber optics are essentially commodities; the price of any transmission service is largely determined by how much of that commodity is required to do the job. DATS and the Sony PCM system each required a full satellite transponder for operation, which eliminated all but major multi-channel network users or those with very deep pockets. DATS also required that all signals be multiplexed together at a central uplink site, adding backhaul costs to the overall routing. The original Intraplex equipment required a full T-1 circuit, also exacting a high price of entry that limited its service application.

Partial improvements were realized by IDB in 1991 as we moved to second-generation Intraplex equipment employing the new APT x-100 algorithm, created by Audio Processing Technology of Northern Ireland. Using 128 kb per 15 KHz channel or 64 kb per 7.5 KHz channel, the algorithm tripled the number of channels that could be derived from a T-1 circuit. We made good use of this increased capacity in building dedicated digital audio circuits between our facilities in New York, London, Frankfurt and Moscow. However, as a carrier, our offering of such services was still limited to cases where we could generate enough ongoing demand from customers to warrant the costs of the full T-1 bandwidth the system required.

Simultaneously, we were looking for a replacement for our aging analog SCPC satellite system linking 35 cities in North America. With the commodity of fiber optic bandwidth becoming less and less expensive, we initially investigated linking the same cities with Intraplex equipment using the APT x-100 algorithm and dedicated T-1 circuits. The major carriers responded by telling us that the proposed network would be second in size only to that of the federal government. Not meeting our cost requirements for the underlying fiber and required digital cross-connect switches, we declined their offers and resumed our search for low cost digital bandwidth suitable for audio transport.

We next worked with Wiltel in Tulsa, Oklahoma. They had recently inaugurated a "bandwidth on demand" service, using part of their fiber network for a major oil company that wanted to teleconference using sub-T-1 digital video codecs. Our concept extended the use of this bandwidth on demand to audio transmission utilizing second generation Intraplex equipment. This system finally allowed us to buy and use only the specific amount of bandwidth that a given job required. No longer would we have to fill up a full terrestrial T-1 with customers to provide cost effective digital transmission. We replaced satellite equipment in some locations on our network, and began transmitting with what is now called "fractional T-1 bandwidth on demand" from Emerald Studios in Nashville to

Los Angeles.

However, three problems remained with this concept: First, while fractional T-1 service was increasingly available from interexchange carriers (ATT, MCI and Wiltel, for example), the local phone companies supplying the digital first and last mile circuits still required full T-1 access with rare exception. Unfortunately, that is still the case, making an otherwise good technology more difficult to cost justify. Secondly, the interexchange "fractional bandwidth on demand" itself commanded a premium rate which increased our costs. Lastly, such "bandwidth on demand" systems required reservations prior to use, which was an awkward match for some broadcast applications. These three elements, which continue to this day, made our use of such technology limited.

As it became more and more apparent that T-1 delivery would not be a viable option for our primary needs, we were simultaneously becoming more and more familiar with standalone codec technology at low digital rates. IDB used the Corporate Computer Systems Micro-56 codec for a fulltime digital satellite link between Tokyo and Los Angeles in 1991. The equipment provided 7.5 KHz frequency response in only 56 kb, and became a mainstay of the industry. The Micro-56 and similar products from other manufacturers such as Comrex, remain in widespread use today.

Our exposure to the CCS Micro-56 also led us to examine the economics of switched-56 and basic rate ISDN service as an alternative to fractional T-1 bandwidth on demand for transport. We determined that the cost of switched-56 or ISDN might be as little as 20% of similar fractional-T-1 service on demand, and so, concentrated on switched-56 and ISDN as the platform for our analog satellite replacement.

As IDB was utilizing such 7.5 KHz equipment, though, our vendor alternatives for extended frequency response codecs were limited. In mid 1991, IDB became one of the initial recipients of the CCS CDQ-2000, a new codec using the Musicam algorithm. Providing a mono 20 KHz channel or "joint-stereo" 20 KHz capability using the same 112 kb, the CDQ-2000 was put into use by IDB on links between our New York and Los Angeles audio control centers. We next considered deploying this product in conjunction with switched-56 or ISDN to replace our original analog SCPC satellite network. IDB was unable to do so, due to three factors. First, some customers objected to the "joint-stereo" or "stereo dependent" coding used in the CDQ-2000, desiring discrete stereo instead. Whether for reasons imagined or real, as the service provider, we had to respect their wishes. Secondly, some of the features of the equipment were overkill for our applications, driving its price beyond our consideration. The greater problem for IDB, however, was that while the unit provided stereo capability at 112 kb, it offered no economies to transmit only a single 20 KHz channel, a feature we felt necessary to field the equipment in a wide variety of applications as our satellite replacement. We also needed the ability to feed dual mono simultaneously to two separate locations using the same stereo codec operating in a mono mode. This was not the CDQ-2000's forte. However, the unit found widespread use for others in the industry, given its ability to provide stereo transmission using as little as two switched-56 circuits or one basic rate "2B+D" ISDN service.

As our experience with switched digital products grew, we saw distinct needs for two different types of customers. One customer was driven by cost while needing to maintain acceptable audio quality. This was typically the radio broadcaster seeking to do remotes under limited budgets. Therefore, optimum digital efficiency of the algorithm was critical along with low cost equipment. The second type of customer was interested in digital technology, but not at the expense of compromising any audio quality. These customers were typically recording studios seeking high quality interconnections with added features such as AES/EBU digital interfaces and timecode transport capabilities along with the audio. The market had essentially separated into two classifications: broadcasters with straightforward 7.5 KHz or 15 KHz requirements needing to cut costs of transmission, versus audiophiles needing extra features and optimal sound quality, albeit at the expense of using more digital bandwidth. After spending an agonizing year working with a manufacturer trying to develop a "one size fits all" codec, we abandoned that approach in favor of responding with two different systems for the two different customer types.

For the quality conscious recording studios, IDB worked again with Audio Processing Technology to develop the "3D2" codec. Providing direct dial convenience with digital quality for two channels, this codec employs the proven APT x-100 algorithm we were already comfortable with from our second generation Intraplex equipment. Instead of T-1 or fractional T-1 circuits, however, the 3D2 uses as many as six separate switched-56 or 3 basic rate ISDN lines to provide flat frequency response out to 20 KHz while operating in discrete stereo mode with minimal digital compression. IDB currently has over forty recording studios on line with the 3D2. These studios appreciate the quality of the sound the 3D2 provides, along with timecode, digital audio interfaces, and the user's choice of sampling rates. Efficiency of the algorithm is a secondary consideration to quality for these users.

The other customer type, however, the broadcaster, is interested in basic digital service to the extent that it will lower their operating costs. Therefore, the equipment chosen must be relatively inexpensive, and provide the desired audio frequency response using as little digital bandwidth as possible. For this demanding application, IDB chose the Telos Zephyr in 1993 for widespread use. Placed directly at popular broadcasting venues, the Telos unit incorporating the MPEG Layer III algorithm can operate in stereo, dual mono, or single mono modes. The building blocks are discrete channels of 15 KHz each at 56 kb, which can be utilized as needed to generate stereo using 112 kb overall. In mono mode, each 15 KHz channel uses only 56 kb. In dual mono mode, each channel can communicate with a different location than the other discrete mono channel. This allows IDB to use the same equipment base to accommodate a single channel mono broadcast for one feed, a dual language sports broadcast for another feed, and a stereo broadcast for yet another feed, all using a common equipment base. Because the underlying network is basic rate ISDN wherever available, or switched-56 where not, we retain the network economies and dial-up convenience while delivering the quality demanded by our customers. IDB was the first user of the Telos Zephyr, deploying it for NHL Hockey backhauls to rightsholders. We are now using it as the core technology to replace our analog SCPC satellite network. During this year, IDB will field the Telos equipment in over 35 cities as our primary means of backhauling signals from point to point.

So the marketplace has now generated a wide variety of codecs from which broadcasters can choose, ranging from low-cost 7.5 KHz mono devices up to audiophile-quality boxes loaded with extra features. Similarly, users can choose the digital superhighway upon which they wish to ride. The economics of switched-56 and basic rate ISDN are compelling for occasional use; however, higher digital rate applications or fulltime users may reasonably opt for "nailed up" T-1 or fractional T-1 circuits for their wider bandwidth and higher performance availability. For instance, a broadcaster may want to use dial-up ISDN for low cost remotes, while utilizing dedicated T-1 circuits for a studio-to-transmitter link. Similarly, inexpensive codecs may serve the occasional remote requirements well, while a more expensive shelf of Dolby or Intraplex equipment is used fulltime for the STL.

During our years of use with ISDN and switched-56, IDB has learned a few lessons that may help broadcasters now exploring their options:

Firstly, switched-56 and ISDN offer attractive rates, but both services have still not matured to the simplicity and availability of voice-grade dial tone. Be prepared to work closely with your local phone company, as well as your chosen interexchange carrier. Your switched digital circuits may be the first ones with which the local phone company has ever worked. An alternative to ease your burden is to work with a no cost travel-agent equivalent of switched digital circuits, such as Global Digital Datacom Services.

Be prepared for everything to take longer than you imagined, and brace yourself for world-class finger pointing should problems arise in the handoff between your local phone company and the interexchange carrier.

The most reliable services in the switched-56/ISDN realm are those known as "DDS" or direct digital service, which involve a dedicated circuit between your premises and the interexchange carrier. These bypass the local phone carrier's central office and switching, which minimizes interface problems. However, this type of circuit is more costly, and effectively "marries" you to the interexchange carrier. The interexchange carrier can make all of the arrangements with the local phone company for your DDS service for an extra fee, which may save you time and personal involvement in any future service problems.

The alternative is an arrangement known as "DSA" for digital, switched access. This circuit is routed through the local phone company's central office switching equipment to the interexchange carrier of your choice. Advantages of this are its lower cost as compared to DDS, as well as the ability to change interexchange carriers much as you would for voice service. Additionally, while you can theoretically receive calls from different interexchange services using DSA access, from IDB's experience, we would highly recommend picking and staying with one interexchange carrier wherever possible. DSA access requires that the user become more involved in network design and troubleshooting, but offers good rates for those willing to do so.

Using one interexchange carrier will minimize intermittent timing and sync problems and phantom hangups, but switched digital service will still require time and attention from the user.

If you are contemplating the use of switched digital circuits for one-time only remote use, plan to arrive as far ahead of the remote broadcast as possible to ring out the circuits. Coming in just before air time will put you within the jurisdiction of Murphy's Rule: anything that can go wrong will go wrong.

Where numerous digital audio systems converge, the user must beware. Otherwise, the cumulative effects of various compression algorithms can wreak havoc with your source material. IDB's experience is not unique nor is our advice complex: we recommend compressing as little as possible, and using as few systems as possible to get the job done. If multiple compression systems must be utilized, operate each of them at the highest digital rate possible so as to minimize the rate of digital compression. Since there are only limited standards for interoperability between various digital systems, the user must be prepared to spend a fair amount of time to debug the system. Remaining flexible is important, as some combinations you try may not produce desired results.

Many things stand between broadcasters and the ideal of switched digital service. Reliability and availability are the primary obstacles at this time. ISDN, though written about for years, is still only sporadically available throughout the U.S. Where it is available, it generally offers cost advantages over switched 56 service. It would also be nice to have features in switched digital akin to that of voice service, including collect calls, 800 numbers, 900 numbers, the ability to receive calls from anyone, regardless of their own interexchange carrier affiliation, and reliable alternate interexchange carrier access by first dialing a 5 digit code as is routinely done for voice service.

Nevertheless, the advent of low cost codecs and reasonably priced digital bandwidth has changed the landscape of audio transmission for the better. Remotes are now possible at a lower cost than ever before. While this brings new challenges to transmission service companies like IDB, it also widens our prospective base of customers. As point to point audio applications are moving more and more to the ground, new digital technology is also lessening the cost of high quality distribution via satellite in a similar manner.

In the first quarter of 1992, IDB began conversion of its DATS system to SEDAT, Scientific Atlanta's "spectrum efficient digital audio transmission" system. SEDAT employs a new algorithm that yields three times the number of old DATS channels in the same amount of bandwidth, while increasing the upper frequency response from 15 KHz to 20 KHz. SEDAT is now the predominant delivery method for high quality distribution of digital audio with thousands of radio stations equipped with SEDAT format receivers. Satellites still offer their inherent cost advantage of distance insensitivity as compared to fiber for widespread point to multipoint applications.

When contemplating digital transmission of audio signals, the choices that a user must make can be broadly divided into two groupings: point-to-point requirements versus point-to-multipoint. Point-to-point applications over switched digital lines can now be served using low cost equipment such as the CCS Micro 56 and 66, or similar G.722 algorithm units from Comrex and others. These will all provide 7.5 KHz mono audio in 56 kb with some degree of interoperability between manufacturers. If enhanced frequency response or stereo is needed, the user can select other products such as the Telos Zephyr or choose from manufacturers offering Musicam implementations, such as the CCS CDQ-1000 or CDQ-2000, or the RE America 660/661. For critical audio needs along with timecode and digital interface extras, the APT codec offers a good choice.

The next step up involves use of full T-1 or fractional T-1 leased lines, or using multiple switched-56 or ISDN dialup lines in conjunction with an external inverse multiplexer. Such configurations will allow the user to deploy higher rate digital systems for point-to-point applications, including Intraplex channel banks, Dolby AC-2, or some of the variable rate Musicam system mentioned previously operating at 192 kb or beyond.

Point-to-multipoint digital distribution can be handled by several systems as well. The CCS CDQ-2000 in conjunction with Comstream RF equipment can offer a relatively low cost digital solution for new networks. Two other choices in this area are the IDC and Wegener Musicam-based systems. All of these provide good quality, and can be deployed on a single channel basis from a customer's premises utilizing Ku uplink frequencies. SEDAT remains the C-band digital distribution choice for major networks, having an established base of thousands of receiving radio stations. However, unlike CCS/Comstream, IDC and Wegener, SEDAT users must aggregate their signals at a SEDAT service provider: IDB or ABC.

The digital superhighway now encompasses both the ground and the sky with fiber and satellites both being pressed into service for their unique attributes. The highway itself, however, is still a work in progress with inadequate directions, specifically where switched digital technology is utilized. Chances are good that nobody has ever tried to go quite where you need to. But for the user who is accepting of the challenge, the new technologies with their lower implementation costs will open many doors for programming that never before existed. The same process is currently underway with video, the primary difference being the higher order of magnitude in bandwidth requirements for video as compared to audio. As ISDN, fractional T-1 and switched 56 have brought alternatives to audio transmission, so will digital compression and ATM, or "asynchronous transfer mode" do for video in the next few years.

In the face of increasingly tight operating budgets, the use of new generation digital equipment can serve the needs of broadcasters well. For those that can afford to spend the time and learn by trial and error, new network possibilities are yours. And for the rest that just want to avoid the hassles, get the job done easily and professionally with a phone call, transmission service companies will continue to serve your needs using new efficient digital technology, both on satellites and on the ground.

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DIGITAL AUDIO WORKSTATIONS AND STORAGE

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CONFESSIONS OF A DIGITAL AUDIO WORKSTATION USER

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Abstract

Three years of working with the same digital audio workstation and evaluating numerous others has totally changed my approach to audio production. This paper details those changes, and offers observations for those who would follow.

Introduction

I began doing production with a digital Audio Workstation (DAW) in September of 1990. Since then, I've learned that an optimized hard drive is a happy hard drive. I know that a quangst RAM card can ruin your whole day. I know that one of the first things you do after buying a any computer-assisted workstation is to buy the fastest, cleanest, meanest power line conditioner and spike protector you can afford. If it's not as fast as a Tripp-Lite Isobar, you probably don't want it. If you can afford an uninterruptable power supply with enough capacity to let you close down the computer safely, buy it.

I also know that using a Digital Audio Workstation has changed the way I do production...and I have survived the transition. In fact, I now find that the thought of doing production the in the linear analog domain is difficult, cumbersome, inflexible and noisy.

Choosing which workstation best met my needs three years ago was easier because, then, there were only about a dozen systems on the market. There are now over 52 different devices calling themselves Digital Audio Workstations.

Before we get to answering some of the questions about digital audio workstations, let first consider the benefits of digital audio itself.

Benefits of Digital Audio

About a year after I had been using AKG's DSE-7000 DAW, I had a session which required that I use some audio that had been recorded on 1/4" analog tape. As I punched up the reel-to-reel machine I thought "What the heck is that noise?". It was simply analog tape hiss. The same level of hiss that I had learned to ignore for almost 20 years. However, now it was obvious and objectionable. If you're going to get a DAW, you almost have to get a DAT, or some other digital system to master on, unless you plan to dub directly from the DAW to whatever you're using on the air.

Is analog obsolete? No. But first let's make sure to distinguish between analog on a wire, and analog on tape. I doubt there are more than one or two people in this room who could tell the difference between a digital master and the same master played back through a set of good A/D and D/A converters.

So analog itself is not the problem; analog tape is. And, since most radio production people have learned to hit the tape pretty hard with the audio, with additional limiting and compression, there are few if any quiet sections during which any tape hiss can be heard. For those situations, there are a variety of analog noise reduction schemes that will keep analog from becoming obsolete.

There are even a few situations in which analog recording may be better than digital. Most engineers still prefer analog for sounds that have very high or unpredictable transients such as gun shots or other percussive events. Even though Apogee and others now have digital signal processors that simulate the effects of analog tape compression, many audio people still prefer analog. Even though analog tape compression is not linear, it is what we have become used to.

For Prospective Buyers

Some basic questions to ask concerning DAWs. (Honest answers will result in better decisions.)

Question#1 How quickly can you learn to use the system? This is actually a double-edged question. To find the answer, you first need to commit to spending the time to learn. Do you have the time? Can you make the time? If you are now not using any kind of computer, and would prefer not to, it's probably wiser to remain in a linear multi-track environment. There are enough digital tape and cart configurations on the market to replace your existing analog devices. However, if the lure of digital non-linear editing remains strong, as it did with me, proceed. WARNING! There are no short-cuts. You will have to deal with some level of computer knowledge. There is no substitute for your own experience. Disparaging comments from others about a particular system, or claims of its super abilities must be taken with a grain of salt. Why? Because their experience may be based on the last version of the software, not the current version.

Question#2 How many people at your facility will be using the system? Although you may have mastered your fear of computers, being the only one in the studio who uses the system will decrease the ability of the system to pay for itself. If your least-experienced person can't quickly be trained to do a simple voice-over on the the system, it's a bad sign.

Question#3a Where do you plan to put the computer and screen that are usually part of the system? The most obvious place for the computer monitor is directly between the audio monitors. But be aware that the emissions from poorly shielded video monitors can get into your audio, causing a rather nasty buzz. Dynamic mics make great noise transducers when placed too close to some video monitors. Usually, though there's a dead spot directly in front of the monitor where noise pickup is minimal. RE-27N/Ds do pretty well, Sennheiser 421s do not.

Pre-existing "space hogging" production consoles can also present a problem. Keyboards, mouses (mice?) and proprietary control panels usually must be positioned to co-exist with parts of your existing setup. Some workstations internalize a lot of channels, which means your console may not have to be as big. Consider a smaller mixer. If you don't already have the space and if "floating" wall, counter-top and ceiling mounted monitor and computer stands don't work, you may have to redesign the room.

Question#3b How do you deal with the added noise these components create. Computer fans and disk drives make noise. If you have a "combo" studio (where the machines and mics are in the same room) find out how far away the CPU and hard drives can be stored. Make sure the space is well-ventilated and dust-free.

Question#4 Can you actually work faster, or does loading in, processing a mix, and backing it up offset any time you might have saved?

Question#5 Can you raise your studio rate or attract new business based on enhanced services and time savings to the client, or will the competitive atmosphere in your market force you to eat the cost of the system just to stay even? Unless you can sell your client on the benefit of the system, you can forget the rate hike. I was able to justify a rate hike with one client who normally spent two to three hours a session doing radio spots. Now he comes in, lays the voice tracks, hands me the music elements and walks out in less than an hour. The one or two hours of time he gains by not having to be in the studio, plus the reduced time it takes me to edit the spots saves time and money for both of us.

Question#6 If a lot of your work comes in from other studios, how compatible with other systems is the one you have in mind? Does it really matter? These are problems which are still being worked out in the industry. It appears as if Avid's Open Media Framework (OMF) may provide an answer. Alesis and Fostex's ADATs and Tascam's DA-88 eight-track linear digital formats offer other solutions. Incidentally, these digital tape formats also offer markedly lower cost and reduced storage space compared to analog tape. I also buy a lot less 1/4" stock.

Question#7 Have you bothered to figure out how many more sessions each week you need to pay for the system? If you've never bought a big piece of gear (over $20K), it may be difficult for you to think about reaching that deep into your pocket. Especially if your used to writing off all your acquisitions in the same tax year. Talk to your accountant about depreciation schedules.

Question#8 How quickly will the manufacturer respond when the system goes down? Next-day-air is about as good as it gets unless the dealer or manufacturer is across town. If you can't get a guarantee of a board or component swap any quicker than that, it's a problem.

Benefits of DAWs

Initally, the most obvious advantages are things like the ease with which different versions of a spot can be made. Accomodating the last minute whims of clients and experimentation by yourself are others. If you currently use analog gain reduction, you'll also probably go through a period of re-evaluation. At first, many users get even more aggressive with their processing in an attempt to simulate the thickening effect that analog tape compression used to provide. Later, many begin to use less processing, using some compression but mostly limiting just to keep the occasional peaks out of the red.

With much of the analog noise out of the way, you may start experimenting with tube microphones and preamps. Without tape hiss, their subtleties will become more apparent.

After you have lived with the system for a while, additional options begin to occur to you. These new options are the direct result of how well you integrate with the functions of the workstation itself. If the system has real-time faders for each channel, and a screen that shows you where the audio is on a track and how it changes in loudness, you can mix more quickly and effectively.

The ability to move audio around with exacting precision also increases the impact of your work. You don't need to use as many effects to "fix things in the mix". Being able to bounce mix to infinity without noise is another advantage.

If you're really retentive and do a lot of micro-editing; removing tongue clicks, taking the breaths out of voice tracks, looping or resizing music tracks, your work will be a lot easier. When you screw up you just take a breath, do a pick-up and keep going.

Time Loss

Mix processing, archiving, back-up, overly-retentive or experimental editing and systems that require a lot of page jumping result in a loss of productivity. So do data management tasks like removing audio that you don't need anymore from the hard drive and optimizing the hard drive. In systems where DOS, MAC and other sensitive files are vulnerable, you will probably have to spend some time doing reconstructive file management when someone accidentally misplaces a file. The results of which vary from not being able to access a particular element to locking up the system. Add about an hour a week to your schedule for these duties and you begin to eat away at all that time saving these systems supposedly have.

The Bottom Line

Is it worth it? With the right system, absolutely. Also know that quite a few manufacturers have lowered their prices within the last six months. If you haven't gotten a recent price quote, you may be in for a pleasant surprise.

If you're trying to sell management on the idea of a DAW, here's the idea in a nutshell, "The DAW is to audio production what a word processor is to typing."

Thanks for your time and attention. If you would like a copy of this presentation, you may find it in the proceedings. Or, if you're a member of a computer bulletin board, I'll supply you with a disk or e-mail it to you for uploading. My MCI Mail number is 347-6635. My phone number is (410) 889-6201. Any Questions?

SAVING THE GOODS: A SURVEY OF DIGITAL STORAGE ALTERNATIVES

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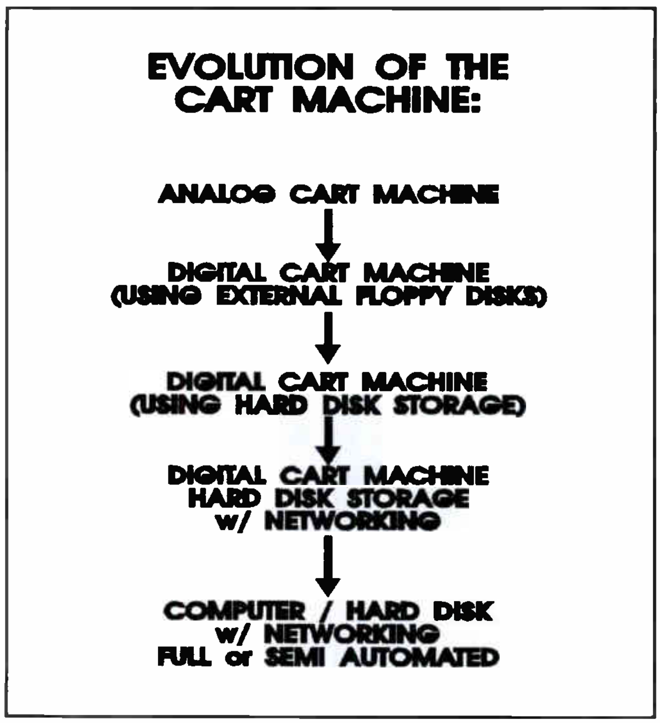
This paper provides an overview of the available digital alternatives to cart machines, and some basic tools to evaluate the selections. Analog cart machines clearly are near the end of their useful life in the broadcast facility. Fewer and fewer are being manufactured each year. New technologies will continue to evolve and replace analog carts. This paper will review the technologies available today, and explain why, when and how to make the transition to digital equipment.

The basic problem facing many broadcasters today is a studio full of aging cart machines. Does one simply replace these with new cart machines? Or is it time to advance to the next level of technology? Simply replacing the cart machines is still an option, however the opportunity presents itself to advance forward. Stations in the position of upgrading their equipment have the option to take the next step beyond analog cart machines. And the choices are overwhelming.

The most simple cart machine replacement can be found in the floppy disk-based cart machine. They function exactly like a standard cart machine, with the added benefit of improved sound quality and being maintenance free. They also have some ''smart'' features, such as automatically detecting kill dates on spots. Since they are based around a removable media, they have the same basic operational characteristics as a traditional cart machine.

The most advanced systems are the hard disk based storage systems which allow multiple studios to be networked together, and interface with both music scheduling and traffic & billing software for true ''paperless'' studio operation. Systems such as described are already in use in radio stations today. Several other options exist, and these will be examined in detail.



To properly compare different systems which are all designed to solve the same general problem, it is necessary decide on a common denominator to apply to the systems involved. To evaluate cart replacement systems, we shall use the following constraints:

1) System is only to be used for spots (i.e. commercials, IDs, PSAs, liners, news reports, weather, etc.)
2) All recording will be in stereo.
3) All recording is at 32 kHz Sampling Frequency.
4) All recordings are made using data compression (where available.)
5) If use of data compression is limited to a specific sampling frequency, that sampling frequency is used and compared to others which use 32Khz with data compression.

While it is certainly possible to use a digital cart machine replacement for playing music as well as playing commercials, it is not as cost effective as playing music from other sources. Until hard disk and high density floppy disk prices drop significantly, the cost per minute of recording on hard disk is still much higher than recording on CD or Mini Disc. Plus, CD jukeboxes are ideal for a live-assist or walkaway operation. Satellite programming also provides a very clean, efficient way of playing music.

II. OVERVIEW OF FORMATS

A digital recording device that could be considered for use as a playback system for on air broadcast can be placed into one of the following categories:

TAPE
RECORDABLE DISC
FLOPPY DISK
HYBRIDS
HARD DISK

Tape Formats include DAT and DCC (Digital Compact Cassette.) Radio Systems in Bridgeport, New Jersey has developed a DAT machine specifically for on air use with cart machine-like controls. The only DCC hardware presently available to broadcasters is the consumer grade equipment available at a local hi-fi store. At this time an ''industrial strength'' DCC machine specifically for broadcast use does not exist.

Recordable Disc formats include not only Recordable CD (which has just recently become very affordable), but also the newer MiniDisc, introduced in a broadcast version by both Sony and Denon. The on air playback device for Recordable CD is any standard CD player, whether consumer grade or professional quality. Likewise, playback of MD (MiniDisc) can be on any MD player, although choices at this time are very limited.

Floppy Disk formats include the removable media digital cart machines such as the DCR1000 series cart machines by Fidelipac and Broadcast Electronic's Disk Trak. Air Corp is developing their Air Cart which uses a 128 megabyte magneto-optical disk.

Hybrids are represented by the DigiCart™ from 360 Systems and the DDS™ (Digital Delivery System) by Radio Systems. The DigiCart™ is a cross between a floppy disk and a hard disk based system. It has the cart machine-like control surface of the floppy disk units, but uses a built in hard drive for storage. On the outside, it's a cart machine. On the inside, it's a hard disk storage system. DDS™ by Radio Systems is multi-user platform that does not require a traditional computer terminal for the operator to access spots.

Hard Disk based systems represent the fastest growing product category in the broadcast industry. The typical hard disk systems consists of a computer complete with color monitor, keyboard and mouse, a hard drive, sound cards and operating software. With over 20 different manufacturers offering some type of hard disk system, the choices to the broadcaster are many. This category is definitely the most difficult to evaluate due to the vast variety of choices available. Many systems also have production and editing software built in. The equipment described in these categories represent equipment which is available or in development as of December 1993.

III. WHY SWITCH TO DIGITAL?

This is a good question. If a station has a studio full of decent quality cart machines that are less than 10 years old, is it really worthwhile to switch over to digital? What is the real motivation for unloading all the studio's analog cart machines? What are the goals in doing so? Let's consider some of the possibilities:

1) Improve the sound quality of your broadcast signal. Consumers have become acutely accustomed to the sound of CDs. They are listening to CDs at home. They are listening to the CDs on the radio. Their ears are gradually becoming more finicky and demanding. The last thing you or your advertisers want is for your listeners to mentally "tune out" anytime a commercial airs. Digitally recorded commercials will match the sonic quality of the CDs you're playing, and give your station a more consistent sound.

2) Reduce the physical storage & space requirements. With the increasing number of LMAs and duopolies, more stations are finding they now have to squeeze two or three stations into the same space originally occupied by one. More than ever, space is a premium. Analog carts take up incredible amounts of space. Hard disk systems allow much smaller rooms to be used as ON AIR studios, with out compromising quality. Cart storage is no longer an issue, since all carts are stored are now stored as files inside a hard drive.

3) Simplify On Air Operation. Analog cart machines couldn't possibly get any simpler to operate, could they? Can we possibly take something as basic as an analog cart machine, and actually make it easier to operate? The answer is yes. The hardware and software replacing cart machines is now capable of handling some of the decision making previously handled by the on air staff. Digital cart machines can now automatically read a kill date and prevent a spot from airing. Most systems can automatically begin recording when they detect audio, for perfectly cued spots. Some hard disk systems will intelligently look at a schedule created by a third party traffic and billing program, and automatically program a day or week's worth of breaks.

4) Simplify Maintenance. It can cost up to $3000 in parts and labor to keep a single analog cart machine operating and maintained for 5 years. This includes replacing heads, rollers, and alignments. With digital systems, maintenance is nearly eliminated altogether. Some systems have no maintenance requirements at all.

5) Improve reliability. The reliability concern might just be the one reason many stations have refrained from getting into the newer digital equipment. Relatively speaking, this technology is new. Most of the systems are first generation, although there are a handful of 2nd and 3rd generation systems available. This entire product category, digital storage systems, is at its infancy. The good news is systems will become more and more reliable and less "buggy" as the software matures. Also, systems which use removable media (floppy disk cart machines) don't have the reliability concerns and the "all your eggs in one basket" syndrome that the hard disk systems have.

6) Optimize cost / performance ratio. Spend less money to accomplish a given task. A simple comparison: Analog carts have an approximate life span of 500 to 1000 plays before they are

replaced (or sent off for rewinding) at a cost of about $5 each. Digital "carts" have a life span of a few million uses before they need to be replaced. Add that up over a few years time.

7) Allow Networking between On Air Studios, Production, Traffic, and Billing. This capability simply doesn't exist at all with analog systems. Not only will radio stations in the future be tapeless, they will also become paperless. Most of the commonly used traffic and billing software packages in use in radio today can be integrated into a hard disk storage system, such that traffic information is imported into the software of the storage system, and each days stopsets are automatically assembled. This can be accomplished in a very short time with a few simple keystrokes on a computer.

8) Savings of new digital system versus. new analog system. Are the digital systems really cheaper than new analog systems? Or in the long run, is it cheaper to stick with analog? Upon close comparison of costs of analog carts over time versus digital, you will find the digital systems are less expensive than most of the better quality analog machines.

9) Convenience. Computers are everywhere, and that mass appeal has resulted in competitive pricing, readily available parts and accessories, and competent service. Maybe in the future it will be possible to pick up replacement disk drives at a 24-hour discount computer warehouse.

10) Lack of Handling. Because hard disk formats eliminate handling altogether, your air staff are freed up to spend more time taking phone calls and concentrating on their program. There are no more carts to pull, sort, stack, load, play, unload, and put away. And no more carts to accidently get lost or damaged. Of course, floppy disk based cart machines don't enjoy this benefit.

11) Automation. For the live station wishing to make a switch to satellite automation, hard disk systems are the perfect choice. Two general types of automation are Live Assist and Walkaway Automation. With Live Assist, all the breaks are pre-programmed and pre-sequenced. The jock presses the ON button on the console at the beginning of a break. (The break is started by a manual trigger.) With Walkaway Automation, all the breaks are pre-programmed and pre-sequenced, and are triggered by either another machine (perhaps a satellite relay, CD or tape machine EOM), or are time driven.

Digital systems allow the opportunity to rethink how tasks are accomplished in the air studio. They allow a chance to get away from operational pattern which evolved from the use of analog systems. For example, analog carts have generally been recorded with one spot per cart. If the on air operator needs to play five spots, he has to pull, load, unload and put away five carts. Quite a bit of mechanics involved. Digital systems don't require the same level of work. To optimize the cost-per-unit-of-storage ratio for the floppy disk and recordable disc systems, it is necessary to revise the "one event per cart" rule. For example, a whole hour's worth of sixty second spots can be stored on a single $15 MiniDisc, or ten recorded sixties on a $8.75 floppy disk. The cost of storage per sixty second spot becomes small change (compared to approx. $5.00 per analog cart). The loading/unloading time is reduced by a factor of 10. Thus, taking advantage of loading up multiple spots per digital cart allows greatest financial and physical efficiency.

IV. OBJECTIVES

Here are a few items to think about when considering systems to replace your cart machines.

1. User Friendliness. One of the greatest benefits from most of the digital systems is they

take the process of constantly locating, sorting, loading, and putting away carts, and turn this into a one button operation. There is a great range in the user friendliness of digital systems. The floppy disk cart machines are by far the easiest to use - virtually no training is required for the air staff, since they are used exactly like analog cart machines. Hard disk based systems, despite their complexity, can also offer simple, one button operation. Determining the skill level of your operators will help determine which format can be used at your station. If your station is 100% automated and on air staff never touch the playback system, then user friendliness is not such an important issue.

2. Cost of Hardware. Costs vary according to which "format" (as described above) you decide on. The greatest range in price is found in hard disk systems, ranging from around $6500 up to $70,000 depending on number of studios, storage time, simultaneous playback channels, and other things.

3. Cost of Storage Media. It is very important to compare the cost of storage media by putting the cost into meaningful terms. The easiest way to evaluate the storage costs is in terms of dollars-per-minute of storage. Many manufacturers will tell you "600 Meg is included with the basic package" or "1.2 Gig is an additional $2000". This information is meaningless unless you know how many minutes of recording time you get. The manufacturer will need to know what sampling frequency you prefer to use, stereo or mono, and compressed or linear, before they can give you the number of minutes of recording time. Translate this into cost per minute. When comparing the costs of similar systems, make sure you compare in terms of the same sampling frequency, recording mode, and data compression. In a few cases, some systems are only available in one combination of sampling frequency and data compression. For example, the MiniDisc format is only available using a 44.1kHz, data compressed recording mode. Likewise the DigiCart_ only offers data compression when recording at 48 kHz sampling frequency. All other sampling frequencies are recorded linear.

4. Sound Quality. With digital recording, audio bandwith is a function of the sampling frequency used. The resulting bandwidth is roughly half the sampling frequency. 32 kHz Sampling yields a bandwidth of 15 kHz audio, typical FM broadcast quality. The Bit Rate (or number of bits in the Analog to Digital Convertors) used ultimately affects the dynamic range of the recording. 16 Bit systems seem to be a standard among most of the systems available today. Also, not all data compression schemes sound the same.

5. Ease of Back-Up. One of the concerns of hard disk systems is placing "all your eggs in one basket". If your hard disk crashes (and you don't have a backup system) you will instantly be losing income (in the form of spots scheduled to air) which can not be regained. Thus, you must have a suitable means for back-up of your spot library. The easiest method is to purchase a second hard disk, and keep a copy of everything in your library on it. Each manufacturer handles this problem slightly differently

6. Editing Capability. All digital systems (except tape) offer the ease and flexibility of editing. Editing capabilities on computer based systems tend to be easier to use than those in the floppy disk systems, simply because of the availability of waveform viewing and other visual cues. With most digital systems, editing is nondestructive. If you don't like an edit, you can quickly and easily revert back to the original recording.

7. Random Access. One of the greatest benefits of digital is random access. Audio is instantaneously loaded and cued. There is no

waiting for tape to rewind, and no endless fast forwarding in search of a music bed or sound effect on tape. The benefit gained by random access is time. The time savings will literally add up to months of time saved per year. Keep in mind tape based digital storage systems (DAT or DCC) do not offer random access.

8. Compatibility with Air Studio. Does this equipment interface with your broadcast console? What type of audio connectors does it have? Does it have console remote functions? Does it fit in with the rest of the equipment in the air studio? Most equipment designed specifically for broadcast use will interface properly to your existing equipment. Use caution, however, when selecting equipment designed for consumer use.

9. Automation Ready. Even if your station is not using any automation now, it makes sense to select a system with the potential to be automated. After all, you never know what the future may bring. Automation capability provides good insurance against future changes. Many systems are perfectly friendly in the manual operation mode, and switch into live-assist or full automation with no trouble at all. Other systems are strictly for manual control, and cannot be used in an automated environment. Removable media (floppy disk) machines have limited automation potential since a human is still required to load and unload the carts.

10. To Handle or Not To Handle? Many stations prefer to have a digital format which closely emulates a cart machine. It provides the on air staff with reassuring tactile feedback which they have grown accustomed to with analog carts. The transition from playing analog carts to digital carts is nearly seamless. On the other hand, some stations prefer to have all the commercials stored in a centrally located server, allowing simultaneous access of a commercial from a number of different studios.

11. Reliability. After deciding on a particular system, find out how many are in use in the field. For how long? Try to get a user list from the manufacturer. What experiences have you heard from other stations using this system or another? It doesn't hurt to ask around. Find out if the manufacturer offers any trial period.

12. Sampling Frequency Flexibility. The trade-off with different sampling frequencies is hard disk space. Using a lower sampling frequency can help conserve pricey hard disk space. Of course, the sacrifice is in audio bandwidth. A compromise might be to use 32kHz for spots with music (for best audio performance), and reserve 22 or 26 kHz for voice only spots. Any system which allows sampling frequencies selected on a cut by cut basis will allow the most efficient use of hard disk space.

13. Data Compression Flexibility. Data compression ratios can almost be directly applied to the price of hard disk storage. As an example, consider two hard disk systems which both store 10 hours of audio. The hard drive in the system which records linear will cost about 4 times more than the drive in the system which uses a 4:1compression. Sending audio through multiple compression schemes within the audio chain is another concern not to disregard. After as few as 3 or 4 passes through different data compression schemes, your audio may become degraded. (Wasn't the point of digital audio to improve the quality of audio?) If your audio chain is already loaded up with devices which introduce data compression (such as digital STLs, digital remotes using Switched 56, and satellite feeds), use linear recording whenever possible. Otherwise, you may want to use compression to help keep hard disk costs down. Ultimately, hard disk prices will drop so low you will be able to afford recording everything linear. Cost of the hard drive will no longer be a concern. Systems which allow data compression flexibility offer the most

desirable cost of storage ratio with minimum sacrifices to sound quality.

14. Networking. This is the first step towards the paperless station. When your Traffic, Billing, Programming and Production services are all communicating together, record keeping becomes automated.

15. Speed. Digital systems are ultimately faster than analog. Some digital systems are faster than others. Test drive each for yourself to compare. Time the system, from selecting the cut, loading, cueing, playback and ''putting away''. Or, if the system is to be used in an automated environment, see how long it takes to set up schedules and playlists.

16. Strength/ Stability of Manufacturer. What is the history of the manufacturer? How much experience have they had specifically in broadcast? Will they still be in business five years from now? It is very difficult to predict this last question, just use your judgment and common sense. Remember the rule of survival of the fittest. The manufacturer who offers the strongest product at the most reasonable price will by default be successful - demand from the market will dictate this. The 20/80 rule will prevail - 20% of the manufacturers will provide 80% of all the systems in use. Figure out who that 20% is now if you are planning to make a long term commitment.

V. A Quick Summary of Formats and How They Score

A. Tape Formats - At a mere 13 cents per 60 second spot, DAT is by far the cheapest way to digitally record spots for on air playback. The hardware is relatively affordable, the software is inexpensive and readily available. The biggest drawback to DAT is the lack of random access. Other than the Radio Systems DAT machine, most DAT machines are not on-air friendly.

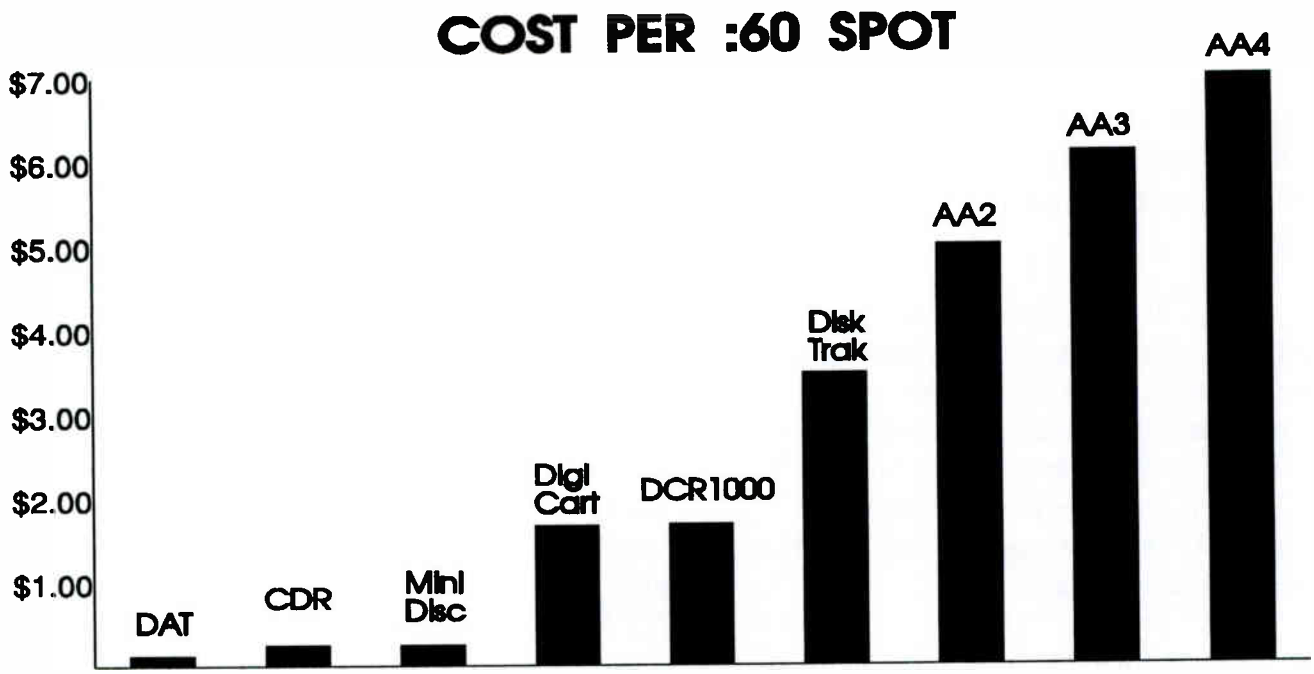
B. Recordable Optical Disc.

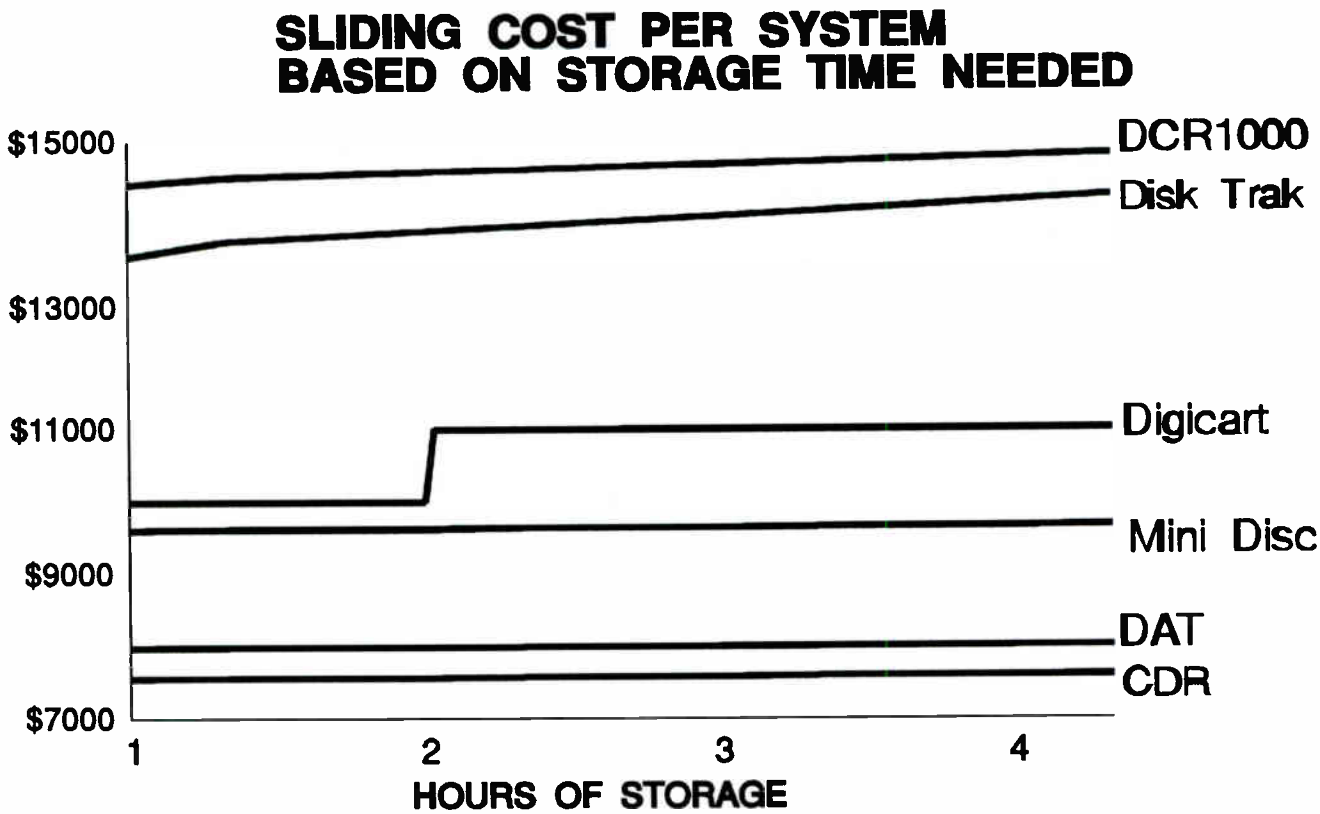
1) At 25 cents per 60 second spot, recordable CD is the least expensive format which offers linear recording, full automation capability, easy on air use, and full random access. The production studio hardware is now available for $3500 (Marantz CDR-600), about the price of a decent analog cart recorder. The playback hardware (any CD player) ranges in price from as little as $150 up to $1500. The disadvantage to recordable CD is the discs are not reusable. Once recorded, you cannot go back and erase and re-record. Thus, although the initial cost per spot is very low, the recordable disc may only be used once.

2) MiniDisc offers all the advantages of Recordable CD, plus the ability to erase and re-record spots on a disk. Spots cost 25 cents each, and the carts can be erased and reused several million times. Hardware costs are compatible to mid priced analog cart machines. MiniDisc does not offer linear recording, everything is recorded at 44.1KHz sampling using ATRAC data compression. At this time current playback hardware does not permit any automation. Playback requires complete manual operation.

C. Floppy Disk Systems. These are by far the easiest digital system to use and implement. They most closely resemble and behave like a traditional analog cart machine. At $1.70 to $3.50 per 60 second spot, the software can be a bit expensive compared to some of the other options. Both systems available today record using APT-x data compression only. Sampling frequencies are usually cut selectable. The floppy disk cart machines are the closest thing to a straight replacement for an analog cart machine.

D. Hybrids. The DigiCart™II by 360 Systems offers the best of both worlds, it's painlessly easy to use, and hardware costs are low, since only one machine is needed per air studio. Soft-


COST PER :60 SPOT
$7.00
$6.00
$5.00
$4.00
$3.00
$2.00
$1.00
DAT
CDR
Mini Disc
Digi Cart
DCR1000
Disk Trak
AA2
AA3
AA4


SLIDING COST PER SYSTEM BASED ON STORAGE TIME NEEDED
$15000
$13000
$11000
$9000
$7000
DCR1000
Disk Trak
Digicart
Mini Disc
DAT
CDR
1
2
3
4
HOURS OF STORAGE

ware prices per 60 second spot run $2.50 (Removable Audio Disk) to $6.86 (1 Gig internal hard drive, capable of holding 8 hours). The DigiCart™ will handle any mode of operation from full manual to totally automated. With additional external hard drives, one on air machine can hold up to 48 hours of stereo spots. Both sampling frequencies and data compression are cut selectable. The only drawback to the DigiCart™ is it's not networkable and cannot interface to an outside traffic and billing program. As a straight cart machine replacement, the DigiCart™ is ideal. The DDS™ by Radio Systems will be officially introduced at the 1994 NAB Convention in Las Vegas.

E. Hard Disk. These can be either very easy or very difficult to use. The user interface is strictly a function of software. Storage costs are inexpensive compared to analog carts. Although the cost per 60 second spot can range anywhere from $4 to $10, hard disk space is infinitely reusable. There are no long term costs for cart replacement or rewinding. Hard disk systems are capable of any level of automation, from full manual control to live assist, satellite, and walk-away automation. Hard disk systems also allow networking, and interface to third party traffic and billing software.

The best way to compare hard disk systems is to break them down into two categories - hardware and software. For many of the systems available, the hardware consists of a 386 or 486 PC, color monitor, keyboard, mouse or trackball. Some systems use touch screen monitors, some use a mouse. The number of Digital Signal Processing (DSP) cards determines the number of simultaneous playback channels. The hardware differences between systems varies greatly. It is important to compare the actual hardware closely when evaluating systems.

The software ultimately determines which systems are fast and easy to operate and have the most features. The graphical user interface (known as "GUI") becomes the ultimate difference in determining the user friendliness of the systems. The GUI effects the speed and ease of operation. A good GUI should consist of the following elements:

1. Pattern Recognition. Most analog cart machines generally have the same front panel layout, positioning and size of buttons. If you were to ask one of your jocks to draw a picture of a cart machine, likely he would know what a cart machine is supposed to look like. Software generated (virtual) cart machines should also follow this pattern. Is the position, spacing and colors of buttons on the monitor set up in a similar pattern to real cart machines? Are the buttons too close together? Are they in the right position relative to each other? The answers to these questions will determine which systems are easy to use and learn.

2. Anomalies. In an ON AIR studio using analog cart machines, an example of an anomaly would be if the STOP lamp is flashing on one machine but not on the other five. With a split-second glance, you can immediately tell which cart played. This method of instantly acquiring information must also follow with software based systems. When the START button has been pressed, is it easy to determine which virtual cart machine is playing? Can you tell at a glance which cart machine is loaded and which is empty?

3. Displays. The meters, timers and display information on a virtual cart machine must closely emulate that of a cart machine. Does the software-generated LED meter on the screen follow the audio in an expected way? Are there enough units of LEDs to provide meaningful information? (An example of a poor display - a LED type VU/Peak Meter with only 3 or 4 segments.) Do waveforms provide enough resolution, on both a vertical and horizontal scale to allow fast editing?

4. Action versus. Reaction. Does touching a button on a touch-screen monitor result in the appropriate reaction and expected result? Does sliding a software fader using a mouse cause the expected audio gain increase or decrease? Does the button behave in a similar way to the real thing? Does the system have the right ''feel''? For example, pressing the START button, whether by touching a screen or by clicking a mouse, must start audio instantly, just like hitting the START button on a cart machine.

The key is not to allow the hardware/software combination hinder the performance of the operator. (Don't mix up the good GUIs and the bad GUIs). All operating procedures must be accomplished effortlessly, freeing the operator to focus on his or her program, rather than the mechanics of the system. In other words, it's got to be a ''no brainer''.

VI. CONCLUSIONS

On the following page are some questions to ask about your station which will help qualify which type of digital storage system is best for you. These questions will also help manufacturers determine what system you need. The Cart Machine Replacement Survey can be used to keep track of the different systems your station evaluates. You may want to create your own survey for evaluating hard disk systems, as they could not be thoroughly covered in detail in this section.

STATION QUESTIONNAIRE

Station Type:	AM	FM	AM/FM Combo		OTHER (Describe)
How many different, simultaneous ON AIR studios?	1	2	3	4	
How many PRODUCTION studios at our facility?	1	2	3		
Other STUDIOS? (i.e. NEWS, TALK SHOW, etc.)	1	2	3		

1) Are spots to be recorded in MONO or STEREO, or BOTH?

2) Which SAMPLING FREQUENCY will be used?

3) Do we want to be able to mix and match different SAMPLING FREQUENCIES?

4) How many HOURS OF STORAGE is needed to get started?

5) Will we record spots using DATA REDUCTION only or LINEAR recording only? Or will we want to be able to select according to cut? (Cut selectable Data Reduction).

6) Do we need to interface with any satellite programming?

7) Do we need to interface with any CD juke boxes?

8) Do we want a system that provides the tactile feeling of carts (like what our jocks are used to now)?

Or.....

9) Would we rather have everything consolidated, away from the jocks, without them actually touching the media.

10) Is any type of automation in our future? Satellite? Live assist? Walkaway?

Additional comments:

CART MACHINE REPLACEMENT SURVEY

Name of Product:

Manufacturer:

How long has manufacturer been in business?

Type of System: TAPE RECORDABLE DISC FLOPPY DISK
(Circle One) COMPUTER/HARD DISK HYBRID

Which SAMPLING FREQUENCIES can be used?
16 kHz 20 kHz 22 kHz 26 kHz 32 kHz 44.1 kHz 48 kHz
(Circle all which apply)

Are the Sampling Frequencies cut selectable? YES NO

What type of DATA COMPRESSION is used?
NONE apt-X ATRAC
ADPCM
DOLBY AC2 MUSICAM II PROPRIETARY OTHER

Is DATA COMPRESSION cut selectable? YES NO

At a glance, does this system appear user friendly? Can I sit down in front of it and pretty much figure it out on my own (without the coaching from the manufacturers rep)?

YES NO

Comments:

At a glance, does this system look like it might spook the jocks? YES NO

Comments:

Price: Approximate cost of start up hardware. $

Cost of Software (in terms of $ per minute of storage, based on my desired sampling frequency and recording style- linear or compressed): $ per minute

Editing Capability: Will this system allow simple trimming of heads and tails of spots?

YES NO

Automation Ready:

Can this system be used in a live assist environment? YES NO

Can this system be used in a **fully automated** / walkaway environment? YES NO

Do the jocks touch the media? YES NO

Does this system interface with any traffic and billing software? YES NO

Which ones?

Speed: How long does it actually take to select, load and cue up a spot, ready for play?

Hardware: (For Computer based systems) What type of computer platform is used?

286sx 386sx 386dx 486sx 486dx Proprietary

What **SPEED** Clock? 16Mhz 25Mhz 33Mhz 40Mhz 50Mhz

Is the computer hardware supplied only by the manufacturer? YES NO

Or... can I buy (or trade) for the computer at a local supplier? YES NO

Additional Comments:

A SYSTEMS APPROACH TO NON-TRANSCODED AUDIO DELIVERY: A STEP TOWARD DIGITAL COMPATIBILITY

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Abstract

The practice of employing multiple, tandem, bit-rate-reduction encoding/decoding/re-encoding schemes for storage and transmission of high quality audio has generated industry-wide concern as AM & FM radio stations prepare to enter the digital era. It has been demonstrated that considerable impairment of the original source material can be predicted with even one such transcoding process. This paper will discuss the benefits and limitations of some contemporary coding approaches and will introduce the concept of how complete digital audio radio station plants can be designed to insure that the benefits of digital audio are realized without the risk of quality degradation.

INTRODUCTION

Globally, digital audio is transforming the business of radio broadcasting in many positive ways. Digital audio promises faster, smoother production and low cost automation, storage and transmission; and it will provide superior broadcast audio to the radio listener. It has recently been demonstrated that digital audio can also have a dark side with unexpected and severe quality degradation if radio stations and networks do not follow some radically new rules regarding facilities planning and station design.

Digital bit-rate-reduction or "compression," as it is sometimes called, is an essential element of today's broadcast equipment. When properly managed at the system level, digital bit-rate-reduction schemes perform the valuable service of permitting us to cost-effectively store, manipulate, edit and transmit CD-quality stereo sound by using low cost PC platforms and existing communications channels. However, when improperly managed at the system level, digital bit-rate-reduction systems can actually cascade negative effects and significantly degrade the overall quality of transmitted audio. How does the local station engineer or, for that matter, the network or transmission technical staff begin to plan a new digital facility while avoiding the pitfalls?

This paper introduces the concept of viewing the complete path from RPU transmission to our listeners' ears as one integrated system in which the integrity of the original source-coded material is maintained. This concept focuses on a new device by California Digital called the daX or digital audio Xpress. This unit is the first to be introduced to broadcasters which utilizes digital audio in a system-wide compatible manner and which may be used to model digital equipment interconnection architecture.

The author recognizes that at this early stage of radio's adoption of new digital technology, it is difficult to consider all of the system compatibility alternatives and to provide the answers to all of the digital transcoding compatibility issues. Instead, this paper is intended to provide engineering practices and guidelines for consideration when designing and building digital radio facilities, keeping in mind the need for total system compatibility.

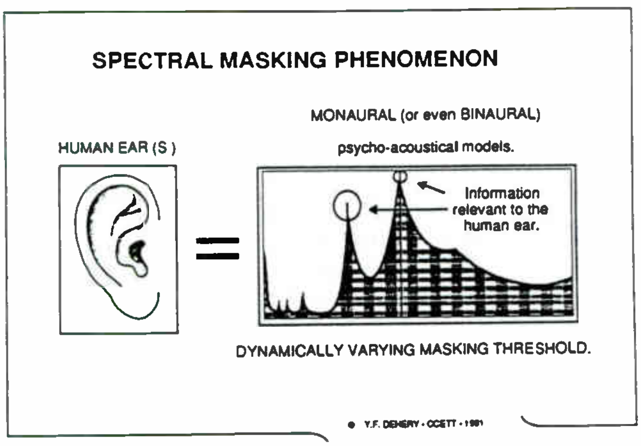
BACKGROUND

From a historical perspective, the main obstacle to implementing digital audio transmission systems has always been the bandwidth requirement. Uncompressed digital audio is bandwidth-intensive and simply will not conveniently or cost-effectively fit into the transmission systems which are used by radio broadcasters. To make digital audio practical, technologists have needed to explore new methods of coding audio material from the analog domain to the digital realm and devise new modulation approaches to transport the resulting bitstream.

Compression systems have been around since the 1950s. Early systems used the "Huffman Squeezing" method which took advantage of the redundancy of the letters E,R,T,N and O in text. Then, as now, the goal of any digital bit-reduction or compression algorithm was to identify and eliminate similarities or redundancies in data and, if possible, to remove unnecessary information. This has the result of minimizing the bandwidth or storage requirements of the transmission or storage medium.

Advances in psycho-acoustical modeling techniques have resulted in efficient approaches to bit-rate-reduction without perceived impairment of the original material.

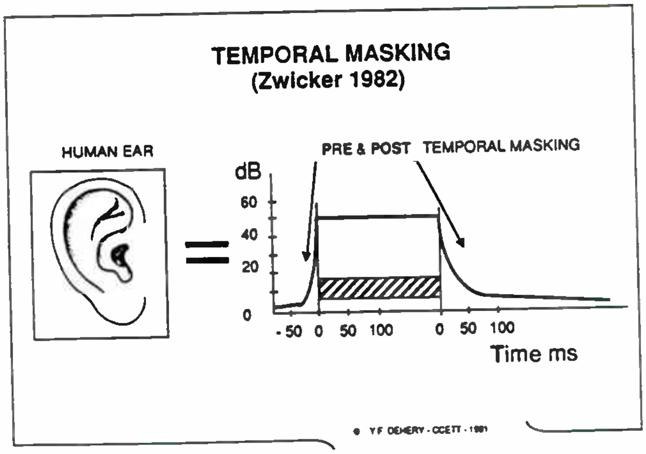
Figure 1 [1]



Typical approaches use the psycho-acoustic theories of Temporal Masking and Spectral Masking (critical band theory) under the larger umbrella term of "perceptual coding" to develop the PCM bit-rate-reduction algorithms.

Perceptual Coding systems take advantage of the belief that in the frequency and time domains, the presence of certain audio energy can actually "mask" adjacent audio signals.

Figure 2 [1]



ISO/MPEG LAYER II

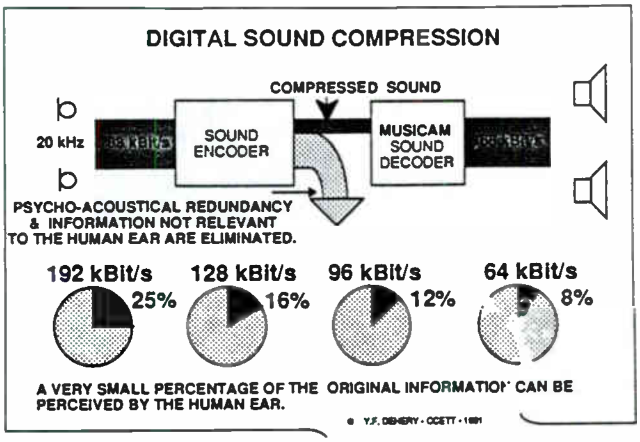
One of the systems using the "masking" approach, MUSICAM® (Masking pattern Universal Sub-band Integrated Coding And Multiplexing), has been adopted by standards organizations ISO/MPEG (International Standards Organization/Motion Picture Experts Group) as the Layer II standard. This standard has been proposed to be used by most US DAB proponents. The MUSICAM® system was developed jointly by French, Dutch and German engineers and has undergone a great deal of international testing and scrutiny in its short history.

BENEFITS OF BIT-RATE-REDUCTION

Figures 1 and 2 demonstrate the masking effects mentioned above. It can be seen that a large amount of audio information may be discarded from the transmission aggregate with no negative impact on the processed and decoded audio. In other words, since the discarded data bits will not be heard, nor will they, in most cases, contribute to enhancement of the decoded audio, they need not be included in the transmitted datastream. Therefore, the throughput and bandwidth requirements can be radically diminished.

Figure 3 illustrates the irrelevant information content for several coding rates using a perceptual approach. As will be discussed in later sections of this paper, this same benefit can also be a detriment if careful steps are not taken to avoid tandem codings, because once information is discarded as unnecessary during a coding process, it can never be recovered.

Figure 3 [1]



BIT-RATE-REDUCTION USERS

Virtually all digital modulation and transmission modes can benefit from some form of compression. This fact is evidenced by the continued impetus to pack more information into less bandwidth "real estate." In the transmission segment of a system, power is usually less of a limiting factor than is bandwidth, so less bandwidth usually means reduced operating costs.

It's now possible to transmit full duplex, 20 Khz. stereo audio and duplex 4.8kBps data to distant points over a single 2B+D ISDN Basic Rate Interface, or what is known by the phone companies as two DS0s plus signalling overhead. Digital compression technology now enables us to implement full duplex 7.5Khz. circuits over a single Switched 56 path using the CCITT G.722 standard hardware. In comparison, analog frequency-extender/translation techniques need six POTS (Plain Old Telephone Service) lines to yield only 5 to 7Khz. duplex service.

THE CAVEAT: TRANSCODING

For the purposes of this paper, the term "Transcoding" shall be defined as that group of processes employed to perceptually code/decode/re-code source audio in an A to D, D to A, A to D fashion. At first glance, this process may seem to be remote in its probability of occurrence. But, however, such is NOT the case! Consider the diagram in Figure 4. This diagram depicts a typical radio station of today being fed from a distant point (a network) by audio, which itself has been fed from a second remote point. The system described has a mixture of currently-available digital and analog equipment. The only departure from the present-day norm in the subject system is the DAB transmission/reception component, a system which, in the author's view, all US radio stations will eventually embrace.

Figure 4 details the very low percentage of the original input source material (1.411mBps, 44.1Khz. sample rate, 16 bit x 2, CD) which remains after cascaded transcodings have taken place. In this example, a coding rate for all processes of 256kBps per stereo channel was chosen for two reasons: (1) The ISO tests in Stockholm, Sweden, in May 1991 concluded that the 256kBps per stereo channel layer 2-coded material was statistically identical to the original material [2]; (2) It is believed by the author that 256kBps will be available to carry the stereo information from future FM DAB transmitters to their intended receivers. It should be noted that digital rate converters are available to implement lower bit rate codings, however, if lesser rate aggregate codings are introduced into the chain, the original material-retained numbers only worsen.

AES/EBU

A question has been raised as to the use of the AES/EBU digital I/O port which is available on some editors, automation systems and satellite receivers to alleviate transcoding issues. Since the AES/EBU interface is located after the perceptual decoder and before the coder in systems with which the author is familiar, no relief from transcoding problems can be expected by its use.

A POTENTIAL SOLUTION

A transmission system which addresses the problems inherent in transcoded systems is described in Figure 5.

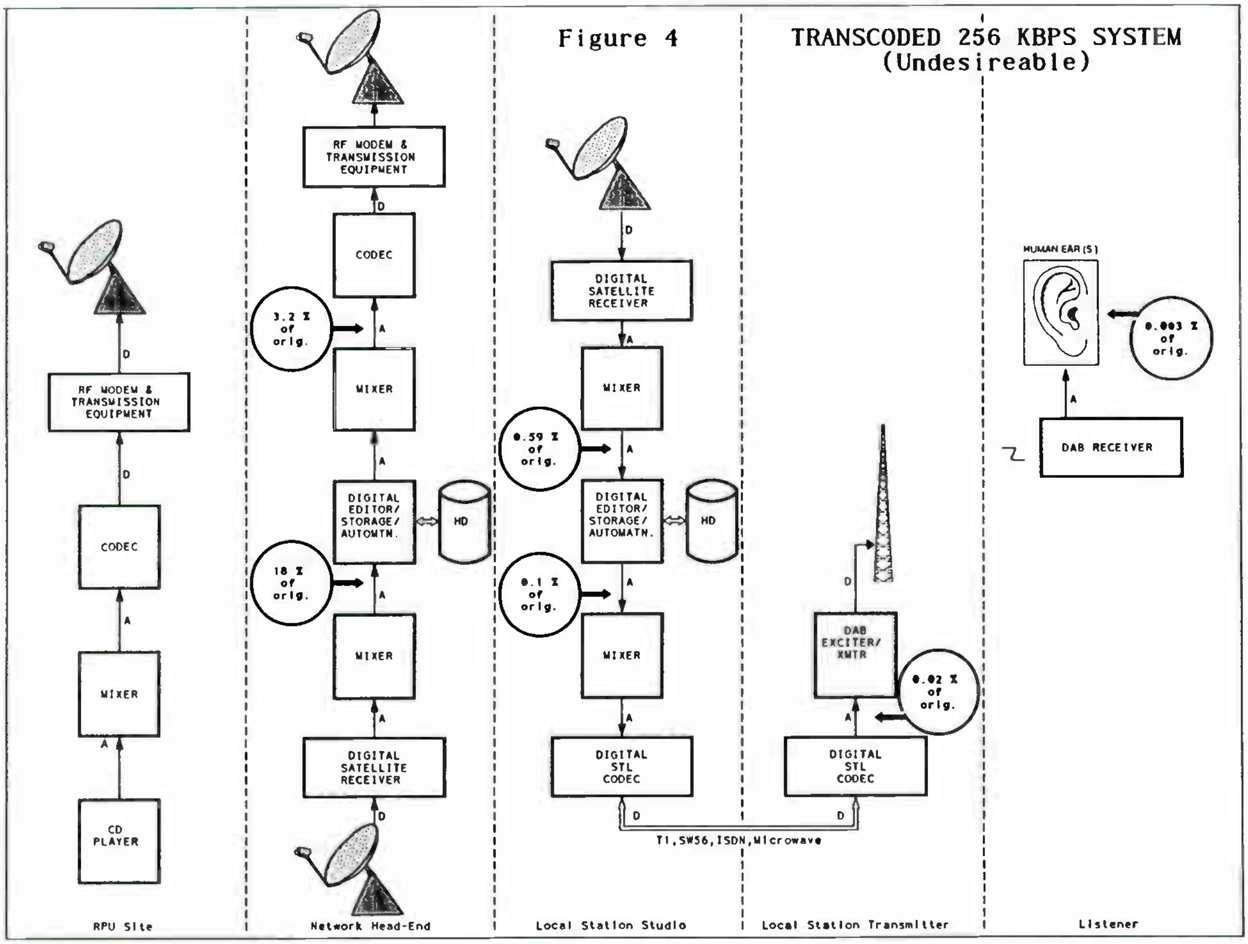


Figure 4 TRANSCODED 256 KBPS SYSTEM (Undesireable)

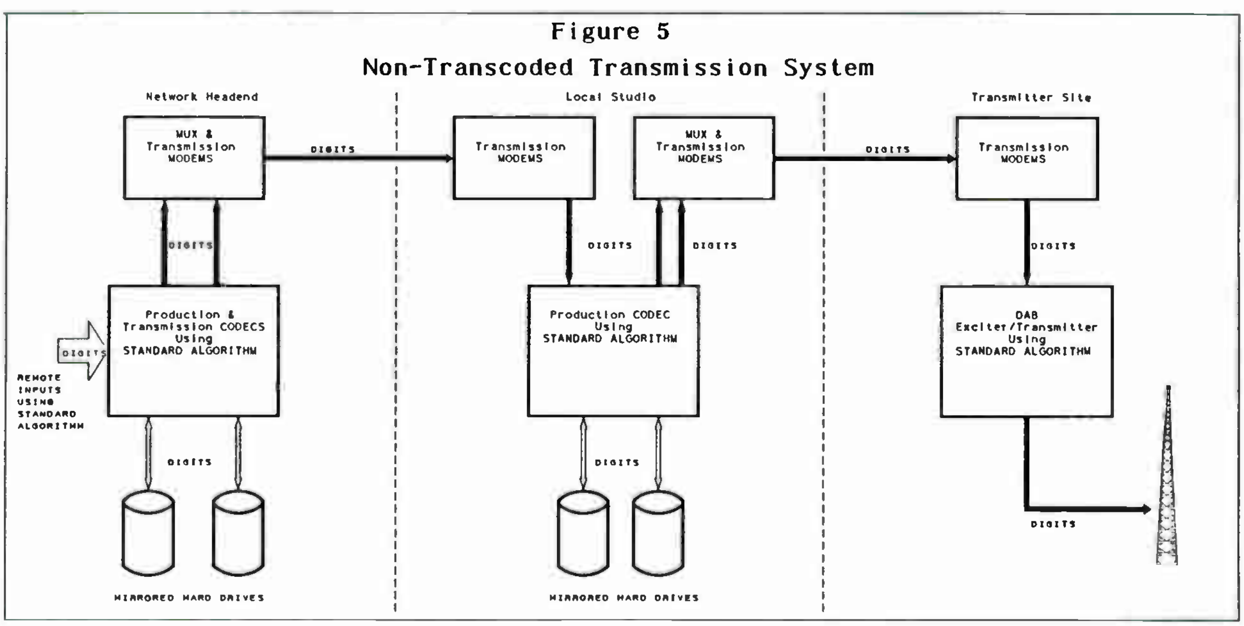


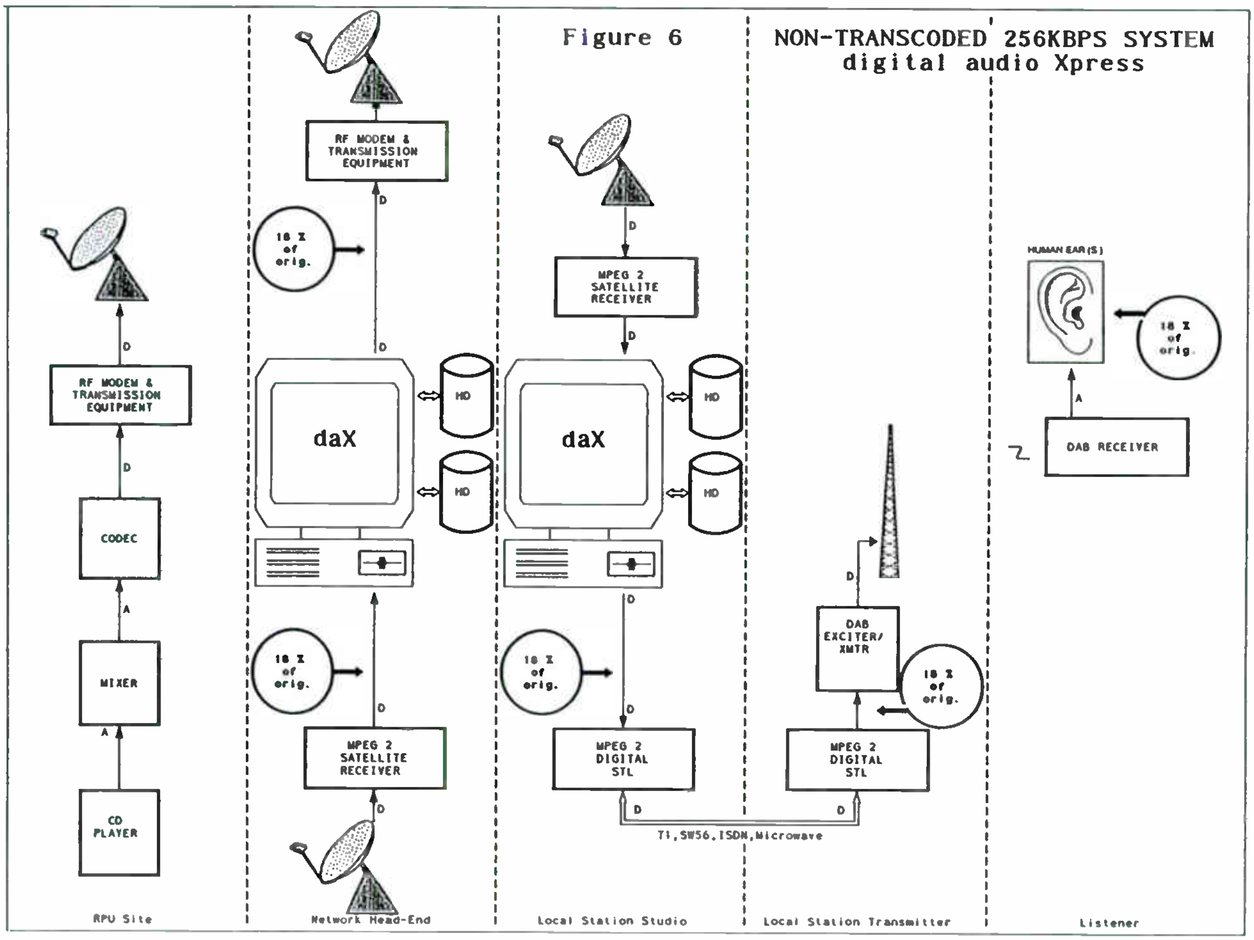
Figure 5 Non-Transcoded Transmission System

The basic premise underlying the design architecture of the non-transcoded system is simply to avoid any downstream re-codings of the audio information. To use this approach, a "reverse engineering" tactic is taken. In this scheme, the same perceptual coding algorithm must be used to source code the original material as is employed in the transmission of the DAB signal to the listener. Then, at each intermediate point in the chain, care must be taken to keep the information in its un-decoded digital state. Decoding to analog is needed only for local monitoring or for output where compatible digital facilities do not yet exist. If remote audio is to be stored and edited at the network head-end, the original information must be coded with the system-standard algorithm as must local station production audio. All editors in the system must be able to manipulate the existing data using the system-standard algorithm as well.

digital audio Xpress

Recently, a number of networks have identified system requirements to our firm which embody many of the concerns about transcoding and digital system compatibility that have been raised in this paper. The daX (digital audio Xpress) product was developed as a response to these requirements.

As can be observed by inspection of Figure 6, by adopting the non-transcoded architecture utilized in the daX product, a unity position regarding coding/decoding may be realized. Comparison of the percentage of original material retained at the listener's location in Figures 4 and 6 reveals that the non-transcoded architecture yields a better recovery figure of approximately **6,000** times that of its transcoded counterpart.



CONCLUSION AND CHALLENGE

The radio broadcasting industry is experiencing a paradigm shift away from traditional analog delivery of signals to the rapidly expanding scenario of DSP-controlled, algorithm-generated audio.
As we have seen, it is now possible to send very high quality, single generation Compact Disc audio from a remote site through our transmitter chain to listeners in the field with no discernable degradation. Research is continuing into algorithm enhancement and other related fields, which may one day lead to a technique for digital mixing, allowing combining of discrete serial bitstreams.

With a decision on a US DAB standard perhaps only months away, the time is right for us as technical professionals within the industry, to take steps to insure timely and orderly migration to compatible digital systems.

ACKNOWLEDGEMENTS

The author wishes to thank Paul Donahue, Gannett Radio Division; Mark Durenberger, CBS/Teleport Minnesota; and Dr. Larry Hinderks, Corporate Computer Systems, for their valuable assistance in the preparation of this paper.

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HDTV STATION ISSUES, PART I: TOWER PREPARATION, SITE SELECTION AND FACILITY PLANNING

Sunday, March 20, 1994

Moderator:
Charles Jablonski, NBC, New York, NY

*Papers not available at the time of publication

STRUCTURAL CONSIDERATIONS FOR A SUCCESSFUL TRANSITION TO HDTV BROADCASTING

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Abstract

With the final selection of an ATV system by the FCC, broadcasters are now in a position to start planning for the transition of their transmission facilities from NTSC to simulcasting both NTSC and HDTV, and, later on to exclusive HDTV transmission. Integral to this transition is the structural capacity of existing tower structures to support the additional loads imposed upon them by the addition of HDTV broadcast antennae and transmission lines. This paper will address the requirements and various aspects of a tower structural design analysis and the results of which will assist the broadcaster in implementing a successful transition to HDTV broadcasting while complying with current design standards, government regulations and insurance requirements.

INTRODUCTION

Many existing broadcast tower structures have been in service for 25 years or longer. During the course of their service life, most of these towers have probably seen the addition of owner and tenant broadcast and communications equipment for which they may not have been originally designed to support. The further addition of a second antenna and transmission line required to broadcast an HDTV signal simultaneously with the broadcaster's current NTSC signal will result in additional stresses being imposed upon the tower's structural components for which they may not have the necessary capacity to resist.

In addition to the possible increases in tower equipment loading, the design standards and minimum requirements for design wind loads have changed over the years to the point where we may find that even though a tower's original equipment design loading has not changed over the course of its service life, based upon current recommended design standards, the tower's structural components may already be considered as being stressed beyond their allowable load carrying capacity.

Thus, the owner of a broadcast tower faces a problem that is twofold, how to safely support the additional loadings of an HDTV broadcast antenna and transmission line, and how to bring his tower up to date with the most recent recommended design standards. A structural design analysis is the means by which the tower owner may reach a decision regarding the integrity of his antenna support structure.

COMPONENTS OF A STRUCTURAL ANALYSIS

The variable components of a structural design analysis are the equipment, or appurtenance loads imposed upon the tower and the magnitude and method for applying the wind loads. The appurtenance loads are well defined and may be simply described as the forces resulting from the dead weight of the appurtenance, which is a downward acting force, and the shear, or horizontal force, produced by the wind blowing against the projected area of the appurtenance. Depending upon the location of the appurtenance on the tower, bending and torsional (twist) loading may result in additional forces being imposed upon the tower's structural components.

These appurtenance loads are supported by the tower through the tower legs and its bracing system of diagonals (X-bracing) and struts (horizontal bracing) which transfer the horizontal forces produced by the wind into the tower legs and points of tower support. These points of tower support are the tower base and in the case of a guyed tower, the tower guy lines.

Now, obviously the larger the appurtenance loads are, the larger or stronger the tower structure must be in order to safely support them.

TYPES OF SUPPORT STRUCTURES

Basically, what we are talking about are two types of support systems - self supporting and guyed.

Self Supporting Towers
A self supporting tower may be described as a cantilevered space truss whose design and resultant structural member stresses follow a very linear pattern of solution. Generally speaking, as long as the revised design loads are less than the original design loads, or, if greater than originally designed but applied at a proportionally lower elevation on the tower, the existing structure will most probably satisfy your requirements. Should the revised design loads be greater or applied at a higher elevation on the tower than originally designed, a structural investigation will be required to determine the adequacy of the existing tower members to support the revised design loading.

Guyed Towers
A guyed tower may simply be described as a beam-column on elastic supports, with each support possessing a different degree of stiffness. This is a much more complicated structure than a self supporting tower. The stiffness at each guy level is generally dependant upon two items, the size of the guy strand and the amount of initial tension in the guys at each guy support level. These different degrees of stiffness affect the distribution of tower stresses and the resultant tower deflection. Any change in the tower's appurtenance loading, by addition, deletion, or relocation, will result in a non-linear redistribution of tower stresses and deflection. It is obvious then that for a guyed tower any change in tower design loading will require that a structural design analysis be undertaken to determine the adequacy of the existing members to support the revised loadings.

REQUIREMENTS OF THE OWNER

Prior to beginning a structural design analysis a number of things will be required by the engineer, among the most important being a complete description of the existing tower structure. When the tower to be analyzed was initially constructed, the original supplier most probably provided a set of drawings which would have included an elevation and cross sectional views of the tower. Both of these drawings would have included information describing the geometry and the overall dimensions of the tower structure. Also included may be information regarding structural member sizes and strength of material. These drawings may also show the initially erected antennas and equipment and possibly any future equipment for which the tower had been designed

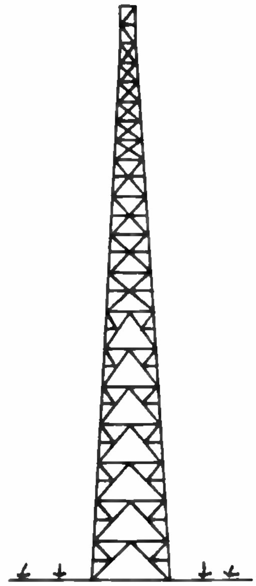

FIGURE 1. Self Supporting Tower

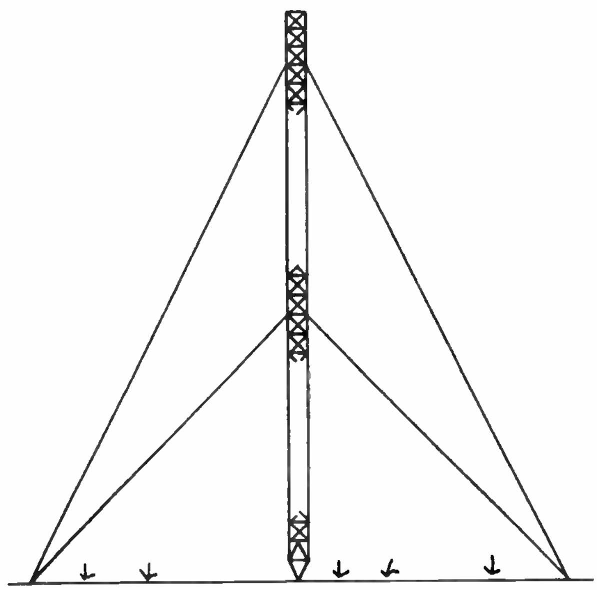

FIGURE 2. Guyed Tower

to support.

In addition to the tower drawings, a set of foundation plans indicating the locations and construction requirements for the tower base and guy anchors along with a copy of the geotechnical report should have been provided.

The above items along with the original contract documents outlining the specific design requirements should be in the tower owner's files. Along with the original information, documentation which describes any modifications undertaken since the initial tower construction should also be included in these files.

Along with the complete description of the tower's structural configuration, an itemized inventory of existing antennas, transmission lines, and other appurtenances installed on the tower must be provided. This inventory should include type and model number, dimensions and mechanical properties when available, transmission line size and type, and specific location on the tower.

If the above information is not readily available, a further investigation must be undertaken to gather the information in order for the engineer to provide an accurate analysis for the existing structure.

The next question that must be answered is that concerning the design standard and design wind loading to specify. Most older towers were designed under the early Electronic Industries Association (EIA) RS-222 Standards. These standards followed the even earlier Radio-Electronics-Television Manufacturers Association (RETMA) TR-116 Standard. Both of these early standards address wind loading in the simplified terms of a uniform pressure. The most recent revisions of the EIA-222 Standard now address wind loading in terms of a basic wind velocity with a force equation including factors which cause the effective wind pressure to escalate as the elevation of the tower increases.

There is currently discussion going on within the EIA TR-14.7 Subcommittee which writes the tower design standard regarding the proper EIA design standard to recommend for use in the redesign of existing towers. At this time there does not appear to be a clear consensus regarding this issue. However, the building authority governing local construction may specify within their building code the design standard and minimum wind loading which must be considered should tower reinforcing be required. Their required standard may possibly be a design standard other than an EIA Standard. Further, the tower insurer may require a minimum standard and wind loading in order to maintain insurance coverage for the property. These sources, along with local climatological data, should be investigated by the tower owner prior to specifying the design parameters to be utilized by the engineer.

With the above information in hand, the engineer will now be in a position to work with the broadcaster in conducting an analysis to determine the impact of adding the selected HDTV antenna and transmission line to the tower support structure.

ANALYSIS RESULTS AND RECOMMENDATIONS

Upon completion, the analysis results will reveal that the broadcaster has one of three options. The first option will be that the tower and its structural components have sufficient capacity to support the additional loadings of the proposed HDTV antenna and transmission line and their installation may proceed as planned. The second option will be that one or more of the tower's structural components are stressed beyond their allowable load carrying capacity and must either be reinforced or replaced in order for the tower to support the proposed additions. This will require that prior to proceeding with the planned installation, structural modifications will be required to be implemented. The third and final option will be that the addition of the proposed HDTV antenna and transmission line will prove to be so detrimental to the tower support structure that the necessary reinforcing would prove to be prohibitively expensive or beyond the bounds of good engineering judgement.

With the results of the first option being obvious and the results of the third option requiring the construction of a new tower or some other change in plans for implementing HDTV broadcasting, the balance of this paper will address the ways in which a tower structure may be reinforced to support additional loadings.

For a guyed tower, the first and simplest method to attempt to relieve member overstress is to adjust the stiffness of the tower at the guy points by revising the guy line initial tensions. By adjusting initial tensions we may be able to shift stresses from an area that is overloaded to an area that has sufficient capacity to support additional loads. This method of relieving member overstress works quite well; however, it has its limits in the amount of adjustment that can be made in the initial tensions before we begin to see detrimental results start to show up in the resultant tower deflection, maximum guy line forces, and additional member overstress. If a guy line is stressed beyond its allowable load carrying capacity and the amount of overstress cannot be reduced sufficiently by adjusting the guy line initial tensions, or, if by adjusting the guy line initial tensions we compromise some other portion of the structure, the existing guy line may be replaced by a guy of a different size and capacity.

In the case of overstress in a tower leg, it is fairly obvious that replacement is not the most economical solution. What we need to do is determine the best way to make the overstressed leg stronger in order to support the additional loads. This may be accomplished in several ways, two of which follow:

1. **Redundants**
   Redundants are secondary bracing members which are introduced into the structural system to reduce the unsupported length of the tower leg. By placing the redundant at mid-height of a bracing panel, we in effect have created a column half as long as the original, and thus have increased its stiffness and ability to support additional loads.

2. **Reinforcement**
   When the installation of redundants alone would not be sufficient to allow the existing tower leg to support the additional loadings, reinforcement in the form of additional steel material may be added to the existing tower legs to create a larger and stronger structural member. Several articles regarding strengthening existing structures along with research information have been detailed in publications by a number of leading authorities. Particular care in the design and field installation should be taken when this type of reinforcing is required.

Overstressed diagonals and horizontal struts may be either replaced with members of a larger capacity or reinforced, depending upon the type of member involved and the nature of its loading.

In addition to the main structural member requirements, end connections for the diagonal and strut braces, tower leg splices and guy connections at both the tower and anchor ends must be investigated. Connection bolts and pins may require replacing with larger or stronger material. Connection plates may require reinforcing around bolt and pin holes.

Additional investigations must be performed on the tower foundations to insure the adequacy of their design to support the revised loadings. Specific recommendations

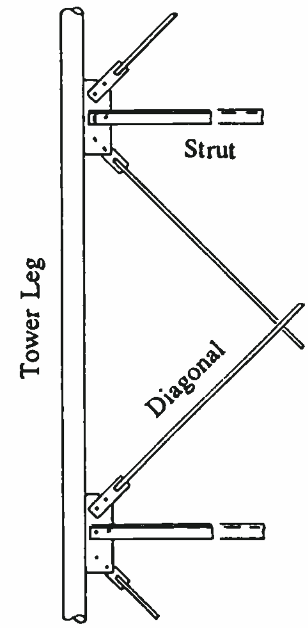

FIGURE 3. Typical Tower Members

for reinforcement or revision to the foundations will be made based upon the type of foundation construction involved.

Once an initial design analysis has been completed and recommendations reviewed, the broadcaster and engineer will now be in a position to examine various alternatives to possibly lessen the impact of the addition of the HDTV antenna and transmission line. These alternatives may include removing or relocating existing appurtenances on the tower, or revising the HDTV antenna and transmission line type, size, or location. In any event, it is essential that the broadcaster and engineer confer to assure that the optimum in broadcast equipment and equipment location is provided for.

CONCLUSIONS

The purpose of the preceding has been to emphasize the importance of a thorough investigation by a qualified structural engineer when changes or additions are proposed in the appurtenance loading of a broadcast tower. A number of options are available to allow the broadcaster to comply with current standards and regulations and still maintain his tower's structural integrity. In light of the pending advent of HDTV broadcasting, this facet of a successful transition from NTSC to HDTV should not be overlooked.

THE EFFECT OF CHANNEL ASSIGNMENT ON TRANSMITTER AND RECEIVER REQUIREMENTS FOR EQUIVALENT HDTV/NTSC COVERAGE

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Introduction

Broadcasters, especially those who will be moving from VHF-NTSC transmission to UHF-HDTV transmission, are concerned with providing equivalent HDTV coverage to their NTSC viewers. This concern can be summarized by the following questions: (a) What will it take to provide a *reliable* HDTV picture and sound to the viewers who now receive a passable (grade 3) picture and sound? (b) How will the UHF-HDTV channel assignment affect the station's ability to provide equivalent coverage relative to other UHF-HDTV stations and relative to their own present NTSC coverage?

This paper will define the concept of equivalent and reliable coverage and will show the derivation of the correct planning factors, including the effects of sky temperature and Low Noise Amplifier (LNA) at the receiver end. Two approaches to the determination of the planning factors and transmitter size will be presented. One is based on absolute power requirements and the other is based on power levels relative to the existing NTSC transmission. Finally, the areas where knowledge is presently lacking will be highlighted.

Equivalent Reliable Coverage

Equivalent coverage is established when the HDTV channel provides reliable pictures and sound through the Grade B contour of the existing NTSC channel. In this paper, the coverage area is not considered limited by interference from other stations. That is, the coverage and service areas are seen as equal. Grade B contour is defined here as the contour of a Passable (grade 3) picture. This is equivalent to (peak-of-sync) carrier-to-noise ratio (CNR) of 28dB[1] with at least 90% of time availability at the best 50% of the locations. To replace a grade 3 picture with HDTV, a CNR of 16dB is sufficient. However, there are three twists to the 16dB CNR level for HDTV.

First, unlike NTSC, the CNR for HDTV is a *threshold* value below which picture and sound become unavailable. Because it is a threshold value, it must be maintained close to 100% of the time for reliable service. In this paper, 99% availability was selected.

Second, a carrier penalty, also known as "implementation margin," must be added to the theoretical threshold-CNR to compensate for the residual distortions at the transmission plant. In NTSC, these distortions will appear as picture impairments but will not affect the coverage. The reverse is true for HDTV. In HDTV, the distortions will not affect picture quality but will raise the "noise" level. The result is a reduced coverage area unless the carrier level is increased to maintain the 16dB threshold CNR. Simply put, two transmitter/antenna systems, classified as equal and acceptable using standard NTSC measurements, may provide two unequal HDTV coverage contours.

Third, in NTSC, the CNR is defined by the RMS power during sync pulses. Therefore, the power during sync pulses serves as a useful definition for both transmitter size and coverage contours. In HDTV, the coverage contour is defined in terms of the average operating power, and the transmitter size is defined by the peak instantaneous power (maximum amplitude). The difference between the average

[1] This value is consistent with TASO data (1959) based on random noise. TASO's definition of Passable picture is based on CNR of 27dB with 6MHz random noise. Translated to 4.2 MHz, the equivalent CNR is 28.5dB.

operating HDTV power and the peak operating NTSC power is expected to be 10±3dB[2]. The definition of equivalent and reliable HDTV/NTSC coverage is embodied in Figure 1.

Frequency Dependent Factors

Even though the FCC propagation curves are not channel-specific within the UHF band, the CNR at the TV receiver depends *significantly* on channel assignment. There are three elements that contribute to this dependence:

A. Receive antenna effective interception area. This area is defined as a multiplier of the wavelength squared. The larger the area, the higher is the intercepted power level. Therefore, for a given physical size, the interception area decreases with the square increasing frequency. For a constant-gain receive antenna, this factor implies that the effective radiated power at channel 69 will have to be at least 4dB (2.5 times) higher than that at channel 14 for equivalent coverage. This effect can be mitigated by a high gain (>14dB @ ch.69) receive antenna, the largest dimension of which will be at least 4 feet.

B. Downlead cable loss. The loss of a coaxial cable increases with the square root of the frequency. The increase in carrier loss from channel 14 to 69 is approximately 2dB. The high downlead loss and its variation with frequency can be mitigated by adding a low-noise-amplifier (LNA) at the terminals of the receive antenna.

C. Antenna Noise. There are two sources of thermal noise at the receiver. One is the ambient temperature of the LNA, downlead cable and the receiver. The second source is antenna noise. Antenna noise is due partly to the surrounding ground temperature and partly to sky temperature. Sky temperature decreases with increasing frequency and, as shown in Figure 2, its maximum value at channel 14 is equal to 290°K, the ambient temperature.

[2] For 32QAM modulation, a maximum peak RF (not "envelope") to average power ratio between 11 and 13dB, depending on channel filters, is expected.

Receiver Model

The receiver model is shown in Figure 3 along with the expression for the overall noise figure referenced to the terminals of the receive antenna. With a high gain LNA and a short cable between the antenna and the LNA, the system noise figure is the same as that of the LNA alone. The receive antenna, being directional and broadside, aquires half its noise from the sky temperature and the other half from the ground temperature. The ground temperature is assumed to be the same as the ambient temperature.

Fade Margin and Reliability

The signal availability throughout the coverage area is statistical in nature. The signal level varies from noon to midnight, as weather changes and as summer turns into winter. To overcome these and other variations, a fade margin is introduced. Just how big this margin is depends on the required signal availability. In UHF-NTSC, the requireded availability is at least 90% of time for Grade B coverage. For HDTV a larger percentage is required because the critical parameter, CNR=16dB, is a threshold value which may not go lower at any time, else picture and sound may disappear. In this paper, the required availability was set at 99%. Figure 4 provides the fade margins, in dBu over FCC(50,50), as a function of radiation center height and distance from the transmitter.

Link Equation

The required *average* Effective Radiated Power (ERP) to provide CNR=16dB at least 99% of the time at the best 50% of the locations is:

$$(1)\ ERP_{HDTV}(dBk)=20Log(f_{MHZ})-G_R(dB)+10Log(N/N_0)-FCC(50,50)+M(99)-15.13$$

where:
G_R is the peak gain of the receive antenna relative to $\lambda/2$ dipole.
M is the fade margin given in Figure 4.
N/N_0 is the relative system noise for 6 MHz bandwidth. It is defined by:

$$(2)\ N/N_0=F-.5(1-T_S/290)$$

where the system's noise figure, F, is defined in Figure 3 and the sky temperature, T_S, is defined in Figure 2.

Figure 5 is a graphical depiction of the link equation. The receiver parameters (typical) used in Figure 5 are:

TV receiver noise figure=10dB.
Receive antenna gain=10dB any channel.
LNA gain=20dB any channel.
LNA noise figure=4dB.
No mismatches and no baluns.

Choosing a height of 1200 feet above the average terrain, and 56 miles to the Grade B contour, the required ERP would be 21.9dBk at channel 14 and 26.2dBk at channel 69 for a constant downlead loss of 6dB. A downlead loss of 6dB is equivalent to 50 feet of RG-59A cable at mid-UHF band with approximately ±1dB variation to upper and lower ends. Therefore, the required ERP will vary from 20.9dBk for channel 14 to 27.2dBk for channel 69 with a 50 feet of downlead cable. Simply put, the *average* ERP at channel 69 will have to be four times that of channel 14 for equal coverage unless the gain of the receive antenna increases by 6dB from channel 14 to channel 69.

Figure 5 can be developed for various parameters such as lower fade margins, and without LNA. Some comparative results are given in Tables 1 and 2. For Grade B contour of less than 56 miles, subtract 1dB/mile from the calculated ERP.

Transmitter Size

With the ERP known, the transmitter size can be defined by:

$$(3)\ TX_{HDTV}(dBk)=F_T(dB)-P_{BO}(dB)+ERP_{HDTV}(dBk)-G_T(dB)-\eta_T(dB)$$

where:

F_T is the carrier penalty due to residual distortions. A typical value of 1dB, varying by manufacturer, is expected.

P_{BO} is the power backoff level of the final amplifier, below the NTSC operating point, to the average power of the HDTV signal. The backoff level is a tradeoff between the carrier penalty and adjacent channel spillover.

G_T is the gain of the transmitting antenna relative to $\lambda/2$ dipole.

56 miles contour. HAAT=1200' 50' Downlead G_T =14dB G_R=10dB P_{BO}= -7dB F_T=1dB	% of TIME AVAILABILITY HDTV 90	99
CHANNEL 14	ERP=360 TX=114	ERP=2022 TX=639
CHANNEL 69	ERP=1029 TX=325	ERP=5787 TX=1830

Table 1. ERP (kW) and TX(kW) for equivalent UHF-HDTV/UHF-NTSC coverage.
Receive antenna *without LNA*.

56 miles contour. HAAT=1200' 50' Downlead G_T =14dB G_R=10dB P_{BO}= -7dB F_T=1dB	% of TIME AVAILABILITY HDTV 90	99
CHANNEL 14	ERP=25 TX=8	ERP=141 TX=44
CHANNEL 69	ERP=65 TX=21	ERP=367 TX=116

Table 2. ERP (kW) and TX(kW) for equivalent UHF-HDTV/UHF-NTSC coverage.
Receive antenna *with LNA*.

η_T is the transmitter-to-antenna efficiency.

For example, assuming an omnidirectional transmitting antenna gain of 14dB(x25), a line loss of -1dB, a carrier penalty of 1dB and final amplifier power backoff of -7dB to accommodate peak power levels, the transmitter size can be determined from:

$$(4)\ TX_{HDTV}(dBk)=ERP_{HDTV}(dBk)-5dB$$

Applying equation (4) to the example, the required transmitter size, with and without LNA at the receive antenna, is given in Tables 1 and 2.

The results of Tables 1 and 2 show that, for the example chosen, the gain of the receive antenna, even with LNA incorporated in it, will have to be raised from 10dB to at least 13dB at channel 69, two times higher than currently used roof-top antennas, for reliable coverage to 56 miles from an omnidirectional

56 miles contour. HAAT=1200' P_{BO}= -7dB F_T= 1dB	% of TIME AVAILABILITY HDTV	
	90	99
NTSC channel 69 HDTV channel 14	ERP= -15.05 TX= -7.05	ERP= -7.55 TX= +0.45
Same channel for HDTV and NTSC	ERP= -10.45 TX= -2.45	ERP= -2.95 TX= +5.05
NTSC channel 14 HDTV channel 69	ERP= -5.85 TX= +2.15	ERP= +1.65 TX= +9.65

Table 3. Differential ERP (dB) and TX(dB), over NTSC, for equivalent UHF-HDTV/UHF-NTSC coverage. HDTV and NTSC receive antennas *without LNA*.

56 miles contour. HAAT=1200' P_{BO}= -7dB F_T= 1dB	% of TIME AVAILABILITY HDTV	
	90	99
NTSC channel 69 HDTV channel 14	ERP= -26.59 TX= -18.59	ERP= -19.09 TX= -11.09
Same channel for HDTV and NTSC	ERP= -22.43 TX= -14.43	ERP= -14.93 TX= -6.93
NTSC channel 14 HDTV channel 69	ERP= -17.87 TX= -9.87	ERP= -10.37 TX= -2.37

Table 4. Differential ERP (dB) and TX(dB), over NTSC, for equivalent UHF-HDTV/UHF-NTSC coverage. HDTV receive antenna *with LNA* and NTSC receive antenna *without LNA*.

antenna at 1200 feet above average terrain and a transmitter size of less than 100kW.

Power Relative to NTSC

For Grade B NTSC, equation (1) becomes:

(1a) $ERP_{NTSC}(dBk)=20Log(f_{MHZ})-G_R(dB)+10Log(N/N_0)-FCC(50,50)+M(90)-4.68$

The differences between equations (1) and (1a) are in noise bandwidth (4.2 vs 6 MHz), CNR (28 vs 16dB) and fade margin (90 vs 99%).

By subtracting equation (1a) from (1), and assuming in-band (UHF) transition from NTSC to HDTV, no changes in the basic receiver parameters, and assuming no change in the location of the transmitting antenna, the differential ERP, over that required for NTSC, is:

(5) $\Delta ERP=20Log(f_{HDTV}/f_{NTSC})+M(99)-M(90)-10.45$

The differential fade margin, M(99)-M(90), is (from Figure 4) approximately 7.5dB at 56 miles for antenna height of 1200' above the average terrain. Therefore, if $f_{HDTV}=f_{NTSC}$, the average ERP_{HDTV} for 99% availability is -2.95dB below that of NTSC. For 90% availability, the *average* ERP_{HDTV} is -10.45dB below that of *peak-of-sync* ERP_{NTSC}.

Thus, if the present NTSC contour of CNR=28dB is ascertained at key locations, the average ERP_{HDTV} for equivalent coverage can be determined for any percentage of reliability.

Equation (5) is a special application of the technique. In general, all the terms of equations (1) and (1a) will apply.

Similarly, since the size of the NTSC transmitter is:

(3a) $TX_{NTSC}(dBk)=ERP_{NTSC}(dBk)-G_T(dB)-\eta_T(dB)$

then, by subtracting (3a) from (3) and assuming no changes in the gain of the transmitting antenna and line efficiency, the differential power, in dB, of the HDTV transmitter for equivalent coverage is:

(6) $\Delta TX(dB)=F_T(dB)-P_{BO}(dB)+\Delta ERP(dBk)$

For the example chosen earlier with F_T=1dB and P_{BO}= -7dB, the differential transmitter size relative to the NTSC operating point reduces to:

(7) $\Delta TX(dBk)=\Delta ERP(dBk)+8$

Applying equations (5) and (7) to a UHF station with Grade B at 56 miles, the differential ERPs and transmitter sizes are given in Tables 3.

To overcome the requirement for increase in the

HDTV transmitter size over that used for NTSC, as indicated in Table 3, an LNA could be added to the HDTV receive antenna. In the latter case, equation (5) becomes:

$$(8)\ \Delta ERP=20Log(f_{HDTV}/f_{NTSC})+M(99)-M(90)-10.45+10Log(N/N_0)_{HDTV}-10Log(N/N_0)_{NTSC}$$

The differential ERPs and transmitter sizes for various channels and availability for a typical outdoor HDTV receive antenna with LNA are give in Table 4.

Missing Knowledge

The procedures derived in this paper can be improved once certain information becomes available. The areas were information is lacking are:

A. FCC Propagation Curves and Fade Margins.
The present curves apply to narrow-band, NTSC transmission, were most of the picture information is near the carrier. In HDTV, the picture information is evenly spread over the entire channel. The wider bandwidth required for HDTV may affect the conversion factor from incident field at the receiver to transmitted power.

B. Receiver CNR Penalty.
As the channel equalizer at the receiver attempts to eliminate the effect of multipath, it reduces the CNR margin. For heavy multipath, the equalizer may force the CNR below threshold. The passband performance of the path between the transmitter and the receiver, sometimes called "channel characterization," is required for urban, suburban and rural environments in order to determine the CNR penalty at the receiver.

C. Transmitter CNR penalty.
Reliable information on the tradeoff, as a function of backoff level, between CNR penalty and adjacent channel spillover is not publicly available.

D. Reliability and Performance of Outdoor LNA.
Besides protection against adverse weather and electrical storms, an acceptable LNA should exhibit minimum of linear and non-linear distortions and provide proper isolation against man-made noise. Not meeting these criteria may incur CNR penalty.

Conclusion

The significance of Tables 1-4 is that they demonstrate the desirability of adding an LNA to the HDTV receive antenna for stations wishing to provide *reliable* HDTV coverage for at least 56 miles from a transmitting antenna 1200' above the average terrain. Further, the calculations show that in the case of HDTV, which does not exhibit graceful degradation similar to NTSC, proper coverage and system analyses must be channel-specific.

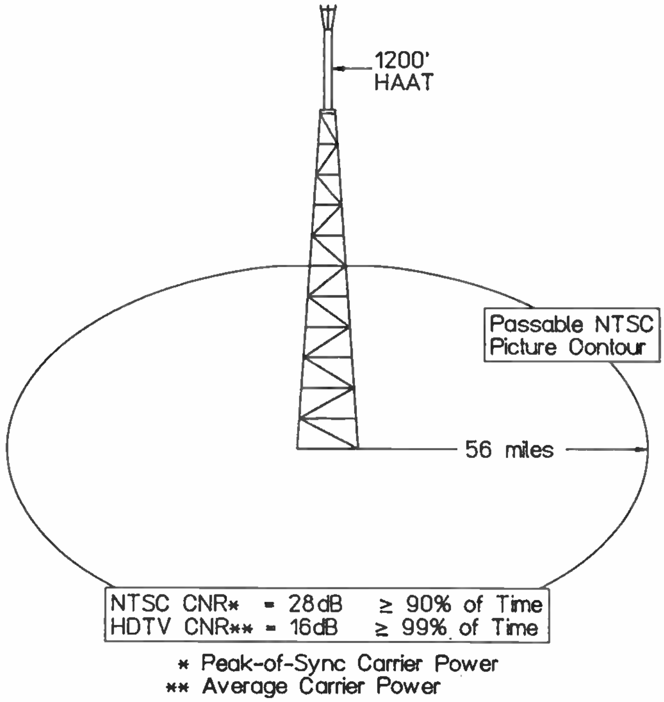

Figure 1

Equivalent HDTV/NTSC Coverage

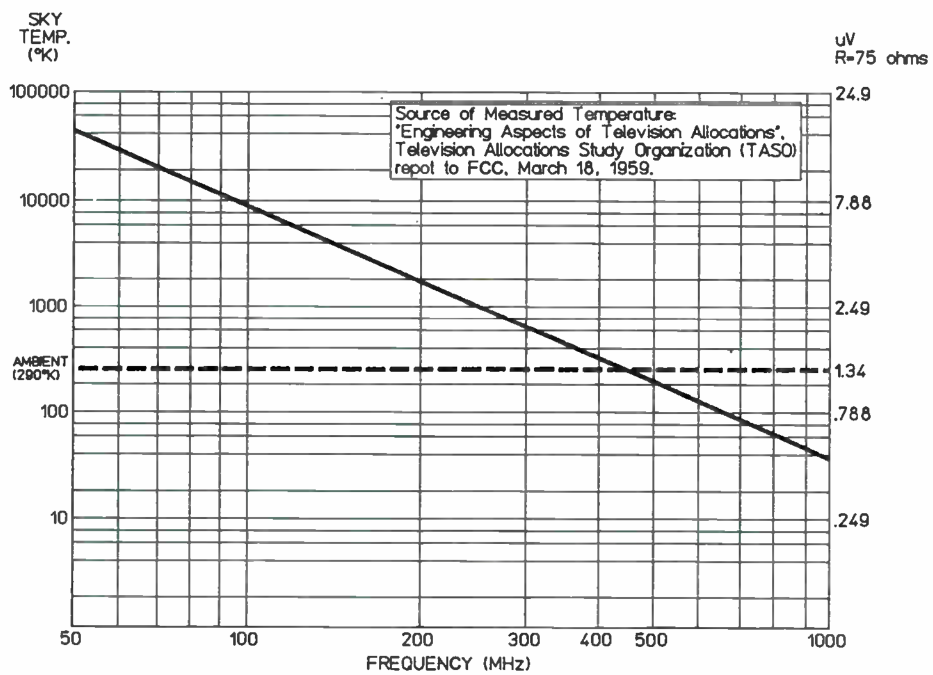

Figure 2

Maximum Sky Temperature

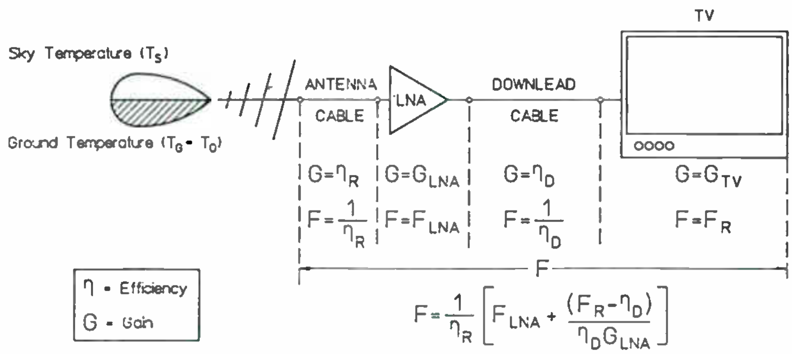

Figure 3

Receiver Model for CNR Analysis

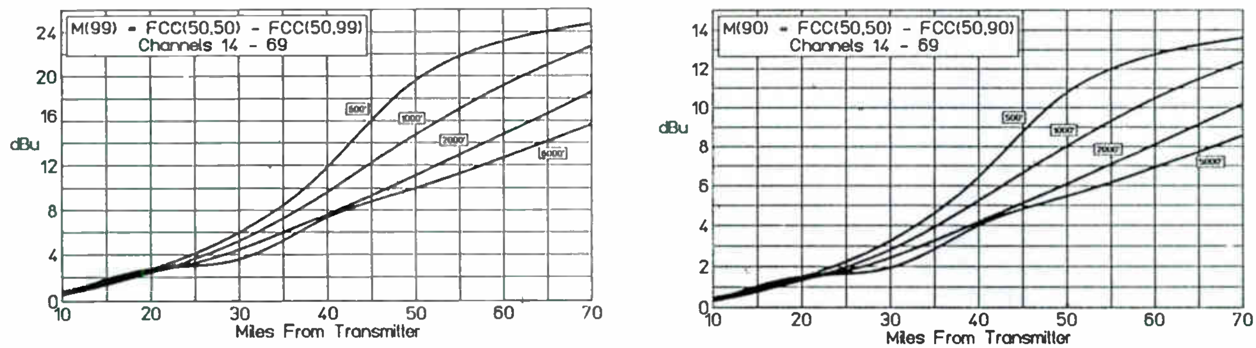

Figure 4

Fade Margin vs. Distance From Transmitter for Various Transmitting Antenna Heights Above Average Terrain

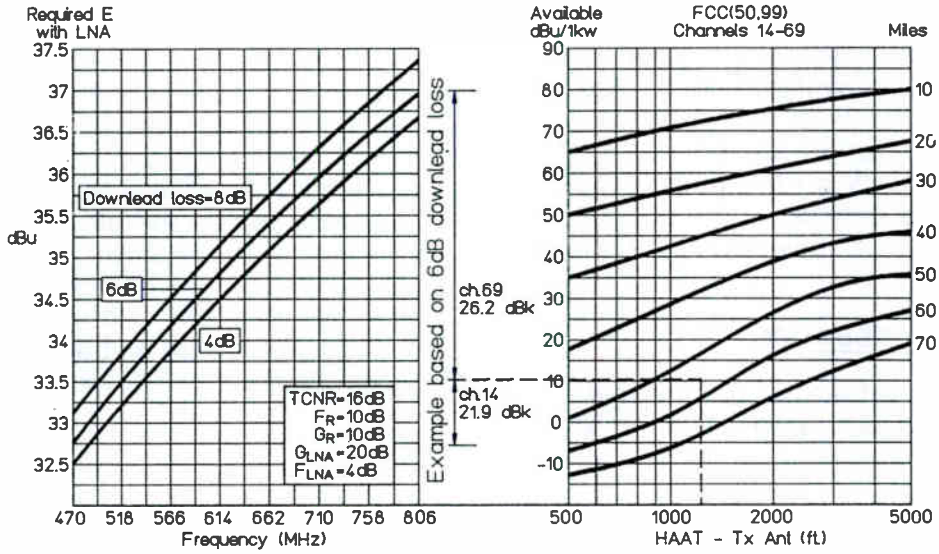

Figure 5

Required ERP in dBk with the use of a LNA. Thermal and antenna noises only. Availability at least 99% of the time at the best 50% of the locations.

AN HDTV RF SYSTEM FEASIBILITY FLOWCHART

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ABSTRACT

As the schedule for HDTV implementation advances and release of the transmission standard becomes imminent, broadcasters must focus on how they will provide HDTV service to their viewers. Schedules and budgets must be prepared now to insure proper funding for implementation. A key step in the process is to determine where the new HDTV antenna, transmission line and transmitter will be located and how system performance will be affected by these choices. The greatest challenge to HDTV implementation will be finding available tower capacity.

This paper presents a detailed list of important items to examine when planning an HDTV RF system installation. A step-by-step flowchart lays out the process for making tough RF system choices and defining an optimum configuration. Use of the flowchart results in a well thought out implementation scenario with schedules and budgets. Some items on the flowchart are performed by the station, others require input from consultants, tower specialists and industry experts.

INTRODUCTION

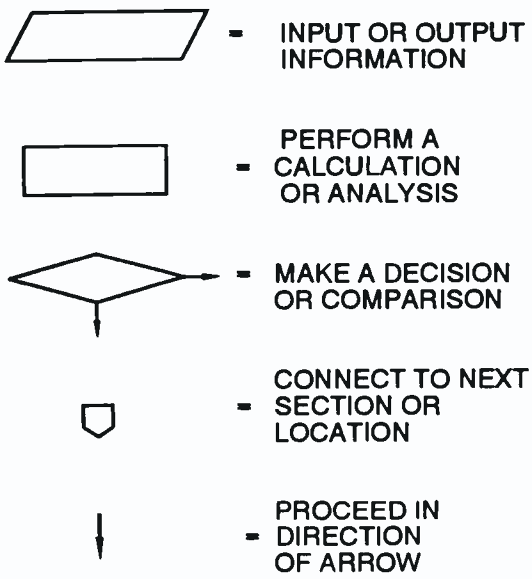
The issues related to getting an HDTV signal on-the-air are complex and inter-related. They include the following:

- Scheduling- when will service begin?
- Licensing
- Industry capacity
- Intra-station handling
- Coverage
- Multichannel possibilities
- Tower space availability
- Required equipment
- Installation
- And of course - how to pay for it!

The last of these issues, cost, will no doubt be the driving force behind any solution. However, perhaps the most serious logistical problem will be finding a suitable location to mount an HDTV antenna and transmission line. The majority of all towers in the U.S. were built over 20 years ago and were designed to satisfy codes which are not in use today. Unlike the initial installation where the tower was designed to handle the specified equipment, now the mechanical constraints of the tower will determine the allowable equipment. [1,2,3,4]

The following pages present a unique flowchart methodology for studying the items listed above. They show the process by which HDTV RF system decisions should be made and how they are interrelated. The flowchart consists of a one page condensed overview outlining the overall process and paths. It shows each step that must be performed in a generalized manner, several operations may be contained in each box. The items are then divided into six sections and the process is fully expanded. Further breakdown of some boxes is necessary and will require separate consideration.

Each box on the flowcharts indicates an activity which must be performed at the point shown. The charts use typical programming flowchart symbol definitions as shown below.



SECTION 1
REGULATIONS, SCHEDULING & LICENSING

This section examines the issues related to how, when, and why the station will implement HDTV. Before any budgets can be prepared, the station must fully understand; when HDTV will begin, how the licensing process will operate, what the available channel allocations are, how industry capacity might effect obtaining equipment, and what method of signal delivery and production will be used within the station. At the end of this section a decision must be made to continue on with the process or wait.

Many of these items are informational in nature and can be extracted from various industry sources (NAB, MSTV, and various magazines). Others, like determining the studio implementation method, are as complex as the RF system choices made in this flowchart. They will require significant research. (5)

SECTION 2
COVERAGE ISSUES

Determining the desired optimum coverage for the new HDTV service is performed in section 2. To define the optimum coverage area, we first determine NTSC coverage and audience and then compare it to the possible HDTV audience. The NTSC and HDTV audiences may not be the same since population shifts may have occurred or are expected to occur. Once the audience is determined, a coverage area is described and calculations are made to establish the ERP and height required to cover it. A comparison of these values is made to the allowable maximums and a desired optimum coverage area is defined.

This optimum area may or may not be obtainable with the RF system to be defined. It indicates the total desired area. The final coverage area is dependent on the choices which follow.

SECTION 3
TOWER USE AND ALLIANCE ISSUES

Section 3 examines the logistical questions which are critical to finding a location for the new HDTV antenna and transmission line. It defines some of the alternatives and initiates discussions on tower options. Multichannel operation may have significant benefits by reducing the total aperature needed by several stations and this option should be pursued with other stations in the market at this point.

Once the tower and possible other tenants have been defined, the applicable tower standards must be determined. Some standards make reference to a national standard such as ANSI/EIA RS222E, others are unclear and/or modify these standards based on local conditions. After defining the standards an initial tower stress study is performed with the existing equipment on the tower (see section 5). On the second loop through the program an HDTV system is added in section 4. Note that the program loops back near the beginning of section 3 when either the tower is not modifiable or the system doesn't meet requirements. This point is chosen because a full re-examination of tower and multi-channel possibilities may be required to define an acceptable system.

SECTION 4
SYSTEM PROPOSAL

In this section the results of the initial stress study and the desired coverage area are input and an HDTV RF system proposal is formulated. This is the heart of the feasibility study process. Decisions are made on the new HDTV antenna, transmission line, combiner, filter and transmitter based on the available weight and windload capacity. The other variables of power handling, pattern performance, and bandwidth are also considered. The process starts with finding an antenna which, when mounted in the available tower space, best fits the desired azimuth coverage pattern. The antenna is then fed with several transmission line types and sizes. Total system gains and windloads are calculated and a combination is chosen which provides an acceptable ERP and pattern within the available capacity. The resulting antenna/line combination is coupled to a transmitter through any required combiners and filters and the necessary transmitter output power is calculated. The system parameters are then tested for power handling and a decision is made as to whether the system meets the requirements. This section loops back within itself since the proposal must perform as a system and any one parameter may initiate a re-examination of the trade-offs. The process will thus typically require several iterations.

The output from this section consists of a resulting coverage contour, a list of proposed equipment, and a table of mechanical data to input to the stress analysis on the second pass-through.

SECTION 5
STRESS ANALYSIS

A computer analysis of mechanical stresses on the tower due to the existing equipment and proposed new HDTV equipment is performed in this section. The analysis determines whether the tower arrangement exceeds the applicable stress standards. If the tower does exceed the standards, possible modification scenarios are examined and estimated.

An important part of the stress study is the field inspection and inventory of every item on the tower. Member meausurements are also made at this time if sufficiently detailed drawings are not available. The inventory data is prepared for entry into the program by splitting into two categories, one for discrete items (antennas, lights) and the other for linear items (lines, ladders). Any information which is input to the program on the first loop will remain as entered on subsequent loops unless modified by the HDTV system proposal. The proper wind and ice loading criteria for the location are entered and the analysis is run. The program outputs load values for the guys, legs, members, and anchors which are checked against the standards. The program continues if the tower does not require modification or if the modifications are acceptable.

SECTION 6
SYSTEM SELECTION AND REPORTING

When an acceptable tower arrangement has been determined, final system selection and trade-off occurs. Methods for adding the equipment and modifying the transmitter building should be determined. A plan for installing the equipment should be developed and any off-air time negotiated. At this point the overall system technical performance is tabulated and a decision is made as to whether the system performs as desired. This includes all the ancillary items in the system as well as installation. If performance is acceptable, the system costs are determined and a decision as to whether the costs are reasonable is made. Failure at either of these decisions returns to section 3. Once the final system has been decided, detailed bills of material, schedules, layouts and a report are generated.

CONCLUSION

It is hoped that this flowchart will aid stations in their efforts to implement HDTV and find a feasible solution to their unique installation problems. Keep in mind that these decisions should not be made in a vacuum since the other stations in the market may be able to help reduce your system costs by pooling tower resources. The author would like to thank K. Cozad of Andrew Inc. and J. Windle of Stainless Inc. for their assistance in reviewing the flowchart.

REFERENCES:

1)T. Vaughan and J. Banker: *"Antenna and Transmission Line for the Simulcast Period,"* 1992 HDTV World Conference

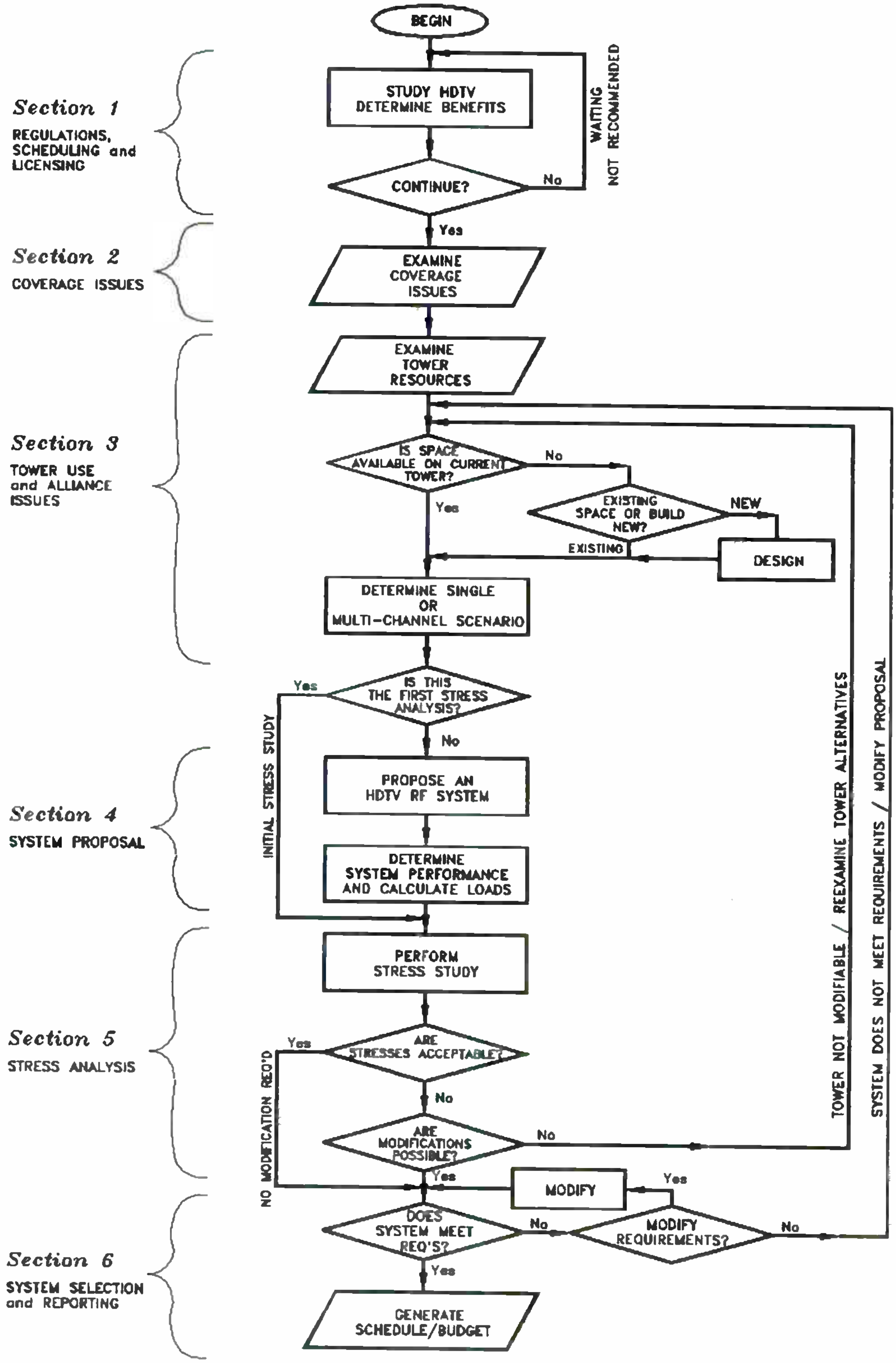
2)K. Cozad: *"Practical Implementation of Transmission Lines and Antennas for the Simulcast Period for NTSC/HDTV"* 1993 NAB Engineering Conference

3)T. Vaughan: *"Effect on Tower Stress Due to the Selection of Antenna and Transmission Line for HDTV,"* 1993 HDTV World Conference.

4)J. Windle: *"Preparing Your Tower for HDTV,"* 1993 HDTV World Conference

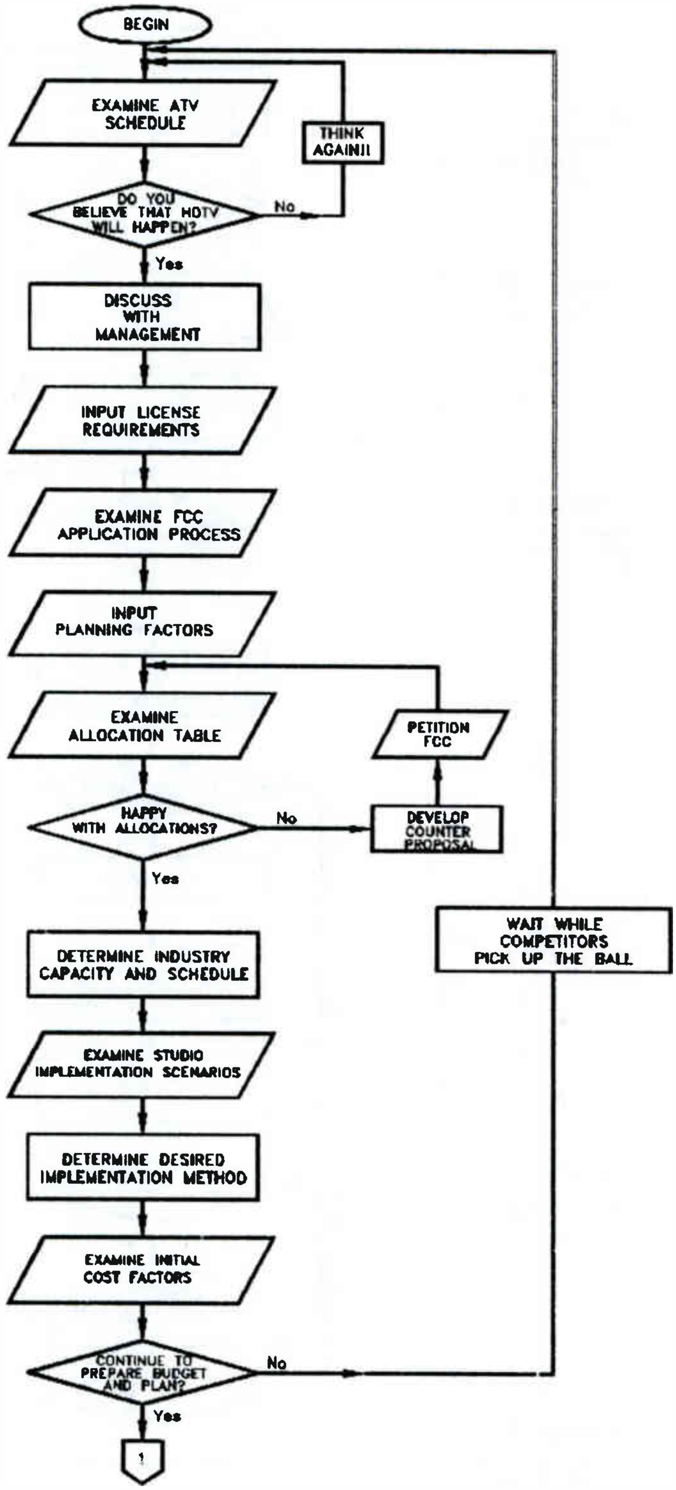
5)S. Weiss, R. Stow: *"NAB 1993 Guide to HDTV Implementation Costs"* 1993 NAB PRESS

OVERVIEW



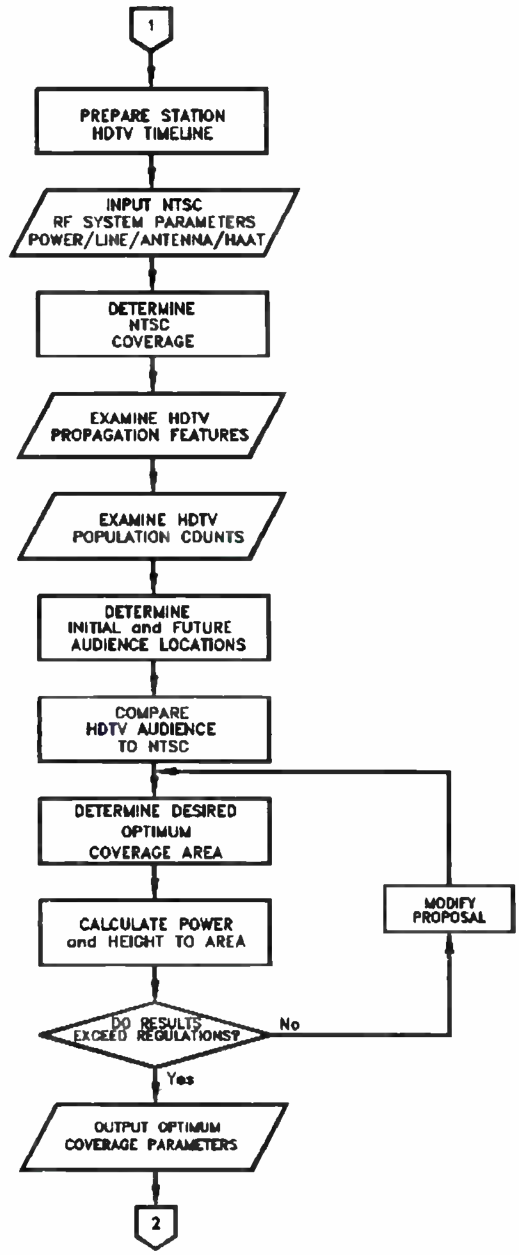
HDTV RF SYSTEM FEASIBILITY FLOWCHART

SECTION 1
REGULATIONS, SCHEDULING AND LICENSING



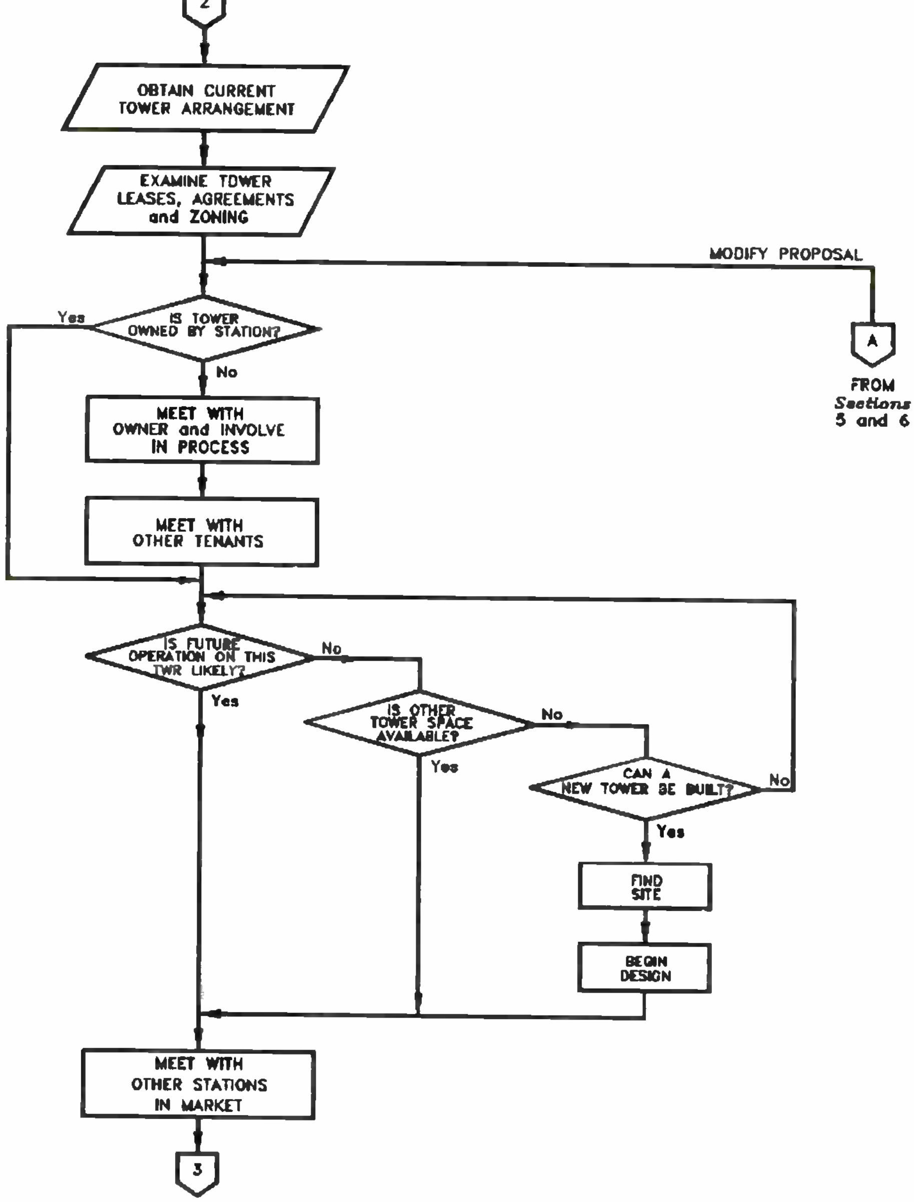
HDTV RF SYSTEM FEASIBILITY FLOWCHART

SECTION 2
COVERAGE ISSUES



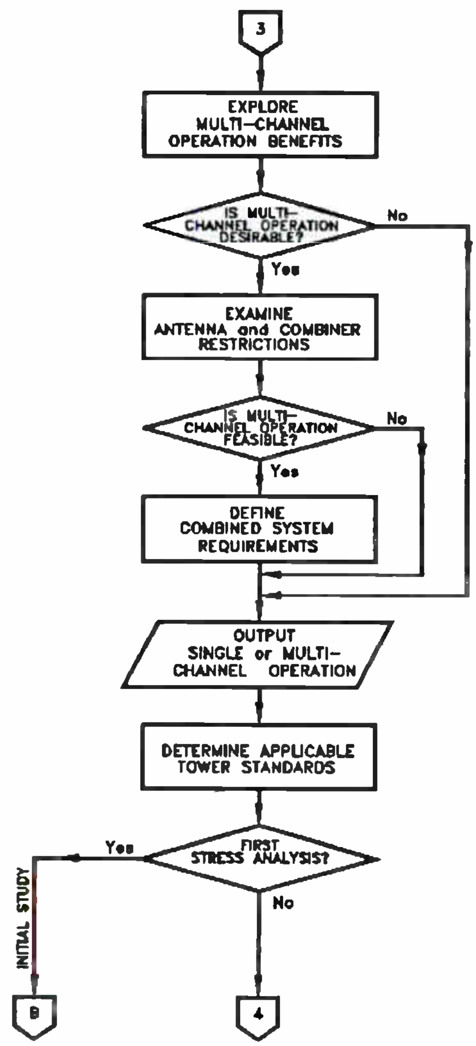
HDTV RF SYSTEM FEASIBILITY FLOWCHART

SECTION 3 (PG 1 OF 2)
TOWER USE AND ALLIANCE ISSUES



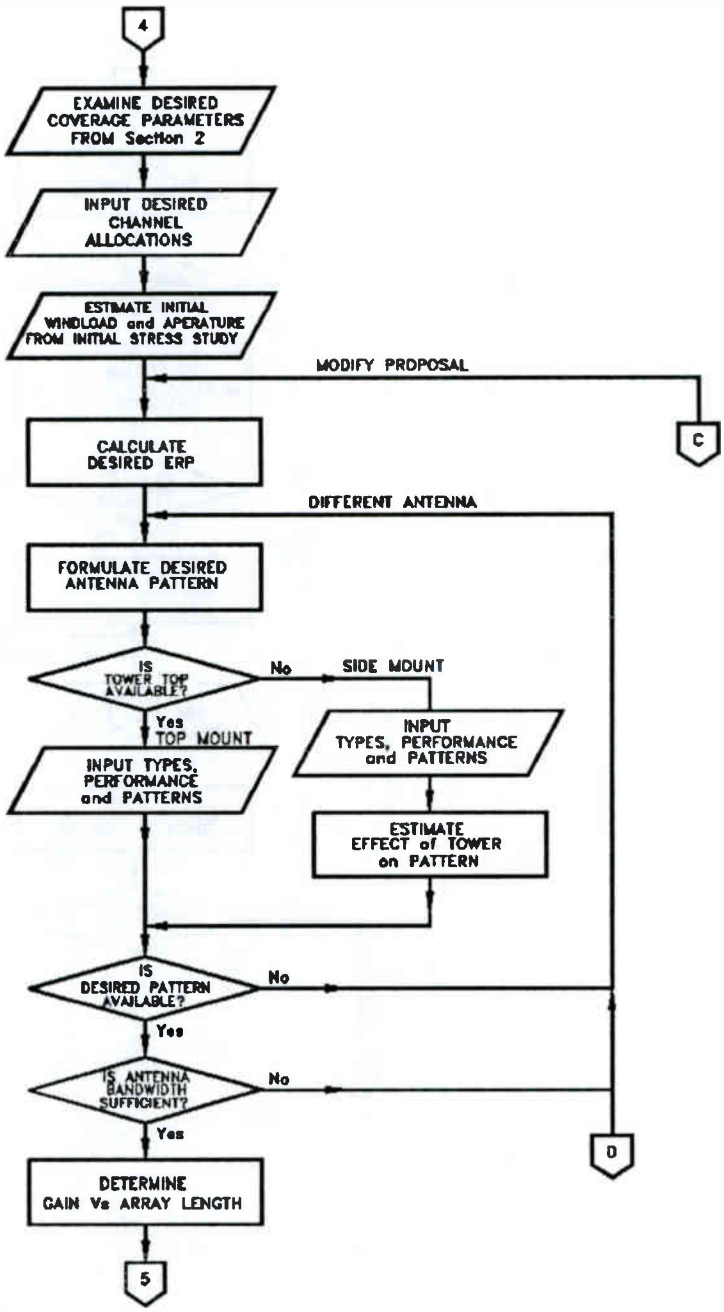
HDTV RF SYSTEM FEASIBILITY FLOWCHART

SECTION 3 (PG 2 OF 2)
TOWER USE AND ALLIANCE ISSUES



HDTV RF SYSTEM FEASIBILITY FLOWCHART

SECTION 4 (PG 1 OF 3)
SYSTEM PROPOSAL



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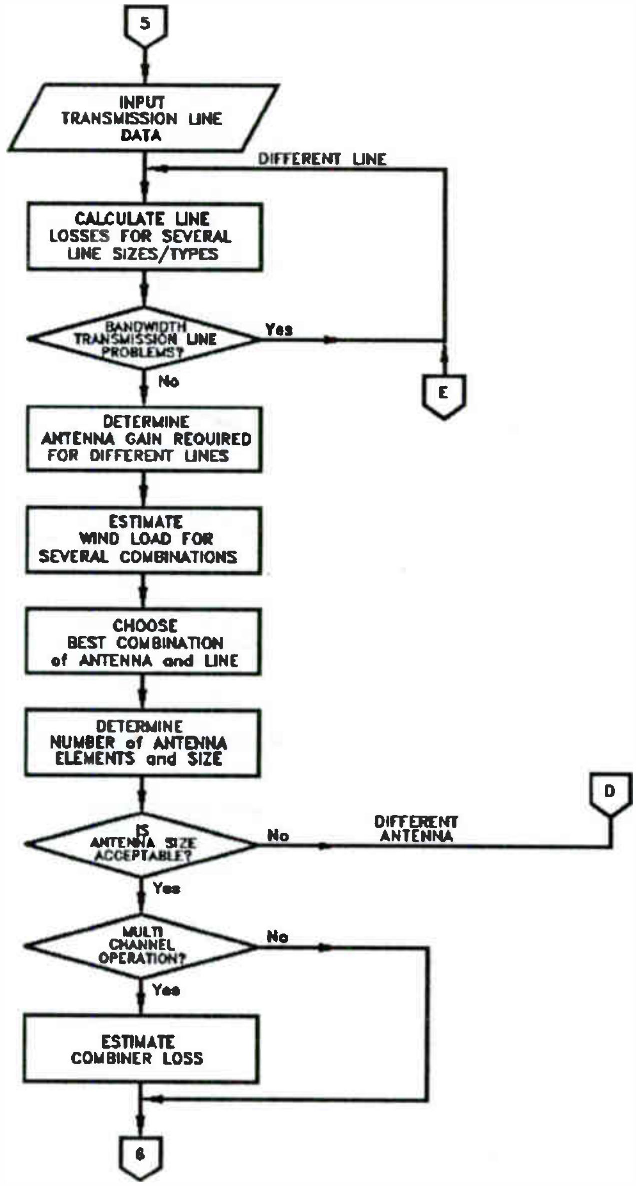
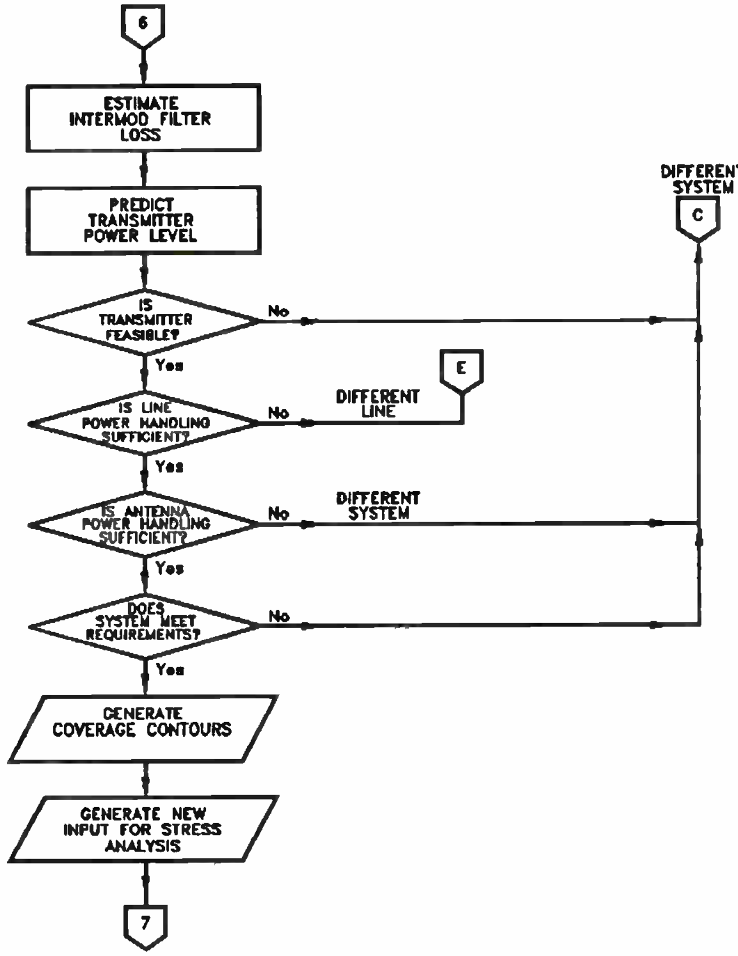


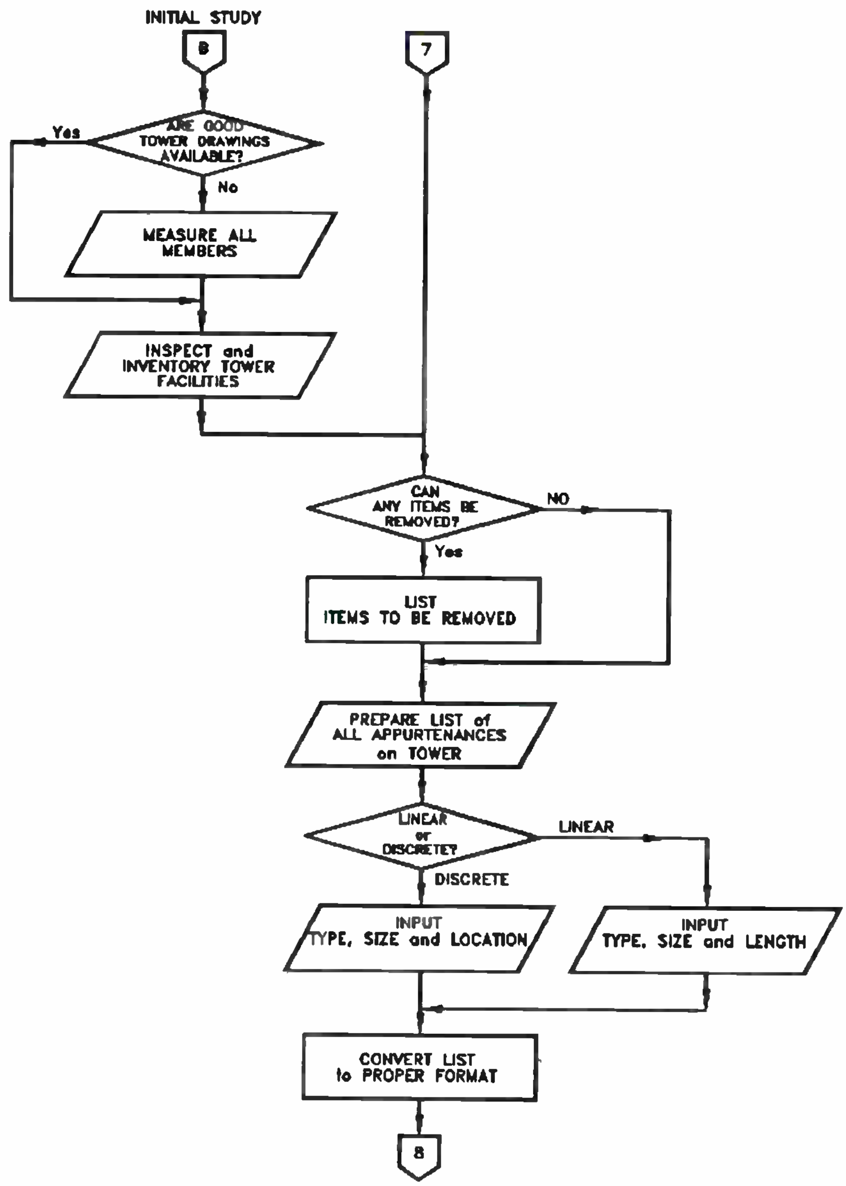
HDTV RF SYSTEM FEASIBILITY FLOWCHART

SECTION 4 (PG 3 OF 3)
SYSTEM PROPOSAL



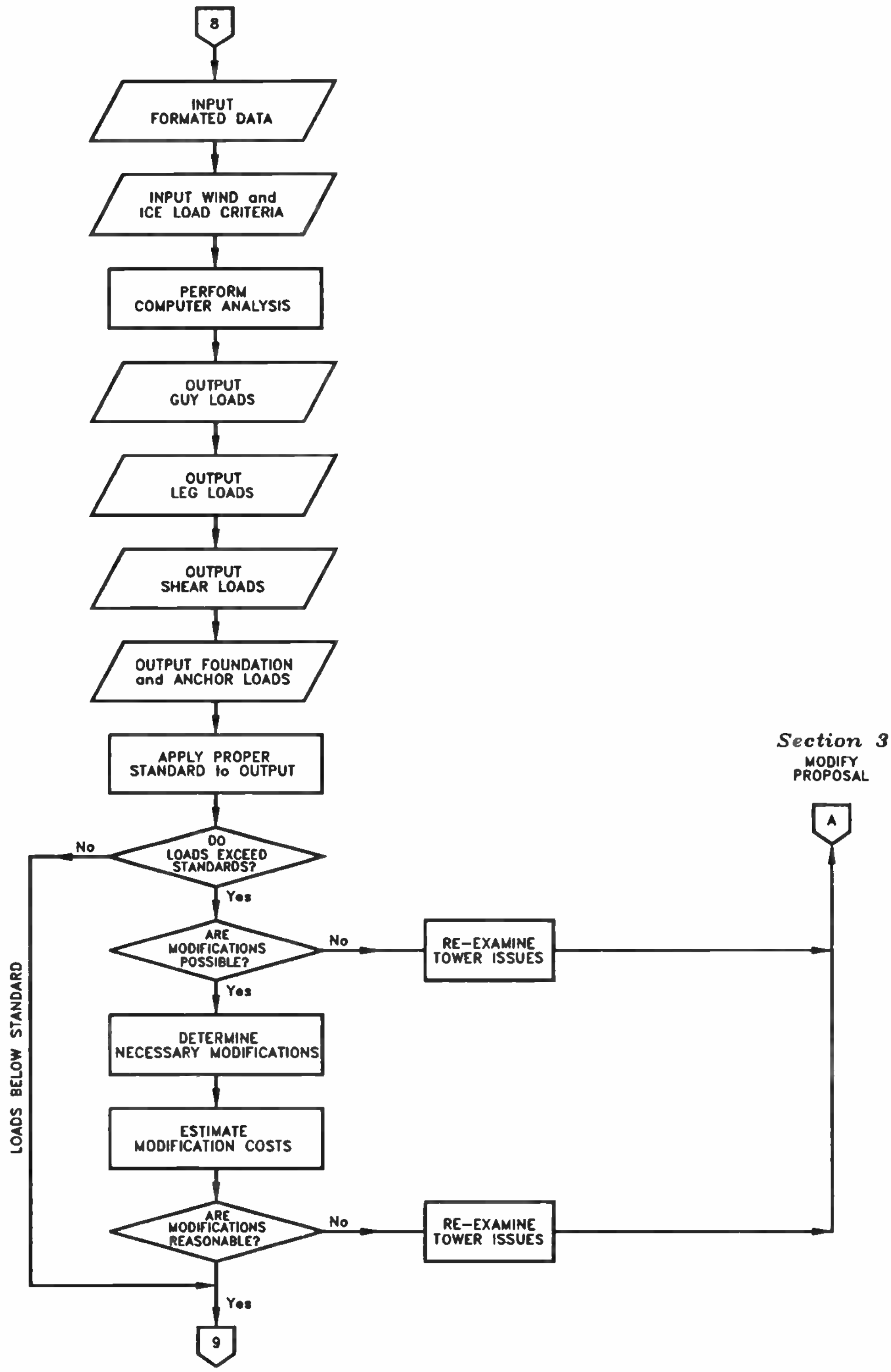
HDTV RF SYSTEM FEASIBILITY FLOWCHART

SECTION 5 (PG 1 OF 2)
STRESS ANALYSIS



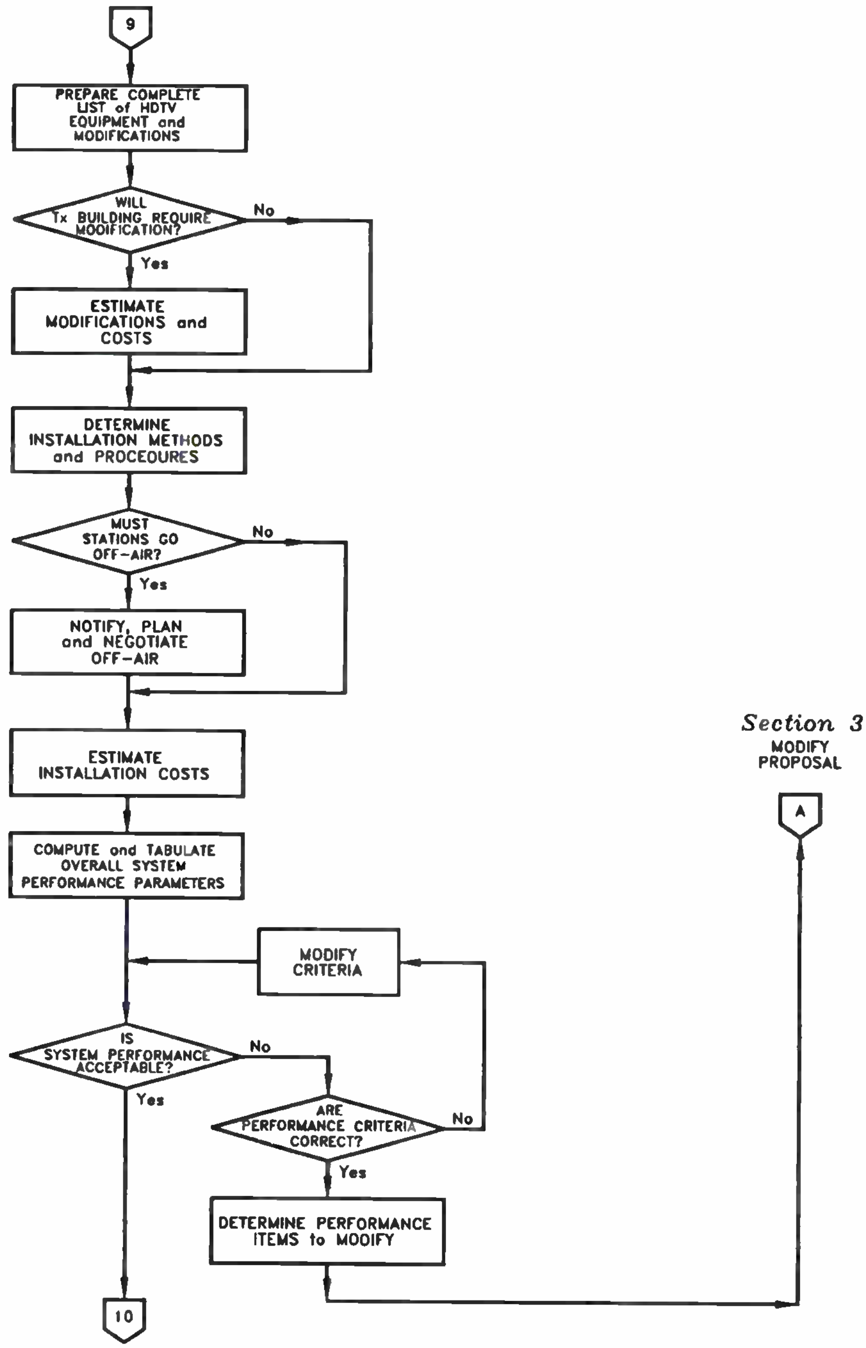
HDTV RF SYSTEM FEASIBILITY FLOWCHART

SECTION 5 (PG 2 OF 2)
STRESS ANALYSIS



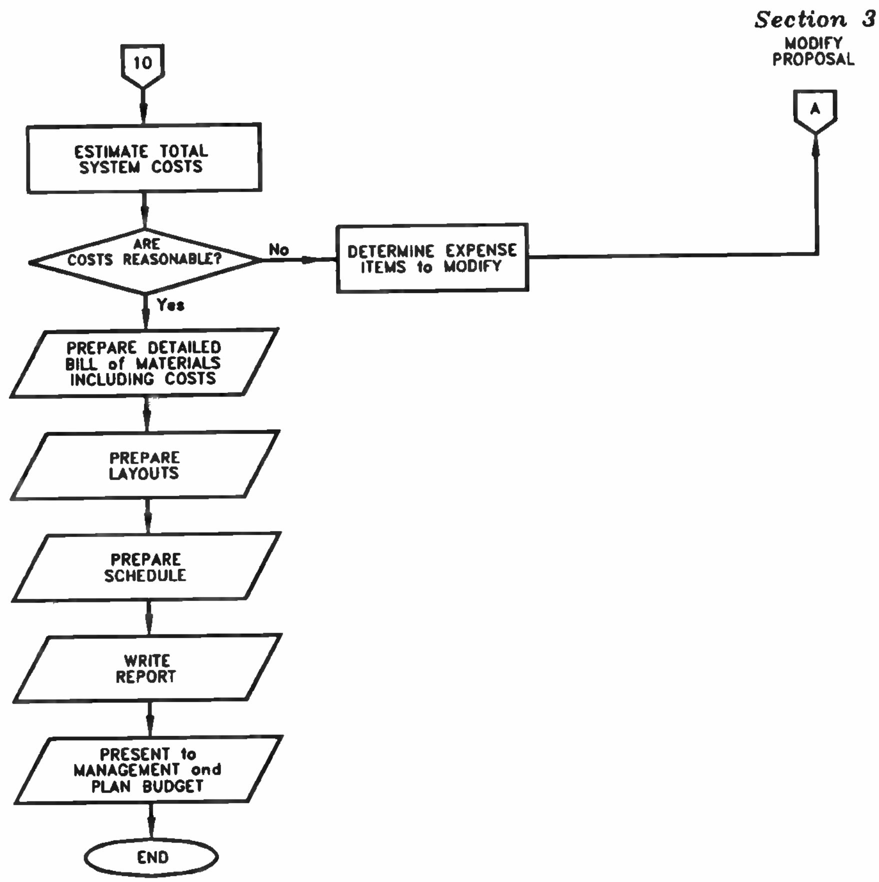
HDTV RF SYSTEM FEASIBILITY FLOWCHART

SECTION 6 (PG 1 OF 2)
SYSTEM SELECTION AND REPORTING



HDTV RF SYSTEM FEASIBILITY FLOWCHART

SECTION 6 (PG 2 OF 2)
SYSTEM SELECTION AND REPORTING



HDTV RF SYSTEM FEASIBILITY FLOWCHART

HDTV COVERAGE OPTIMIZATION USING ADVANCED TECHNIQUES

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Abstract

A version of the TIREM propagation model has been integrated with detailed terrain data and the latest U.S. Census data to predict accurately the number of potential viewers within a particular signal strength contour. This method of coverage prediction has the principal advantage that potential audience, rather than geographical area, is being maximized. Using this technique, optimized antenna patterns can be developed, or can be synthesized using "off the shelf" antennas with a particular combination of electrical and mechanical beam tilts, to provide saturation coverage to a pre-defined potential audience. Traditional methods of coverage prediction are reviewed and discussed. The advanced method is then presented, using an example that incorporates specific topographies. The benefit of the advanced technique is explained in terms of a potential viewership comparison for this example. Various presentations of the results of the technique are included.

COVERAGE

Providing superior coverage is one important goal of broadcasters. Of course, the FCC requires the provision of a *principal community* signal level[1] over the city of license, but obviously broadcasters want to provide a usable signal to as many potential viewers as possible. Design of a transmitting antenna system that provides the highest signal levels to the greatest number of viewers, with the least amount of energy wasted into the atmosphere, vacant land, or bodies of water, is essential to achieving this goal.

If a broadcast station is located in the plains and the audience is evenly distributed about the transmitter site, the choice of antenna is obvious. If the transmitter site is in a less than ideal location, and if there are terrain obstructions that effectively block the signal in certain directions, the choice of an optimum antenna requires some serious thought. The costs in time and effort to select the correct antenna are relatively small, when compared to a $250,000 antenna system that is expected to last 15 or 20 years. A serious mistake made in the planning stage probably will not be corrected for many years and could be seized upon by the competition in attempting to establish that the coverage of a competing station is sub-standard.

TRADITIONAL METHODS OF ANTENNA SYSTEM SELECTION

A traditional, moderately low-technology method of antenna pattern optimization might begin with determining what coverage areas are important to the station management. If the station carries Spanish language programming, for instance, there will obviously be certain communities, having large Hispanic populations, that will need to be covered. Beyond that, however, management may want to ensure flexibility. If the station were later sold, maximum coverage generally will be more valuable to the buyers than focused coverage.

Often, a prospective transmitter site has already been identified. It is part of the engineer's job to provide management with a list or map, showing those areas that potentially could be covered from that site. One way of showing potential coverage areas is the shadow map. To construct a shadow map, a terrain profile is drawn for each azimuth from the site to determine the line-of-sight in that direction. The potential terrain-limited coverage can then be plotted on a map. While terrain profiles can be drawn from topographic maps, today it is much more common for computers to generate them from a terrain database. With appropriate software, the computer can also integrate the process of transferring the shadowed areas to a map. Figure 1 shows such a map. This map consists of terrain profiles drawn at each 1° of azimuth. Areas covered by lines are shadowed from the transmitter site and probably would not receive reliable service.

Once the areas of potential coverage are identified, it is necessary to determine what antenna parameters are necessary to serve each optimally. Certain communities may be identified for preferential treatment, and many communities may require a similar antenna for optimal service. After examining a large number of such communities, the commonalties can be exploited in synthesizing an antenna pattern that will be close to optimal for all (or at least most) of them.

In determining the optimum transmitting antenna for a given community, one must determine the depression angle from the transmitter site to some point in that community, usually its so-called "reference coordinates." The reference coordinates are generally those of the main post office, and can be obtained from a number of sources, including the U.S. Census, the National Atlas of the U.S., and the FCC Rules. Communities that are not incorporated may not be explicitly listed, so their reference coordinates must be obtained from another source or directly from a topographic map, if necessary. The depression angle from the transmitting antenna is obtained from the distance to the reference coordinates and the difference between the center of radiation and the estimated receive antenna elevations. Using the 4/3 Earth radius model, it is approximated by the formula:[2]

$$Ø = (^1/_D) \tan^{-1} [\Delta E - 16978 \sin^2 (5.89*10^{-5} * D)]$$

where D is the distance between the sites (in kilometers) and ΔE is the difference in elevation between the sites (in kilometers).

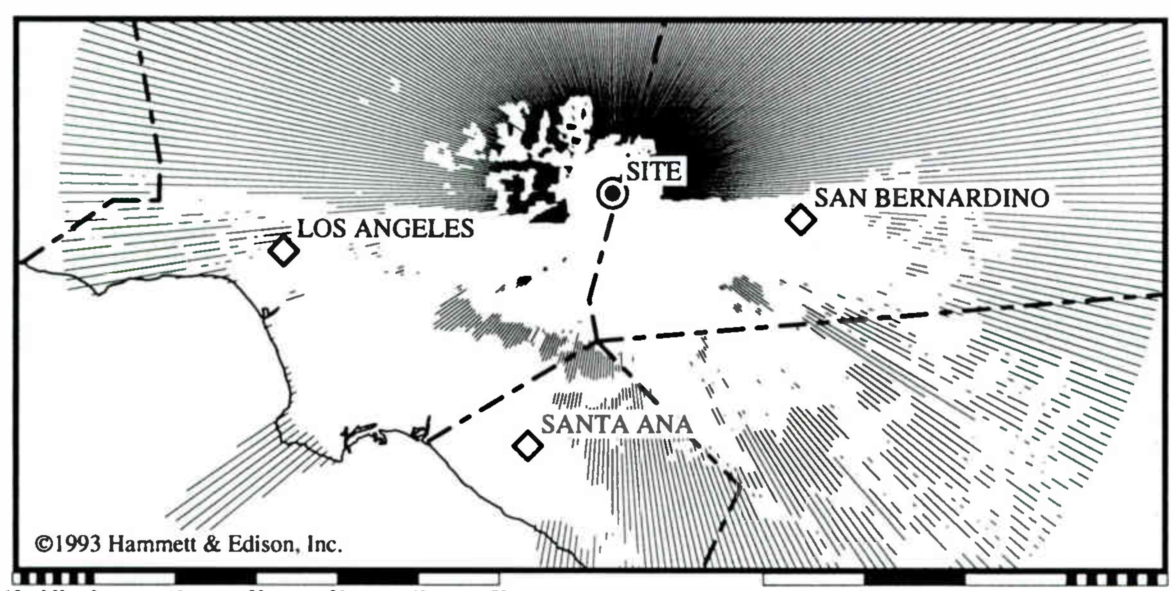


Figure 1. A terrain shadow map. This map shows significant terrain obstruction (shadowing) to the north of the transmitter site. Some obstructions to the south and east are also apparent.

For the transmitter site described in Figure 1, the depression angles to 25 select cities were calculated that are within the potential coverage area. Figure 2 is a plot of these depression angles with respect to their respective azimuths from the transmitting site. The most obvious feature of this figure is that only azimuths between 105° and 265° are included. This implies that radiation in other directions would be wasted. It also implies that a directional azimuth pattern should be used and offers the antenna manufacturer some latitude, since the shape of radiation envelope is not critical outside the 160° arc of interest. Two communities of particular interest have been highlighted for this hypothetical station: the city of license and the most populous nearby city (Los Angeles). In considering which antenna pattern to use, it is necessary to ensure that these two communities are appropriately treated. The shadow map shows terrain obstructions to the south and east of the transmitter site, so an azimuth pattern with lobes toward the east-southeast and west-southwest, and a relative minimum toward the south seems an appropriate first choice. Since the range of azimuths being covered is from approximately 105° to 265°, the axis of symmetry of this azimuth pattern would bisect that arc at about 185°.

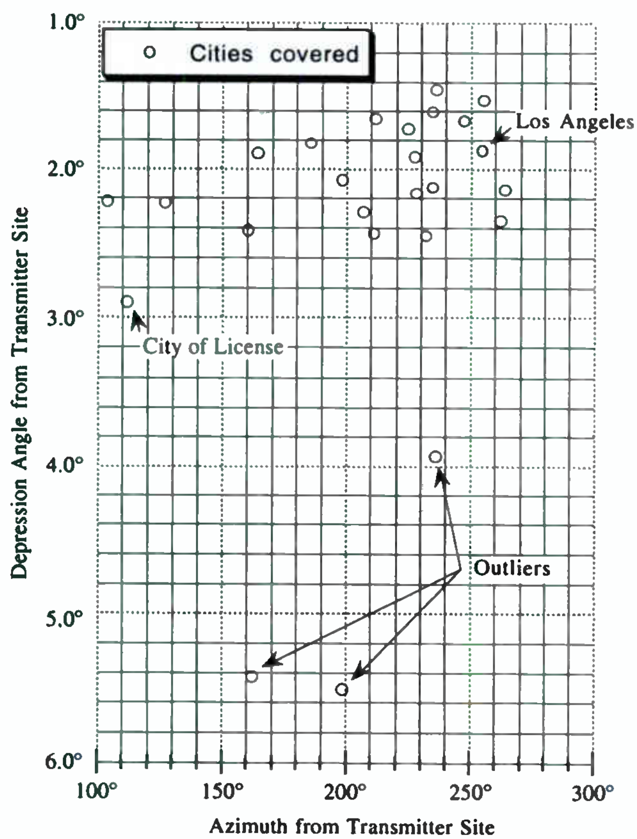


Figure 2. Depression and azimuth angles from transmitter site to 25 communities in the potential coverage area.

Another feature of Figure 2 is that nearly all of the cities are clustered at depression angles between 1.4° and 2.7°; there are only three "outliers." Since these outliers are at relatively steep depression angles, they are expected to be close to the transmitter site and could be treated by using "null fill," if necessary or desirable. The range of depression angles (neglecting the outliers) is 1.3°. This implies that the elevation beamwidth should be about 1.3°. A broader elevation pattern than that would again imply wasted power.

Electrical Tilt Only

At this point, it is a common practice to calculate the average of the depression angles and order an antenna with that amount of built-in electrical beam tilt. In the example case, the average depression angle is about 2°. Two of the existing stations at the site described in this example selected antenna elevation patterns sufficiently wide to cover most of the cities shown in Figure 2, and employed electrical tilt of somewhat less than 2°. The choice of a tilt of less than the average presumably was to generate stronger signals in the more distant population centers.

The selected azimuth pattern in each case was a broad cardioid, covering the range of azimuths shown in Figure 2, with an axis of symmetry of slightly west of 185°. The choice of an axis of symmetry of greater than the average presumably was to generate stronger signals in the Los Angeles area. As will be shown, these elevation and azimuth pattern choices were not optimal.

Electrical and Mechanical Tilt

While use of the average depression angle provides a reasonable "first-cut" approximation of the required elevation pattern, some additional effort can result in a pattern that fits the depression angle data much more closely. Some sort of curve fit will describe this depression angle data fairly well. Figure 3 shows the data fitted using a cubic polynomial, although a French curve or some other numerical technique likely would also work well. With the desired antenna azimuth pattern and elevation beamwidth already determined, the goal is to approximate the fitted elevation curve as closely as possible using electrical and/or mechanical beam tilt.

Application of mechanical beam tilt has the effect of adding a sinusoidal variation of the beam tilt with azimuth. An initial guess for the correct amount of mechanical tilt to employ is one-half of the peak-to-peak variation in the required tilts. In the example case, the minimum amount of tilt required is about 1.9° and the maximum is 2.6°. Thus, an appropriate amount of mechanical tilt might be half the difference in the extrema (2.6 - 1.9)/2, or 0.35°. This implies 2.25° of electrical tilt would be required. Figure 3 shows that incorporation of these suggested tilts results in a pattern that closely approximates the curve-fit.

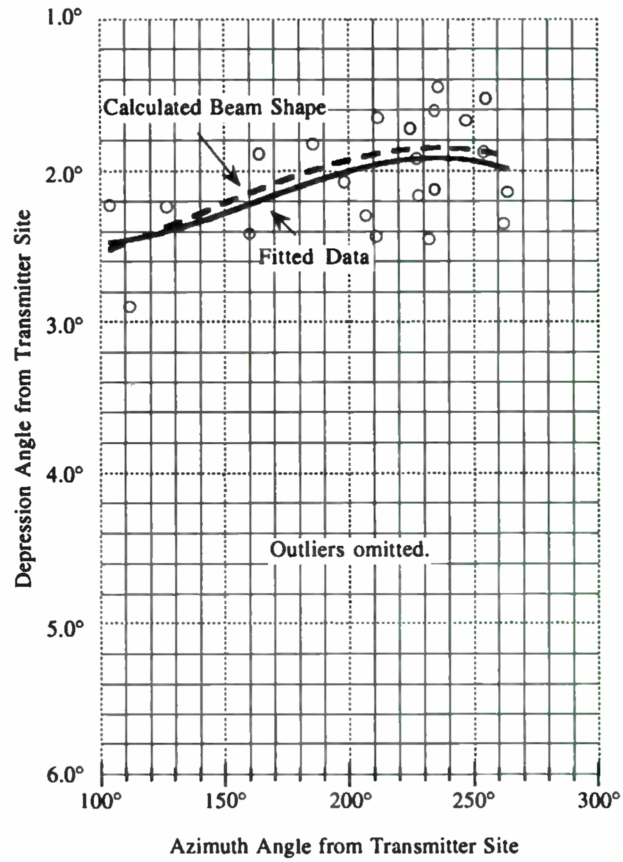


Figure 3. The solid line shows the cubic polynomial fit of the depression angle data shown in Figure 2. The dashed line shows that the antenna beam shape obtained by incorporating appropriate amounts of electrical and mechanical beam tilt approximates this fitted data.

ADVANCED METHOD OF ANTENNA SYSTEM SELECTION

A more accurate alternative to the above labor-intensive approaches can be employed. This approach makes use of sophisticated coverage-prediction software, and U.S. Census and United States Geological Survey data to eliminate many of the estimates and guesswork, replacing them with a system of precise computational optimization.

Details of the Software

Various algorithms have been used for many years to predict path loss between two points over specified terrain profiles. The most familiar "cookbook" of such algorithms is *Transmission Loss Predictions for Tropospheric Communications Circuits*,[3] otherwise known as *National Bureau of Standards Technical Note 101* or the *Longley-Rice Model.* A proper selection of the appropriate algorithm from those offered in that publication requires expert engineering judgment of the pertinent terrain profile between the two points. To prepare a sufficiently detailed map of the coverage in the region around each station would require the analysis of literally thousands of separate and distinct terrain profiles. It is therefore a practical impossibility to prepare a realistic coverage map solely by using manual computation based on Technical Note 101.

To provide a computerized method for predicting coverage, the National Telecommunications and Information Administration (NTIA) developed an integrated system having a number of different propagation algorithms imbedded in it. The basic NTIA publication, *Master Propagation System Users' Manual,*[4] contains algorithms for groundwave and tropospheric propagation prediction from 10 kHz to 300 GHz for a variety of conditions. However, only its terrain-integrated rough-earth model (TIREM) is deterministic. The use of TIREM, together with other programs and appropriate databases, can provide a detailed map of predicted signal strength throughout an area, as contrasted with methods such as those used by the FCC, which treat the terrain in a statistical manner and truncate the available terrain information to produce maps that are merely approximations of coverage.

The TIREM program automatically selects the most appropriate propagation algorithm for computing path loss between any two points. This "expert system" substitutes for the judgment of an expert engineer, who would otherwise be required to select the appropriate algorithm on each of the thousands of paths. TIREM selects the proper algorithm based upon its analysis of the profile of each path. For paths that have characteristics appropriate to more than one algorithm, TIREM automatically calculates a weighted average between the two most applicable algorithms.

The automated preparation of coverage maps depends not only upon the availability of TIREM but also upon the use of a suitably detailed digitized terrain database, such as that available from the U. S. Geological Survey. This database provides terrain elevations throughout the United States at points located on a rectangular grid with a spacing between points of three arc-seconds (approximately 250 feet apart.) Together, the TIREM program and the digitized terrain database accurately predict the time-invariant path loss in the region surrounding a station. When combined with a suitable "driver" program that incorporates the radiation characteristics of the station, one can easily calculate the signal level at any point.

In order to optimize the potential audience from a particular site, knowledge of the population distribution in the region surrounding the site is also required. The U.S. Bureau of the Census provides a database that shows United States population distribution in electronic form, as determined by the most recent Census (1990, as of this writing.) Each database entry consists of the center coordinates of a Census Block (approximately equivalent to a city block), along with the population contained within that Block. In a very rural area, the Census Block may contain only one person; in a dense urban area, the Block may contain 400–500 people. By identifying the Census Blocks with centroids lying within a given signal strength contour, the number of people potentially receiving that level of service can readily be calculated.

Antenna Pattern Optimization Routine

The mechanics by which potential population coverage can calculated at a given signal level have been presented. Three parameters must also be specified in order to determine an optimum antenna configuration. First, one must specify what signal strength contour or contours to use in defining the boundaries of what is and is not acceptable coverage. The optimum antenna for each signal level may be different. The FCC offers three contours: "Principal Community" (formerly called "City Grade"), "Grade A," and "Grade B" as indicators of service quality. While these values do provide some indication of the quality of receiving installation required, they generally over-represent quality coverage. Therefore, somewhat higher values are suggested. For example, the Principal Community, Grade A, and Grade B levels for a UHF television station are 80, 74, and 64 dBu, respectively. Because typical receiving systems will use back-of-set antennas, those values will provide marginal pictures, particularly if only horizontal polarization is employed by the transmitting station. Therefore, it is suggested that 100, 90, and 80 dBu signal strength levels are more appropriate, and that the number of persons within those signal strength levels should be maximized.

Second, the appropriate half-power elevation beamwidth and azimuth pattern must be determined. With computing time being relatively inexpensive, it is certainly possible to use a strict trial-and-error approach to determine this value. In practice, however, an initial guess is made and used to iterate toward the optimum value. While it is possible to synthesize an ideal antenna radiation pattern using this technique, it is generally cost-prohibitive (or physically impossible) actually to build an antenna based on such a calculated pattern. The major antenna manufacturers publish catalogs of their "off-the-shelf" antenna patterns, and it is a simple matter to enter these patterns into a computer and use them to test various alternatives.

Finally, the appropriate amounts of electrical and mechanical beam tilt must be determined. Again, while it possible to determine these values completely by trial and error, it is more usual to make an initial guess and iterate toward the optimum values. A custom driver for TIREM takes specified values of maximum ERP, electrical and mechanical beam tilts, and antenna azimuth and elevation patterns, and calculates the appropriate ERP in each direction for TIREM to use. As the population within each coverage contour is counted, different patterns and combinations of electrical and mechanical beam tilt can be tried, until the covered population is maximized.

For the example design, the resulting computer-optimized values are somewhat different than those found using traditional methods. Using the advanced method, the optimum amount of electrical tilt is found to be 1.25°, while the optimum mechanical tilt is 0.65°, toward an azimuth of 215°T. The optimum axis of symmetry for the azimuth pattern is 190°T. There is also a considerable savings in the amount of time required to determine the optimum parameters.

COMPARISON OF RESULTS

Using the software described previously, it is possible to quantitatively compare antennas designed using traditional methods with one designed using the advanced method. Following the premise that it is the number of potential viewers that should be maximized, the appropriate method would be to count the population within each of the 100, 90, and 80 dBu contours for each antenna system. Figure 4 shows a comparison of potential viewership covered by the different antenna systems developed using the two methods. It can be seen that the antenna system optimized with the advanced method provides

Manual Method				TIREM Method		
100 dBu	90 dBu	80 dBu	County	100 dBu	90 dBu	80 dBu
5,476,885	6,795,715	7,154,306	Los Angeles	5,627,593	6,823,006	7,170,996
1,838,998	2,064,584	2,168,910	Orange	1,861,080	2,082,095	2,178,202
257,386	485,005	669,475	Riverside	481,569	676,723	800,635
214,445	924,261	1,004,057	San Bernardino	937,159	1,009,599	1,039,787
	2	1,179	Ventura		2	1,179
7,787,714	**10,269,567**	**10,997,927**	**Total**	**8,907,401**	**10,591,425**	**11,190,799**

Figure 4. A comparison of the populations covered by antennas designed using a traditional (electrical tilt of 1.9° only) method and the advanced (TIREM) method.

a calculated 100 dBu population coverage improvement of about 14.4%. At the 90 and 80 dBu levels, the improvements are smaller: 3.1% and 1.8%, respectively. While these improvements are less dramatic, they still represent hundreds of thousands of people. Experience has shown, moreover, that even larger improvements can be produced in other situations.

HDTV APPLICATION

The technical details of the Advanced Television ("ATV") system to be implemented in the U.S. have not yet been fully decided by the FCC. However, reasonable assumptions can be made concerning the necessary desired-to-undesired ("D/U") signal ratios that will be required for co- and adjacent-channel operation. The FCC has prepared a "sample" table of allotments for ATV operation in many places.[5] They have not, however, paired the ATV channels with specific existing NTSC licensees. It is an interesting exercise to determine the optimum pairing of NTSC/ATV channels. For some initial estimates, Hammett & Edison has used the values published in the *Record of Test Results for the DigiCipher™ HDTV System*. The tests conducted by the ATTC involved determining D/U ratios for each of the possible combinations of HDTV and NTSC into HDTV and NTSC signals. The approximate D/U ratios obtained in that study for interference into ATV transmissions (32 QAM mode) were as follows:[6]

Interfering Channel	ATV-to-ATV	NTSC-to-ATV
0 (Co-channel)	15 dB	8 dB
-1 (Lower Adj.)	-23	-30
+1 (Upper Adj.)	-23	-23
-2	-38	-44
+2	-37	-33
+4	-57	-57
-7	-58	-58
+7	-60	+58
-8	-58	-58
+8	-63	-63
+14	-63	+58
+15	-63	+58

Using TIREM and the associated terrain and population databases to calculate the signal levels from combinations of NTSC and ATV transmitters, taking into account the required D/U ratios, both the terrain- and interference-limited coverage of each prospective ATV channel can be determined. This method provides "hard" population numbers for coverage comparison, and also shows where interference will likely occur and how many persons could be affected by it. From this information, a table of preferred ATV channel assignments can be prepared for a given area.

This method could be used to achieve the closest possible common coverage between the NTSC and ATV signals, while minimizing interference due to the additional ATV channels. By making changes in the input parameters, such as the antenna patterns, the predicted areas of interference can be greatly reduced, or perhaps even "steered" into unpopulated areas or to areas presently receiving interference.

Figures 5 and 6 show the terrain- and interference-limited coverage for two ATV channels allotted to the San Francisco Bay Area in the sample FCC plan. Notice that the coverage on Channel "B" in the San José area is quite limited, when compared to that of Channel "A." This limitation is due to predicted interference from several existing NTSC and allocated ATV channels in that area operating on the "UHF Taboo" channels[7] with D/U ratios exceeding those presently permitted in FCC Rules or from the table at left. A comparison of the populations covered by each channel shows that the difference in potential audience between these two channels is 19.6%. Obviously, broadcasters should examine carefully the available ATV channel alternatives.

SUMMARY

The accuracy and precision of the presently-available TIREM propagation modeling software is remarkable. When used with the available detailed terrain and population databases, it provides a means of accurately predicting potential viewership, rather than merely geographical coverage.

Antenna Pattern Optimization

The selection of the parameters of operation for TV transmitting antennas is a complex task. Use of the advanced coverage prediction techniques shows that specification of electrical beam tilt is, in many circumstances, not sufficient to gain maximum coverage from the antenna; use of mechanical tilt, sometimes with its own orientation separate from the line of symmetry of the azimuth pattern, can provide a significant improvement in coverage.

The measure for optimizing coverage from a TV antenna should be "saturation" contours of high signal strength, rather than the traditional Grade A or B contours. This is especially true as competing signals from cable and satellite begin to offer improved quality as well as quantity.

Application to HDTV

Based on reasonable assumptions of the eventual ATV interference criteria, it is clear that there are often great differences in the terrain- and interference-limited coverage of different ATV channels tentatively allotted to the same city. Since it is expected that digital transmission will yield more reliable picture quality within the coverage area of an ATV station, maximizing the extent of that coverage area is of primary concern. The analysis made possible by the development of the advanced coverage prediction techniques described above allows precise evaluation of the coverage differences.

CONCLUSION

The advanced techniques described in this paper for predicting terrain- and interference-limited coverage of TV stations have beneficial application for: 1) optimizing NTSC and ATV coverage of potential viewers, rather than just of land area, 2) evaluating possible ATV sites and channels, with recognition of sometimes severe limitations on coverage due to interference, and 3) optimizing service from possible booster, translator, and satellite stations.

The author wishes to thank Robert L. Hammett for his invaluable assistance in the HDTV aspects of this paper, and to recognize Robert P. Smith, Jr. for his development of the proprietary TIREM driver software used by Hammett & Edison, Inc.

ENDNOTES

[1] FCC Rules Section 73.685(a) [47 C.F.R. §73.685(a)].

[2] Davis, Elton, and Phil Tremper, Elevation and Depression Angle Tables (Washington, DC: FCC, 1964), p. iii.

[3] Rice, P.L., Longley A.G., *et al.*, Transmission Loss Predictions for Tropospheric Communications Circuits – Technical Note No. 101 (revised) (Washington, DC: GPO, 1967).

[4] NTIA, Master Propagation System Users' Guide (Springfield, VA: NTIS, 1983).

[5] FCC, Docket 87-268, Second Further Notice of Proposed Rule Making (Washington, DC: FCC August 14, 1992), Appendix D.

[6] Advanced Television Test Center, Record of Test Results for DigiCipher™ HDTV (Alexandria, VA: ATTC, 1992), pp. I-19-25 ff.

[7] FCC Rules Sections 73.698, Table II and 74.705.

Figure 5

Potential ATV Coverage on Channel "A"

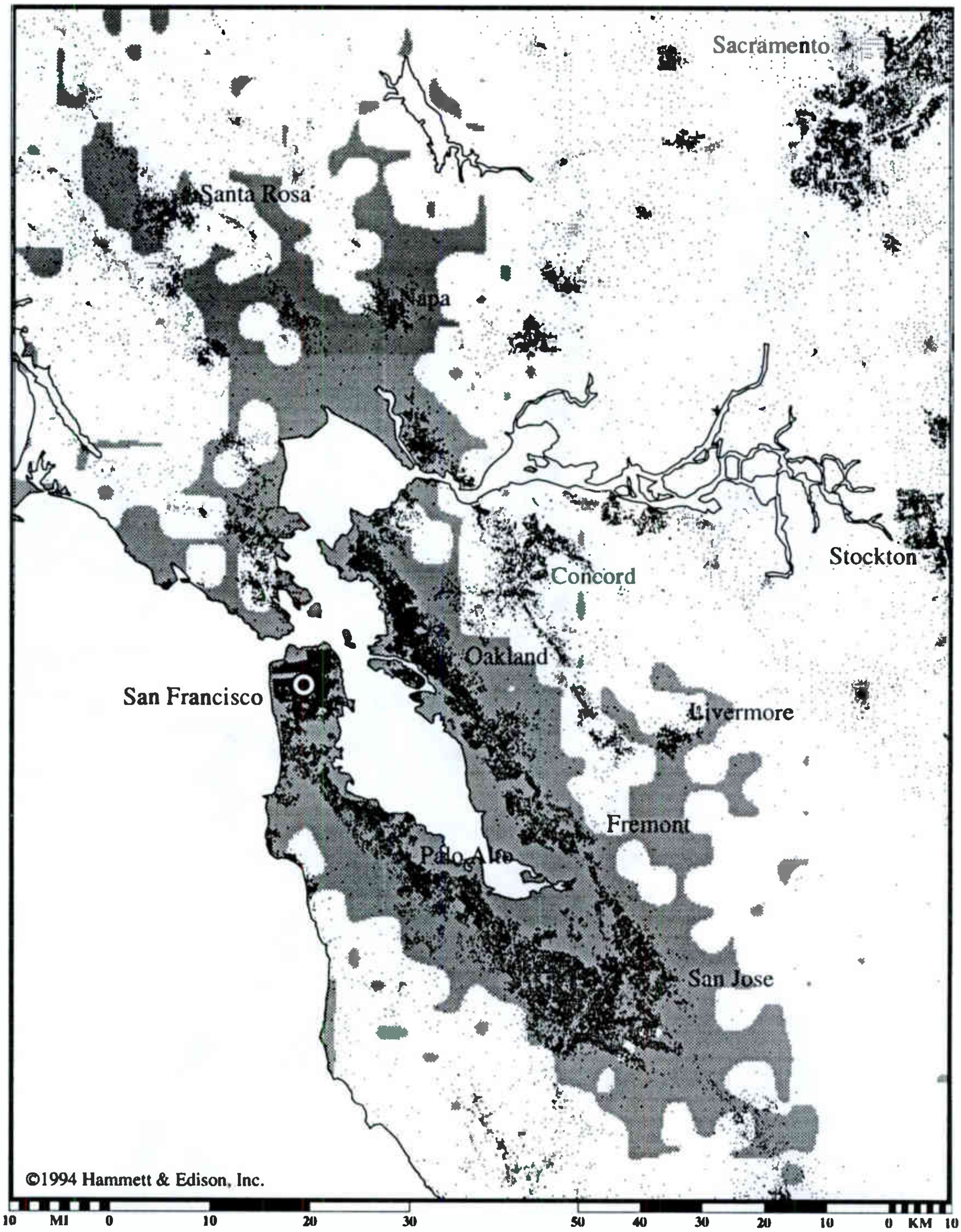


Figure 5. The potential interference- and terrain-limited coverage of an ATV station operating on one of the FCC's tentative channel allotments for a site in the San Francisco Bay area. The dark areas show predicted coverage at or exceeding 57 dBu; the white areas show areas of no coverage. The dots represent Census Blocks. The population covered by this channel is 4,661,516 persons.

Figure 6

Potential ATV Coverage on Channel "B"

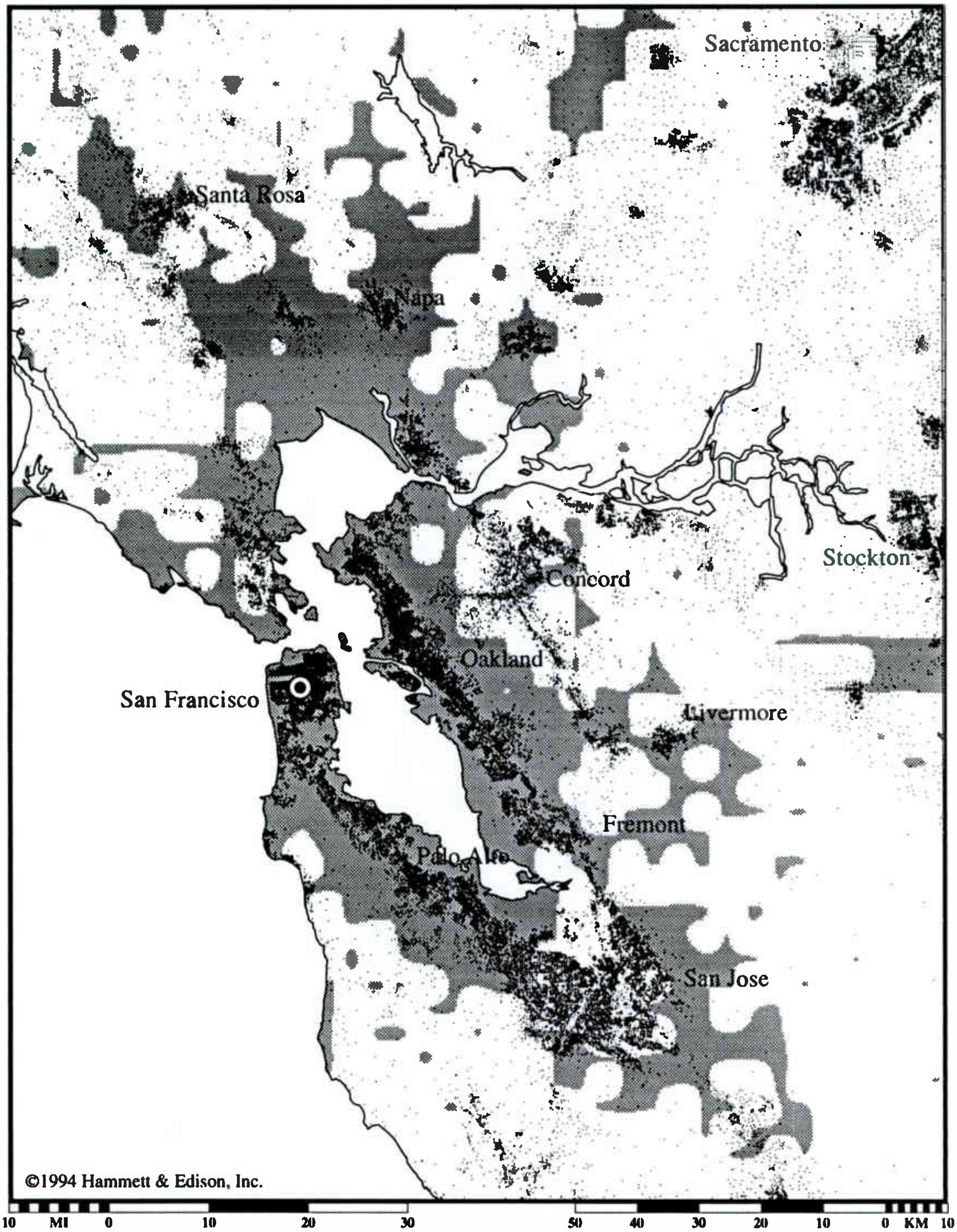


Figure 6. The potential interference- and terrain-limited coverage of an ATV station operating on another of the FCC's tentative channel allotments for a site in the San Francisco Bay area. The dark areas show predicted coverage at or exceeding 57 dBu; the white areas show areas of no coverage. The dots represent Census Blocks. The population covered by this channel is 3,896,576 persons. Note the large "white area" in southern San Jose.

GRAPHICS AND NON-LINEAR EDITING

Sunday, March 20, 1994

Moderator:
Charles Dages, CBS Television Network, New York, NY

***GRAPHICS FOR THE THIRD MILLENNIUM**
David Elliott and Chris Sanis
ABC
New York, NY

***NON-LINEAR NEWS: READY FOR PRIME TIME?**
Donald DeCesare and Frank Governale
CBS Inc.
New York, NY

NEW APPROACHES TO MULTIMEDIA
Michael Rubin
Aurora Systems
Santa Clara, CA

***THE EVOLUTION OF THE POST PRODUCTION FACILITY**
Tom Ohanian
Avid Technology, Inc.
Tewksbury, MA

AN ON-LINE TRUE RANDOM ACCESS EDIT SYSTEM: BACK TO BASICS
Bob Pank
Quantel, Ltd.
Newbury, Berkshire

***NON-LINEAR EDITING USING NON-PROPRIETARY COMPUTER HARDWARE**
Bruce Rady
TouchVision Systems, Inc.
Chicago, IL

TBD
Pierre Rinfret
SOFTIMAGE
Montreal, Quebec

*Papers not available at the time of publication

NEW APPROACHES TO MULTIMEDIA

Michael Rubin
Aurora Systems
Santa Clara, California

Abstract

This paper suggests a new way to approach system functionality and charges manufacturers to innovate more, and duplicate less. Too much computer product design has been done by engineering, with inadequate regard for users, or by industrial designers, with inadequate regard for usability. The interfaces and functionality of the new breed of "multimedia" equipment must be radically re-thought, especially if anyone intends this equipment to be for a broader base of customers than the traditional industry.

BACKGROUND

Since this is supposed to be a technical paper, it will include the requisite graphs and equations, but be prepared, they are saved for the end, and will not excite many readers the way a good Fourier Transform might. But this isn't officially a technical paper; I am an editor. I am a computer user. I am an interface guy.

This paper is about inventions, but not the kind everyone here usually talks about; I am interested in systems that involve *heuristics.*

HEURISTIC: *involving or serving as an aid to learning, discovery, or problem-solving. Self-education techniques (as the evaluation of feedback) to improve performace.*[1]

As electronics replace mechanical devices, the ability of the user to really understand the way the device works becomes obscured. Back when gears turned and you could open something up to see what was wrong, a person with some understanding of how things interact might have been able to figure a new device out. Machines tended to have a form that was somewhat determined by what the thing did and how it did it.

But now, computers give no hint as to their design philosophy, and manufacturers often have no idea how to make a computer obvious or easy to use. The less these devices overtly indicate their internal metaphor (and as the new inventions drift farther from their original film or video metaphors), the harder (not easier) new equipment becomes.

I am afraid that way too much energy has gone into the amazing technologies of our time: HDTV, compression, machines, tools... and not enough attention has been paid to the way these things are supposed to work and the way people using them may want to work.

I tend to think the prevailing attitude is this: ***it's complex because it's professional.*** *Video people are highly technically savvy. Video people like keyboards and numbers.* Or worse: *Now we have the data on a high resolution monitor, we have little pictures instead of numbers, we have cute icons instead of ugly buttons, we have stacked windows and a mouse instead of boring static displays and consoles... therefore* ***we have made things better.***

What's that? These are old news? Okay. There are more insidious concepts like these, but it may make some of you flinch. Like this (excerpted from a highly respected industry analysis): "the tools ought to be as easy to use as an Apple Macintosh. . ."[2] or in a discussion of why digital tools are better than analog ones: "An object oriented graphical user interface (GUI) brings the same ease of use characteristics to video editing professionals as the Apple Macintosh GUI brings to the average consumer. Specifically, a picture representation of source footage is far more intuitive than a written list; similarly, an EDL presented as a visual timeline is easier to use than a written EDL."[3]

Most companies I think would agree with this, and most new product would give evidence in support of that contention. I would suggest that our profound love for the GUI of the Mac and other platforms is a reactionary response to years of un-graphical systems. While I myself am a Mac zealot, using my system for everything — from publishing and graphics to accounting and other things, I am not so in love to be blind to the profound failings in this kind of interface, and see what these failings might mean to the ultimate simplicity of currently-complex software. And it regard to the earlier "industry" quotation: I would suggest that a picture representation of source footage, and a visual timeline are only easier to use than written lists *in some situations*, but certainly not all.

Although I am usually reluctant to admit it, I represent a market for which most manufacturers are woefully unprepared. Just under a decade ago, I began in the world of nonlinear editing. I learned to edit from masters. I learned to use the new equipments being born in places like Boston and San Francisco. And before long, I was a professional editor: cutting TV, commercials, feature films... you know... *editing*.

But unlike those before me, I am a professional editor who has *never* edited using videotape and *never* edited using film. I come to the table with no expectations about how things are **supposed** to work. And today, this is more and more becoming the norm. Kids growing up with Macs and Nintendo looking for "professional" tools — you know the story.

I remember the big catch phrase for the early nonlinear systems was "film-style" which had less to do with being like film and more to do with being unlike traditional videotape editing. The "film style" idea simply translated into "user friendly," but marketing departments did not want to sound like computer companies and they did want to attract the technophobic.

But the technophobic are a dying breed, for the most part. And the historical functions and divisions of labor were more based on technical limitations on video products than they are for some inherent logic in the process or in human capability.

In fact, the ultimate definition of "multimedia" may mostly reflect the concepts of single person authorship and control of the visual and acoustic elements of a presentation. That is, for years we have had tools to manipulate video, audio, and graphics — the only novelty in multimedia is that a single person can do it in a single tool. And there is no traditional metaphor for the person or the tool for doing that.

What we see in the current array of amazing products (or "inventions" since this is a technical paper) is profound attention to features, and on the other hand, pretty archaic notions of human workings.

Think, for a moment, about the divisions between online and offline. Or between editing and paint. Or painting and compositing. Or compositing and animating. Or animating and editing. What I see are inventions caught up in perpetuating the architecture and dogma of the world famous, profoundly influential, videotape facility. With the priesthood of users speaking their mysterious languages; translating the artistic desires of clients into deliverable products. They guard their tools and their trade secrets. They dispense with their wisdom sparingly,

and for a price.
The image I get is like ancient astrologers, or navigators, or doctors... their ranks maintained small; dusty thick books with metal locks that creak like castle doors when slung open to a chosen page. Maps to the "new world" were national treasures and profound secrets back in the 1600s. And few of the world could read the Latin, into which all *important* writings were translated. Books were expensive. Education rare. And those navigators and doctors were godlike. Sound familiar?

The printing press allowed local dialects to be brought together; it allowed people an opportunity to learn to read, as book prices dropped, and a wider array of "software" was available. Latin manuscripts were translated into contemporary languages, and everybody had a map.

Owners of "information" no longer needed to target their books to the few, they were no longer required to be cryptic. They recognized what the big markets looked like: and they didn't look anything like the small ones.

Okay, again this is old news. The Macintosh has allowed the proliferation of video tools — editing, graphics, text — which should end the story. Right? Perhaps, but those "low-end" software companies have perhaps unwittingly made their tools in the **image of the ones they've replaced.** And what they have in effect created does little to advance the way the tools are used other than cheapen them to the point that the old tools are available to a wider market, if you can figure out how to work them. And the cheaper tools are still too slow or too low quality in many cases to supplant their progenitors at the high-end: no matter how you slice it, Photoshop is not Flame.

So what do I suggest?

I would suggest that while you are inventing new tools and bundling your tools with other tools (creating a panorama of redundancy, conflicting interface paradigms, and artificial barriers), try inventing something that improves the way people work. Not the people who are currently employed (they have much to play with), but start looking to the folks who come to the creative process with little emotional baggage.

I, for one, don't want to be told an editing system cannot paint. I don't want to have explained to me why compositing should be so confusing, or why so many buttons are on the screen or why that box won't take a half-finished thing from this box. You built the walls, and now you have to remove them. It will be an unpopular idea, from the standpoint of the *status quo* — it always is. You'd be in good company.

I suggest we re-investigate the creative process. Look to children. Look to the "primitives" of creation — for any media — but also for moving pictures. There are painfully simple basics and they are the same for all these equipments: basics like stacking things up, juxtaposing and re-arranging things, linking things together in different ways, coloring pixels, lining things up in organized, and timed ways.

I have been working for the past year on defining the fundamentals of the creative process, and developing tools that teach as you use, that are self-explanatory, that offer little barriers and no dead-end paths. I will not outline the array of these things, because, unfortunately, I am somewhat prohibited from laying them out to the public. However take, for instance, some of the more well known work done in editing.

What follows illustrates two of the principle concepts that depart from common understanding in post production. The first concerns the basic definitions of nonlinearity, but impacts our understanding of online and offline. The second concerns a simple example of my problems with most GUIs.

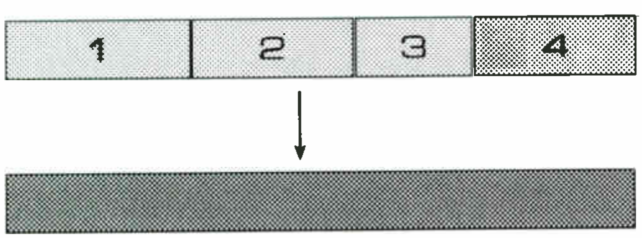
ILLUSTRATION NUMBER ONE

The term "nonlinearity" is vaguely defined as a **pre-**

view, but is colloquially understood as an *extended* preview. The term has evolved from references to edited sequences that could be previewed as if they were assembled to videotape, but were actually not assembled, thus making changes easy and following previews quick.
I have further defined the term to include the concepts of **horizontal** nonlinearity, and **vertical** nonlinearity[4].

Horizontal, or temporal, nonlinearity involves the maintaining the discreteness of shots (or elements) in the temporal dimension. As long as these elements are maintained as discrete and separable pieces, they can be modified and previewed in many digital (and some analog) nonlinear systems. When these elements are recorded to a master videotape, they loose their individuality, and thus they loose their nonlinearity. Loss of nonlinearity in the horizontal dimension is traditionally known as an **assembly**.

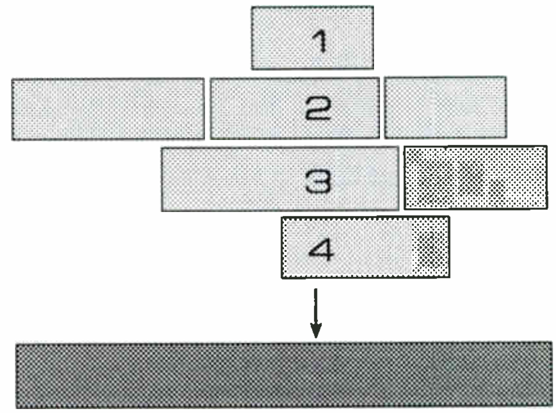
Horizontal Separation and Assembly[5]



Vertical, or spacial, nonlinearity involves the maintaining the discreteness of shots (or elements) in the spacial dimension. As long as these elements are maintained in discrete and separable *layers*, they can be modified and previewed in many graphical, acoustic, or image compositing systems. When these elements are combined and recorded to a master tape or disk, they also loose their individuality, and thus they loose their nonlinearity. Loss of separation in the vertical dimension is traditionally known as a **mix-down.**

The concept is designed to highlight the sameness

Vertical Separation & Mix-down[5]



between what "nonlinear" editing systems do, for pictures, and what audio systems, or video compositing systems do with layers of material. It is a common experience for a video effects editor to create multiple layers for a single effect, composite them, and then wish to change one element right in the middle — which necessitates the fixing and recompositing of the entire effect. This is totally equivalent to the linear videotape editor wishing to make a change in the twentieth shot...

Thus, if effects can be in a constant preview, while maintaining the separation of the layers in the effect, the result is a vertically nonlinear system. By far more common are vertically *linear* layers, that must be mixed together to view. Audio and video both can be defined in terms of vertical layers which are mixed together for reasons of cost and performance; the very same issues that surround horizontal nonlinear systems — "how long do you stay nonlinear, and when do you go to the tape?" Energy must be expended to maintain the separation, either vertically or horizontally, of nonlinear elements. Energy that manifests itself often in terms of money.

While systems are rarely pure horizontal or pure vertical, the computational and storage resources of the system always are balanced for some kind of optimization based on the required task.

One object of this kind of model is simply a reduction in required tools and models for manipulating pictures and sound. In short, tools required to manipulate elements vertically are often the same, whether managing images or sounds; and tools required to work horizontally are often similar to those for working vertically — moving, re-arranging, inserting, rippling, lifting, trimming, etc. Working with a different model, completely different kinds of equipment could be designed that do not emulate current divisions of functionality, but truly integrate these kinds of tasks into a single interface and toolset.

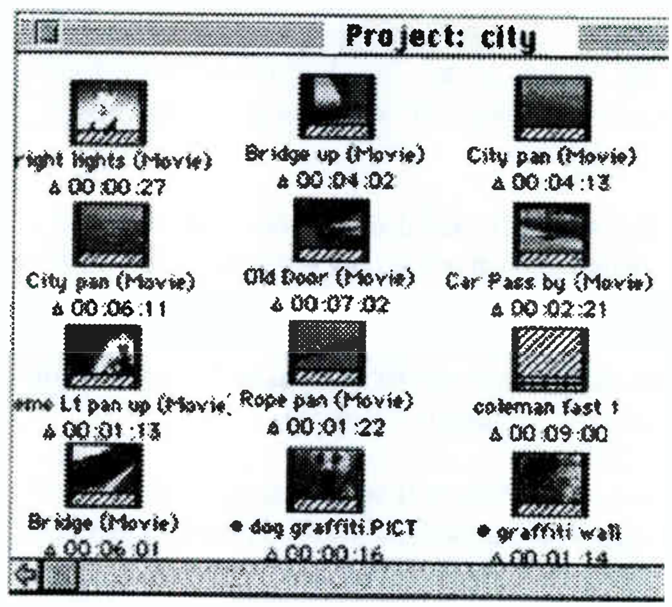
ILLUSTRATION NUMBER TWO

While there is no question that graphical user interfaces (GUIs) have made editing systems more accessible; and the 20-year-old model of timecode editing was sorely in need of modernization, it is extreme and uninspired to translate the *entirety* of functionality to the icons of the GUI.

The role of the icon and iconic representations has its place in the modern computer. Particularly in the hierarchical file structure of computers: the file and folder icons make computer organization understandable even to the technophobe.

Originally, computer users manipulated raw data — and displays appropriate for computing reflected this data. Data — whether coordinates in space, or videotape timecodes — could later be visualized through some (albeit limited) process, to see if the manipulations created the desired effect. The closer the visualization process was moved in time to the data manipulation process, the better for the user. Graphical computer displays originally allowed for a representation of the data to be shown; later, more advanced graphical displays allowed a representation of the object to be manipulated which itself changed the source data. These types of systems can be illustrated in many disciplines, from architecture to editing. But better still is a manipulation of the final object itself, and not a graphical representation of that object. While this may not always be practical in architecture, it has applications for graphics and editing where layer or shot representations can be adjusted to modify databases. Manipulation of the images themselves are often obstructed because systems are designed to manage icons and graphical objects moreso than video. While there is no doubt a place for icon manipulation, it might be suggested that these functions are secondary and not primary.

For illustration here, I will choose a single case to show an obstruction iconic thinking places in "visual computing." The source database in virtually all nonlinear systems (horizontal, mostly, but even vertical and linear systems, but to a lesser degree) has become a series of tiny digitized images. Called many things, from 'clips' to 'picons,' these files generally are represented by a single still frame from the series of frames contained within. This method is consistant with the standard icon protocols of windowed computers displays.



Example of icon-based video files
(courtesy of Adobe Systems,© 1993)

All windowed file representations create icons for files; each file application has a unique icon, and below the icon is the file name, as created automatically or by the user. When more than one version exists of a document or graphic, each is treated by the computer system as a separate file, and each is given a separate icon representing it in the window.

I propose an icon system can exist where files created as versions of one another are represented by a *single* icon, with side-tabs (or similar sub-classification) available that distinguish between the versions.

The benefits of this are three-fold. First, from a screen real-estate economy standpoint, the creation of multiple versions of a document or graphic file does not result in multiple (and numerous) icons — each identical to the other except for the name difference. Screen clutter would be managed and made more efficient. Second, from a project management standpoint, the proposed icon system greatly enhances and streamlines the creative process — offering a path of document versions, different only due to the work done on them — and allowing easy process management. Third, in the folder-file hierarchy common in most personal and professional computer operating systems, the proposed icon system facilitates easy location of, and comparisons between, different versions of a document or graphical file.

This system is based on the idea that two separate documents that are simply versions of a work may be "more related" than any two average documents. By strengthening the bond between "versions" of a work, iconic computer systems and representations can be significantly enhanced.

Software products (like editing systems) are dedicated to the organization of graphical (and visual) documents. Each document is depicted by a "representative snap shot" frame from that document (i.e. graphics are represented by a low-resolution 'thumb nail' of the graphic itself; video and animation sequences are represented by a single chosen frame from the sequence). This type of organization of files is highly visual, but disregards the "relatedness" of the files. The proposed iconic system uses a single 'thumb nail' image, chosen by the user, to represent a *class* of files (as it were).

In the film and video industry, these files are parallel to "scenes and takes," where a scene is defined as a single camera set-up in a given location with a given set of characters, action, and dialog. Each version of this scene (i.e. each repeated performance that is recorded on film or videotape or audiotape) is considered a "take." Thus a given scene may have many takes.

In the traditional computer iconic file system, each take (of any scene) is a unique file, which users then may organize manually into folders to partition the scene; in the proposed iconic file system, a scene is equivalent to a file, and a take is simply a version of that file.

CONCLUSION

If steps are not taken to invent new products that address true multimedia authoring — to simplify the tools rather than blindly accepting the status quo as dictated from the traditional video industry — lesser hybrids will evolve from editing systems or graphics systems and obscure the path for the interested public. The computer industry may not now possess the know-how, nor the video industry the desire, to simplify the required tools.

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AN ON-LINE TRUE RANDOM ACCESS EDIT SYSTEM: BACK TO BASICS

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ABSTRACT

Complementing their use in off-line and special effects editing, disks are now being applied to the mainstream of on-line. Taking full advantage of the medium and using today's technology, an integrated system dedicated to editing has been produced, bringing new suite design and operation. Besides handling current edit requirements future needs and possibilities are taken into account.

INTRODUCTION

The dictionary defines editing as 'making up the final version by selection, rearrangement etc. of recorded material'. History shows that as technologies have advanced so the ways of editing have changed. For moving images, the methods used for film have been quite different from those used for video tape, even though both are aimed at conveying information to our eyes and ears. The difference is in the recording medium and it is this that has always dictated the method of editing.

Although both film and tape continue as major forms of storage, increasingly disks are being applied to TV production. Their use in off-line and special effects editing has already proved popular and served to highlight the limitations of editing on linear tape. Now turning the application of disks to the mainstream of on-line editing has involved both new technologies and new ideas - taking a fresh look at the whole business of editing, from the basics upward, rather than simply mimicking old established practices. At the same time any new system designed for today's production needs must be able to grow to offer more for the future.

The history of video editing is one of continued growth in capabilities: from the early days of 'cut only' to adding more VTRs, timecode, edit controllers and a switcher for mixes, wipes, keying and audio, to today where the suite may include caption generation, retouching and digital effects. These machines, along with their control panels, monitors, interfaces and signal routing, represent high capital investment, expensive installation and costly running. Moreover operation is still restricted by the linear nature of tape.

SYSTEM DESIGN

One of the big bonuses of digital technology is that it brings the opportunity to combine functions into one unit, in this case to make an integrated system dedicated to editing. This involves much more than just the introduction of a new form of storage - disks; to make the most of the opportunities, an entirely new design of suite is required. Looking afresh at the whole business of editing new methods of operation can be introduced while the hardware is greatly rationalised. This is the background to the design of Edit Box.

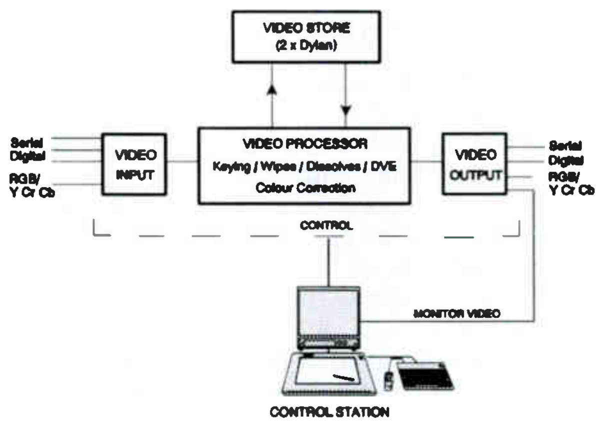

Fig. 1 System concept

Following the structure of other suites, audio and video are processed separately but coordinated within the control system. The requirements for video can be bundled into four sections, the processor, store, control system and input / output interfaces (Fig. 1). For on-line editing of production material it is important to maintain fidelity at all times, dictating that the full CCIR 601 component digital signal is used throughout. Allowance must also be made for a key signal, giving a 4:2:2:4 structure to the system.

VIDEO STORE

The object is to offer a practical length of storage for general editing operations. The use of digital disk recorders (DDRs) gives instant access to video clips with the absence of VTRs' mechanical considerations - spooling and pre-rolls. These are great advantages but stand alone DDRs are mainly built as 'video caches' to complement conventional edit suites. They have a minute or so of storage and give instant access to clips but are essentially linear in operation. One would be required for every video source and there would also be the need to manage the disks to ensure continuous space is available for recordings.

The store is required to supply all video sources and destinations as well as true random access - the ability to read any frame in any order at, or above, video rate - 21 M bytes/s - or more. (Note: This includes non-linear operation). With no such disk system available a new one had to be designed. The result is Dylan, an array of 20, 3½ inch SCSI drives. Two Dylans are used in Edit Box each with its special interface electronics to combine the disks to supply the required speed and agility as well as error checking and correction.

The true random access is very important to the edit operation. A cut edit becomes just an instruction to read frames in a new order. With twice video rate from the store, wipe and dissolve transitions, requiring simultaneous access to two sources, can be executed in real time. Unlike tape operations these do not require any re-recording and the results are immediately available for replay at full resolution. As the edits then only exist as the original clips and edit instructions, nothing is committed, allowing any adjustments to be easily made. More complex tasks, such as multi-channel DVE moves, that may require accessing more video sources, are accommodated. In such cases the video frames are called from the store, processed and the results recorded back as new frames on the disks. On replay these are automatically inserted into the completed sequence.

VIDEO PROCESSOR

The structure of traditional VTR edit suites is based on the real-time processing of all signals, its linear medium being best able to play and record at 30 frames per second. This means the complexity of the work undertaken has to be reflected directly in the processing hardware. To avoid high costs the escape route often chosen for more involved requirements is to use several passes - a limited option unless digital VTRs are used. In contrast, the random access nature of the Dylan disk store is well able to supply any frames at rates up to 30 fps - and beyond, allowing time, where required, to access many sources, and for the video processor to complete its whole set of functions in one pass. Such operations require very close coordination between store and processor, far beyond that found between VTR and switcher. For this reason both the store and the processor have to be integral parts of the system. As a result great economies can be made in hardware: most obviously there is no proliferation of VTRs.

To take full advantage of the store's speed and maintain image quality dictates that most video processing is executed in dedicated hardware, only the less demanding tasks being assigned to pure software solutions. As already mentioned the basic edit operations of dissolves and wipes can be processed in real-time, cuts being created totally by store addressing. The processor handles other usual requirements such as DVE and keying (chroma and luminance) while further special functions such as colour correction, texture and video / matte retouch can be added.

The random access to all pictures enables some other processes to be better undertaken. For example time stretching, available on some VTRs as a frame repeat or drop routine, can become fully processed with the frames interpolated to produce smooth movement rendition over greater speed variation. Also frame average and integrate techniques can be used to produce a range of special effects.

INPUT AND OUTPUT

Since all processing takes place within the system the input and output requirement is kept to a single channel only. Un-edited video and audio is played in, normally from a VTR, and the completed results played out - possibly to the same VTR. Serial component digital video is required for connection to the new generation of VTRs, and analogue component for direct operation with existing analogue equipment. Using the digital route the only interface processing required is between serial connection and the internal parallel operation of the system, making the entire input / output operation 100% accurate.

Audio uses the AES/EBU 48 KHz digital standard with one stereo input and two (duplicated) stereo outputs.

CONTROL

With the new system structure, the established methods of edit suite control are no longer appropriate. Being an integrated system there is the opportunity to do without a large array of separate panels and screens and to place all control into one controller. There is a huge array of functions to be handled including much system configuration and switching. It is essential that the technology is hidden, giving only clear, concise, understandable functions to the operator.

The method chosen uses a pen and tablet working with on-screen menus and graphic displays appearing at the video output, giving access to all functions of the suite. The control station is complemented with a keyboard and 'rat' hand controller but the main operation is with the pen. The menus are defined in software so, looking to the future, control of any new facilities can be easily implemented.

Operation depends on the close coordination of all the system sections. For example the keying of a DVE image over a background involves the fetching of the first background frame into the keyer, with the first foreground frame and its frame of key directed to the DVE, its parameters set and the result sent to the keyer. The keyer output is recorded back as a new frame on the disk store. Then the next frames of background, foreground, key and DVE setting are fetched... and so on. During set up, the visibility of the process must always be clear with provision for preview and adjustment. The control system brings together all operations so that, as the background is shuttled, so the DVE, pictures and key are kept in step. The result can be checked and altered as required.

VTR control is included to handle the loading and replay of material. Although the minimum requirement is for only one VTR, to save re-plugging, set ups and external switching, provision is made for four.

OPERATION

Traditional editing has depended on time code for the identification of frames. Time code was developed for video tape when it was only possible to play pictures at full video rate. Since then VTRs have given us slow motion and freeze frame but access has still required spooling tape. The use of time code may not have led to the best possible way of editing but it has been the only practical method for tape. To give a more direct way of working, akin to the much favoured aspects of film editing, quite different methods have been adopted in the true random access suite. With all pictures instantly available the pictures themselves can be used as reference points. This makes possible editing by pictures rather than editing by time code. For example, the frames of video clips can be displayed, like film clips, as vertical strips (Fig. 2). Then a cut can be made by flicking the pen across the junction of two frames. The cut clip can be moved and joined to another by tapping down on any required 'in' or 'out' frame. Tapping a menu box can change the transition to a dissolve or wipe (defined by SMPTE number) and the result is immediately available for viewing at full resolution. For reference back to VTR material, the original time codes can be revealed.

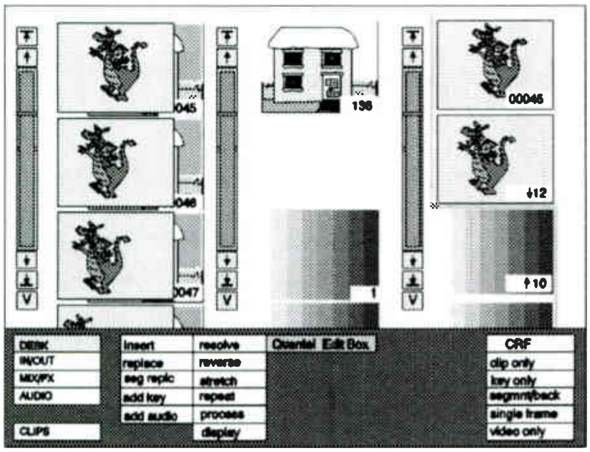

Fig.2 Edit desk display

As could be expected operation does not follow the routines established for linear editing but rather takes advantage of the flexibility offered by the new system. It starts with the loading-in, from a VTR, of the original footage to be edited. If a CMX 3400 EDL has been produced in off-line, the required material is automatically recorded into the store. To allow room for later adjustment, edit tails - extra frames before and after each selected clip - are included in the recording. The edit can then be conformed, a process taking only a moment, and the result can be viewed at full resolution. Although conformed, none of the edits are committed. Using the extra frames of the tails, edits can be changed moving the 'in' points, 'out' points or both: cuts, wipes or dissolves are automatically re-made or may be changed. Equally segments can be replaced, without the restriction of the new one having to be the same length as the old. So the mechanics of basic editing can be swiftly executed while providing great flexibility for the 'rearrangement' mentioned in our dictionary definition.

The many functions not defined by the CMX protocol can be added during the editing process. So, for example, DVE moves can be added or colour corrections applied.

AUDIO

To provide the same random access as the video, the medium used for audio is also disk. The system provides for two main stereo tracks and a third stereo backing track. Default operation makes audio-follow-video edits and, as with the video, tails are retained to allow later adjustment. The chequer-boarding of the audio is shown in an audio menu to allow adjustments such as slipping the edit and altering levels. Since audio must be cut to a far finer resolution than that defined by the fields of video, zooming into the menu allows accuracy down to 5 milliseconds.

OFF-LINE

Historically the phrase non-linear (true random access includes non-linear working) has become synonymous with off-line, a fact that can lead to some confusion. Off-lining often involves handling uncut video shoots or film rushes, to make some initial selections, as well as decisions for the final edit. In such cases the need for hours of digitising time and storage is clear. Also the low cost of off-line allows time to be spent in sifting through large quantities of material. Operation in any on-line suite differs; be it linear or non-linear, the costs are higher. In every case it is wise to arrive prepared with at least some selection made as to the required footage. If so the need for digitising time and storage capacity in the on-line are reduced and its time better employed.

The widely used CMX 3400 EDLs carry much of the basic edit decision information but stop short of issues such as keying and DVEs. Reasons for this include the wide variation in capabilities of such devices and their precise implementation in the edit suite. That situation has been changed with the introduction of the integrated edit suite. The full range of its facilities are defined making possible a comprehensive preparation system. This has led to the development of a compressed video version of the on-line, Micro Edit. Having exactly the same operation, its edit History Archive data can be stored on a magneto optical disk, transferred to the on-line system and used to reconstruct the whole edit from the rushes. If required, further adjustments can then be made.

SPLIT SESSIONS AND RE-WORK

As the video storage is fixed and used as work space for the job in hand, not long term storage, it is important to be able to save a session at any point, for it to be re-loaded and continued later. For this, History Archive files can be made in the on-line system, just as they are for the off-line. All instructions for the edit are automatically stored in conjunction with the video and audio. An MO disk and video tape are used as the removable media which allow the session to be later re-loaded into any similar system. Thus split sessions can be arranged and re-working of completed edits accommodated.

RELIABILITY AND MAINTENANCE

VTR suites comprise a great list of equipment. Of this it is the elecro-mechanical machines, the VTRs themselves, that normally require the most regular service and maintenance, adding significantly to running costs. In the true random access suite only one VTR is needed and, as this is only required for playing in and recording finished results, it has relatively little use. VTR running costs are greatly reduced.

The decrease in VTR activity is due to the use of the suite's own integrated video store. Two major attractions of using disks are that they require no routine maintenance and those available today offer a mean time between failure (MTBF) of over 200,000 hours - more than 22 years. Using multiple disks reduces this statistic but the store remains highly reliable. To ensure uninterrupted operation and security of data there is built-in redundancy, error detection and correction in the Dylan arrays so that, in the rare event of a drive failure, the system continues full operation. When convenient the offending drive is replaced, the missing data re-generated and recorded back.

Building an integrated system has greatly cut the number of items in the suite. Connectors, cables, power supplies and interfaces are all reduced and the use of VLSI and low power components all contribute to a very high standard of reliability.

THE FUTURE

All the functionality of a traditional mid-range, on-line, VTR suite has been provided plus full component digital operation and uncommitted, random access editing. The hardware, including the video store, two effects channels with DVEs and keyers, occupies only 28 inches of rack space, and power consumption is less than 1 KVA. The only support required is a single VTR along with audio and video monitoring (Fig. 3). Control is integrated, disposing of the usual proliferation of panels and screen.

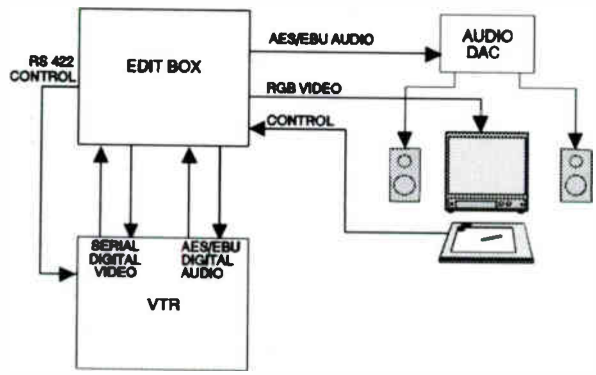

Fig. 3 The complete suite

It is widely believed that the future of editing lies in disk based systems but the real benefits can only be realised if they are designed, from the ground up, around disk use. With disk technology making long video stores economically viable, there is a strong case for avoiding any compromise in the quality sensitive area of mainstream on-line suites. For the long term future a growth path is essential and starting with a full resolution CCIR 601 system provides a sound technical base.

The combination of a true random access video store operating in an integrated, dedicated system has directly addressed what editing is about 'making up the final version by selection, rearrangement etc. of the recorded material'. That is surely a foundation for the future of editing.

DIGITAL AUDIO PROCESSING AND SYSTEMS

Monday, March 21, 1994

Moderator:
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MAINTAINING A 100% DIGITAL PATH FROM THE STUDIO TO THE "ON AIR" RF SIGNAL
Geoffrey Mendenhall
Harris Allied Broadcast Division
Quincy, IL

***STATE OF THE STANDARDS: CODING AND CONNECTIVITY**
Gregory Urbiel
CBS Radio
Southfield, MI

***UTILIZING SECOND GENERATION TRANSMISSION PROCESSORS FOR AUDIO**
Robert Orban
AKG Acoustics, Inc
San Leandro, CA

THE INTERACTION OF AUDIO PROCESSING AND LOW BIT-RATE CODING IN BROADCAST APPLICATIONS
Eric Benjamin
Dolby Laboratories, Inc.
San Francisco, CA

*Papers not available at the time of publication

MAINTAINING A 100% DIGITAL PATH FROM THE STUDIO TO THE "ON AIR" RF SIGNAL

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SYNOPSIS

With the advent of digital STL links, STL modems, and T1 data interface devices, it is now possible to convey digital audio from the studio to the transmitter input digitally, but there is still a final D/A conversion to composite baseband before the input to the analog transmitter. This paper discusses the elimination of all A/D and D/A conversions between the studio source equipment and the RF modulator by providing a digital modulator that can accept direct digital inputs via the AES/EBU digital audio serial data standard. Issues regarding the need for and location of digital audio processing, the DSP stereo encoding process, and peak level control are addressed. Block diagrams showing the location of digital audio processing devices and STL components are included.

INTRODUCTION

For the past decade, the modulation performance quality of the FM exciter and transmitter system often exceeded the quality of the STL, making the STL the limiting factor in the overall transmission system. Recently, fully Digital STL's and Digital STL modems (used to upgrade analog STL's) have been introduced. Analog phone lines are being replaced by T1 digital phone lines capable of carrying high data rates. In addition, fully digital audio processing equipment is now available and in use. This new generation of audio processing equipment accepts a digital audio input, processes this data fully in the digital domain, and outputs this processed data without any analog to digital (A/D) or digital to analog (D/A) conversions.

With these recent innovations, the conventional analog FM exciter has now become the limiting factor in an otherwise all digital system. A fully digital FM exciter that can accept serial digital audio data directly, digitally process this data into the digital equivalent of stereo baseband, and then digitally modulate this information directly to an analog radio frequency carrier now completes the all digital path from the audio source to the transmitter RF power amplifiers. The digital FM exciter is fully transparent to it's digital input, eliminating the additional A/D and D/A conversions of audio data that are required with conventional analog audio processing, analog stereo encoding, and analog FM exciters. Before proceeding into the description of the data path from the source to the transmitter, a review of the terminology that is used will be helpful.

THE AES/EBU DIGITAL AUDIO INTERFACE SERIAL DATA STANDARD

Some years ago, Harris/Allied initiated the formation of a working group of radio and studio broadcast equipment manufacturers to define data format standards for both digital audio and for digitized composite baseband. The results of this working group were the selection of the AES/EBU (Audio Engineering Society/European Broadcast Union) serial data standard defined in the AES5-1984, ANSI S4.28-1984, AES3-1985, ANSI S4.40-1992 and AES3-1992 documents. This is the digital audio data format to be used as the interface standard for audio sources, mixing/control equipment, audio processing equipment, STL equipment, and transmitter inputs. The European Broadcasting Union has republished a standard which is identical to the AES3 standards, except for the use of transformer-coupled inputs and outputs. There is also a "consumer-use" digital interface standard, described in IEC958 and other documents as the IEC 958 Type II ("S/P DIF") Consumer Interface. Although it appears to be compatible with the professional AES/EBU digital interface in limited electrical terms, the data formats of the two standards are incompatible.

The different and changing data format requirements for interfacing digitized composite baseband directly to the digital modulator made it impractical to agree on a universal composite baseband standard. Since the "lossy" data compression schemes cannot be applied to digitized baseband, the need for a composite baseband data format standard has largely been eliminated.

The format of the AES/EBU serial data subframe defined by AES3-1992, is shown in ***Figure 1.***

Some highlights of the AES/EBU data format are:

- The interface format can accommodate 16, 20 or 24-bits of digital audio information.
- The interface handles serial data transmission of two channels of digitized audio over a conventional shielded, twisted-pair wire, for distances up to 300 feet.
- The interface uses standard 3 pin, XLR-type connectors, carrying balanced, RS-422 compatible signals that are polarity independent. The input and out-put impedance for the interface is 110 ohms.
- The data is sent least significant bit (LSB) first, with alternating subframes for Channel 1 and Channel 2.
- The data is self-clocking, and does not require an additional CLOCK connection to synchronize the source and destination. The SYNC data allows digital equipment to recognize the start of each 32-bit block of audio data and synchronize the master clocks.
- There are four ancillary bits for Validity, User Bit, Channel Status and Parity.
- The Validity Bit indicates whether the audio sample data bits are valid and error free.

32-BIT SUBFRAME FORMAT
OF
AES/EBU SERIAL DATA

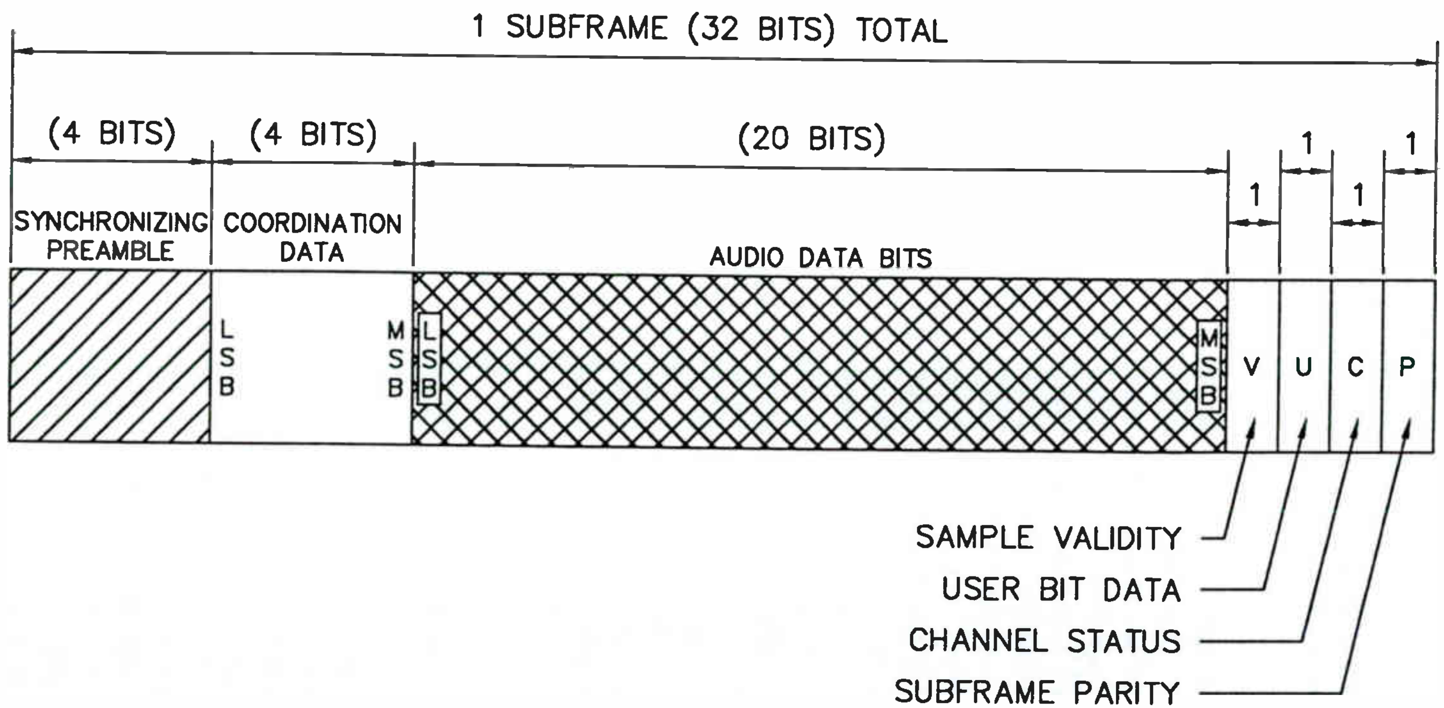

FIG. 1

- The User Bit carries hardware or system specific information.
- The Channel Status carries data concerning emphasis, sampling frequency, and various other information.
- The Parity Bit provides even parity over the current subframe, allowing simple detection of transmission errors.

The acceptance of the AES/EBU data interface standard by all the major broadcast equipment manufacturers makes is possible to build an all digital studio with an all digital link to the transmitter using standard off-the-shelf equipment. The standard AES/EBU transmit and receive chipsets support the three commonly used data rates of 48.0kHz, 44.1kHz, and 32.0kHz. Since the current FM stereo transmission standard limits the frequency response of the left and right channels to a maximum of 15kHz, a 32kHz data rate is often used. There are situations where a higher data rate may be desired, so make sure that all the equipment has the ability to accept any of these three data rates.

DIGITAL SIGNAL PROCESSING

Digital Signal Processing (DSP) can be viewed as a technology that emulates analog functions by simulating these functions in software that runs on specialized, high speed, digital microcomputers. After analog audio is digitized into a series of numbers, these numbers can be manipulated mathematically to simulate processes normally performed by analog circuits. These simulations execute so rapidly, in nearly real time, they can be used as replacements for these analog functions. Filters, mixers, modulators, and virtually any other complex analog function can be performed by DSP if sufficient computing "horsepower" is available. DSP based products offer unique benefits by providing additional functions and precision that are unattainable with analog technology.

"LOSSY" COMPRESSION

"Lossy" data compression is required to reduce the data rate by at least 4:1 in order to meet the requirement that the RF bandwidth of the STL signal fits into the existing channel allocation. There are two types of data compression schemes, (1) "Lossless" which introduce no change to the recovered data and (2) "Lossy" which cause a permanent loss of some of the recovered data. "Lossless" data compression introduces no effect on the retrieved data or audio quality, but current "lossless" compression technology used on personal computers, like *STACKER, DBLSPACE,* or *PKZIP* are not effective in reducing audio data, so virtually all data compression schemes used for reducing audio data are of the "lossy" type. "Lossy" data compression schemes take advantage of perceptual masking and other psychoacoustic effects in human hearing perception to discard some of the audio information without being very noticeable to the listener. There are several different "lossy" systems in broadcast use today that provide satisfactory results under certain conditions. The most sophisticated systems such as *MUSICAM/MPEG*

Level-II and *PAK* offer minimal audio degradation that can only be discerned by trained listeners in A/B comparisons with the original audio source.

Figure 2 is a block diagram of a typical all digital path from the source to the transmitter.

The digital audio sources like CD players and R-DAT record/play machines may be consumer grade equipment with optical or S/P-DIF data interfaces which will require data rate conversion and other special accommodations at the digital console input. Most professional grade digital audio sources will use the AES/EBU data interface standard which is compatible with the digital inputs to the mixing console. The microphone is about the only remaining analog source to the mixing console. The console mixes the data from the various sources in the digital domain and outputs the resulting sum as data in the AES/EBU format. Note that compressed data from the digital sources cannot be mixed in compressed form. It must first be expanded by the source or with a compatible data expansion function built into the input to the digital mixing console.

The AES/EBU output from the console is then distributed to other locations including the transmitter site. Two common ways to deliver the AES/EBU data are either through a digital STL radio link or through a T1 digital telephone line. Some of the newer digital STL's have the data compression and expansion and RF coding functions integrated into the radio units, while another approach uses an external pair of STL modems to compress the data into a baseband format that can be fed through an existing composite STL and then expanded at the transmitter site. Digitally encoding the STL provides the additional benefit of adding more than 20dB of additional fade margin to an existing STL path.

All of the systems used with existing 950MHz STL channels, require "lossy" data compression to meet the occupied bandwidth requirements.

The T1 digital phone line has sufficient bandwidth to convey the AES/EBU data rates without the need for data compression. This allows more freedom in the location of the digital audio processing equipment at either end of the studio to transmitter data link.

If any type of "lossy" data compression is used in the studio to transmitter path, the digital audio processing device should be located at the transmitter site for direct connection to the digital exciter. Examples of "lossy" data compression are *APTx, DOLBY AC-2, MUSICAM, MPEG,* and *PAK.* These and other "lossy" data compression schemes produce an output that is not identical to the input by design of the digital signal processing algorithms used. Among the differences introduced, are changes in the absolute peak levels produced at the output which would cause overshoots and result in overmodulation of the transmitter. By locating the digital audio processing after data compression and expansion, the audio peak levels can be tightly controlled before feeding this data to the digital exciter.

THE DIGITAL PATH FROM SOURCE TO TRANSMITTER

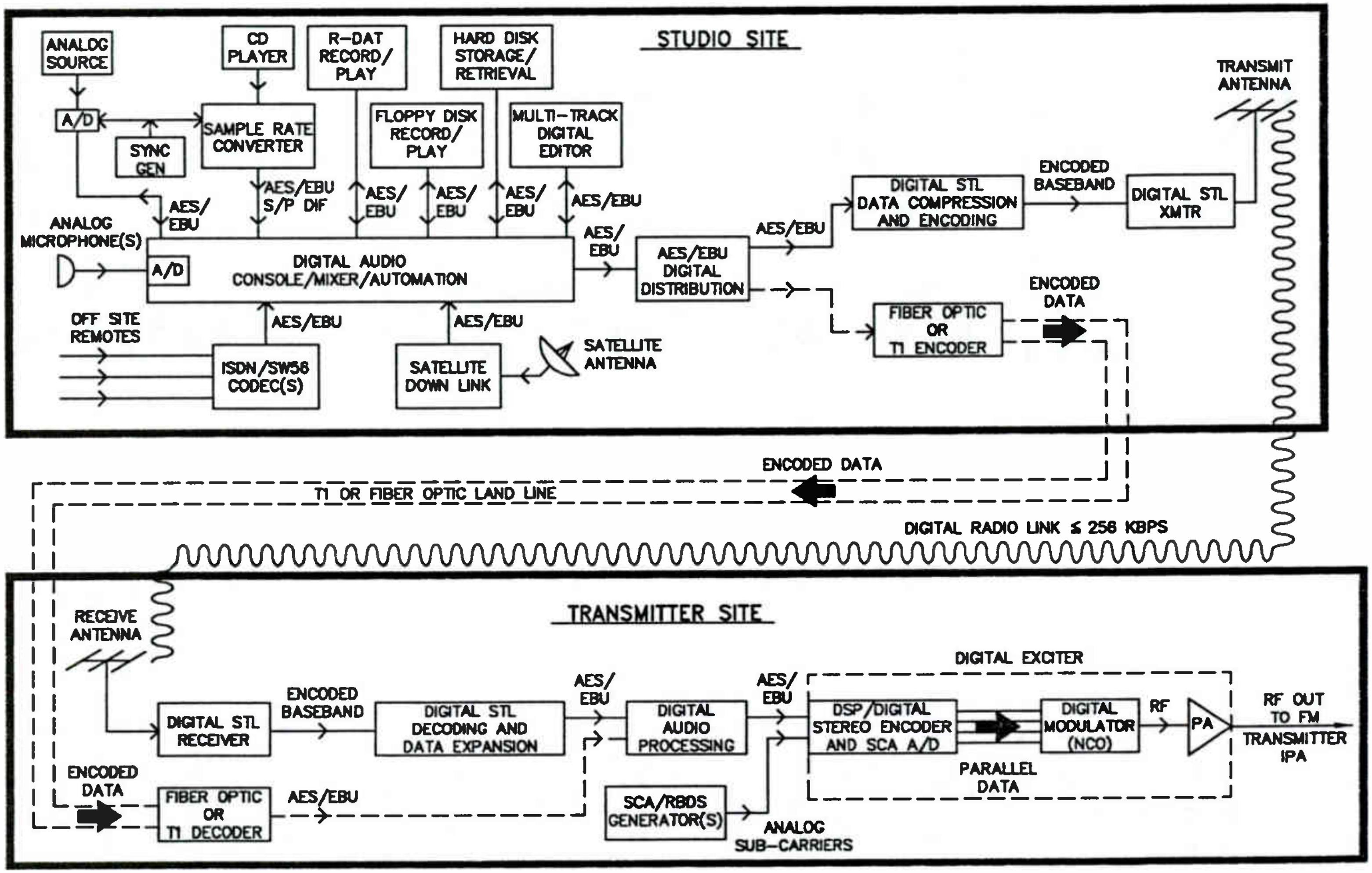

FIG. 2

The digital stereo encoder is located within the digital exciter and has an optional peak limiter that provides a function similar to analog composite clipping without causing the inter-modulation distortion to baseband that analog baseband clipping causes. This feature is not absolutely required when the *OPTIMOD* 8200 digital audio processor is located just before the digital exciter.

The STL or T1 receiver converts the data back to the AES/EBU format for delivery to the digital audio processor. After processing the data, the digital audio processor provides an AES/EBU output directly to the digital input of the digital exciter.

DIGITAL EXCITER DIGITAL INPUT TO RF OUTPUT

Figure 3 is a block diagram of the digital exciter with AES/EBU input, DSP stereo encoder, and digital modulator.

The digital exciter input provides data rate conversion and encodes the left and right audio channels contained within the AES/EBU serial data format into parallel composite data that represents the equivalent of stereo baseband. This parallel data is fed into the phase accumulator of the digital FM modulator. Direct Digital Synthesis (DDS) converts the output of the phase accumulator into a synthesized RF waveform which is upconverted to the carrier frequency and then amplified to the level required to drive the transmitter power ampifiers. No tuned circuits, varactor diodes, or other analog devices are used in the modulation process.

SCA / RBDS COMPATIBILITY

The digital exciter's digital input provides (3) analog subcarrier inputs which are compatible in level and impedance with standard SCA or RBDS generators. Although these subcarrier functions could be generated in the digital domain, there are so many different variations of subcarrier encoding methods used by different services, that the most practical way to provide total compatibility is with analog subcarrier inputs.

SUMMARY

The radio broadcaster can now implement a totally digital system from the audio source to the RF carrier using standard off-the-shelf hardware. The advantages of an all digital path include:

- The elimination of all intervening A/D and D/A conversions and the distortions they introduce.
- Absolute system stability and repeatability day-after-day, year-after-year.
- Greatly improved fade margin for radio links.
- Fully digital, distortionless, FM generation, assures that full digital quality is delivered to the "On-Air" signal without the noise and distortion build-up of an analog system.
- Easy interfacing between equipment without worries about level adjustment or hum pickup.

DIGITAL EXCITER BLOCK DIAGRAM

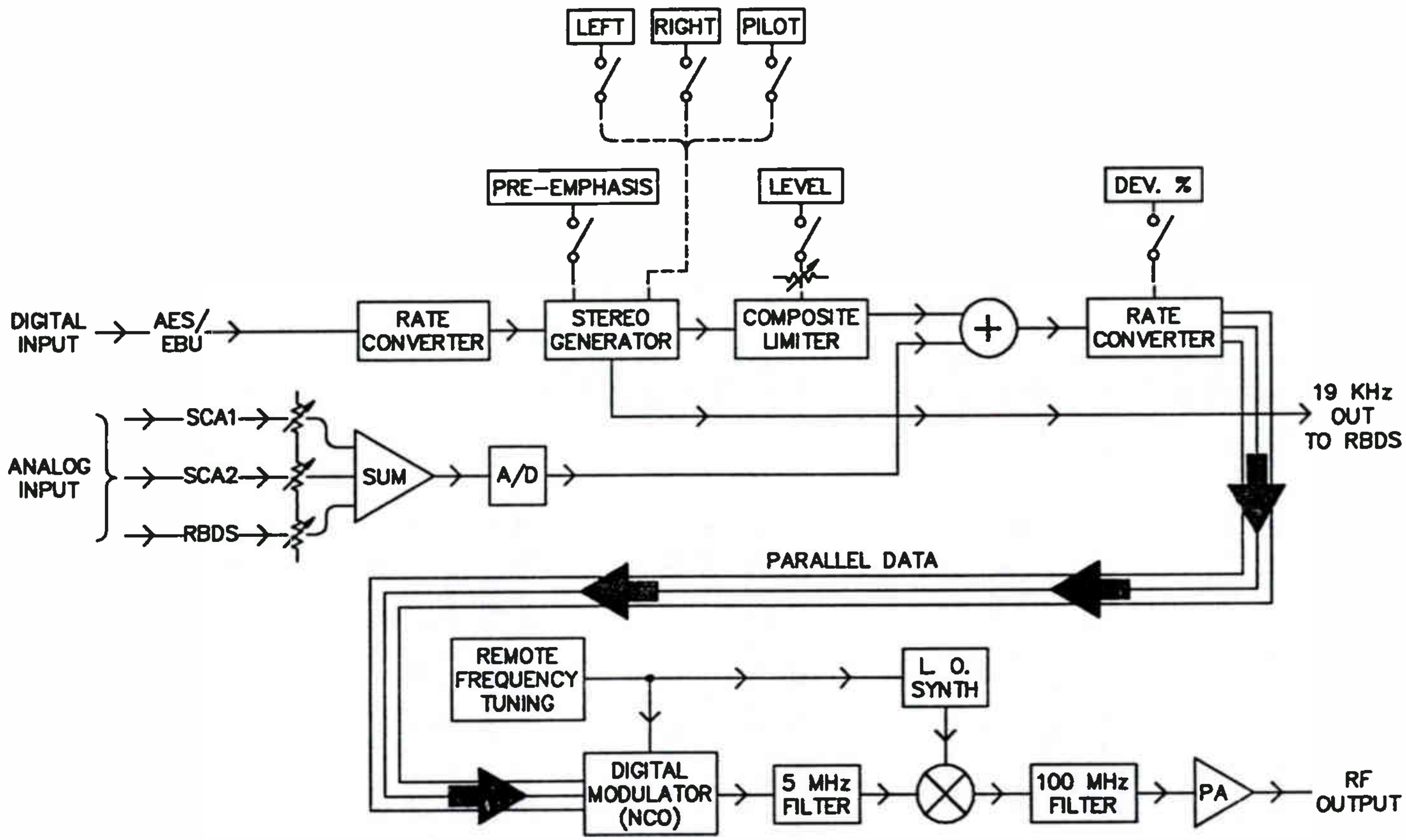

FIG. 3

- Absolute phase matching and differential phase stability between stereo channels.
- Absolute frequency response and amplitude matching between stereo channels.
- Perfectly linear modulation process eliminates the need for pre-distortion.

ACKNOWLEDGEMENTS

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THE INTERACTION OF AUDIO PROCESSING AND LOW BIT-RATE CODING IN BROADCAST APPLICATIONS

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Abstract: Audio processing to control peak levels is applied almost universally in broadcasting. Low bit-rate coding has begun to find application in both distribution and storage of audio. With the advent of low bit-rate digital audio coding, it is prudent to consider the potential interaction between them.

INTRODUCTION

Bit-rate reduction of digital audio works largely by exploiting the inherent inaccuracies of human auditory acuity. By a combination of techniques, the amount of data that needs to be stored or transmitted is reduced by eliminating that part of the signal which cannot be heard.

"The purpose of audio processing is to make most efficient use of the signal-to-noise ratio and audio bandwidth available from the transmission channel while preventing its overmodulation".[1] A second and important purpose of audio processing is to make the program sound louder. Efficient use of the available signal-to-noise ratio is made by controlling the average program level to render it more nearly constant, and by controlling the peaks such that they do not exceed the allowed maximum modulation. A necessary result of this type of processing is that the amplitudes and phases of the signal components acquire a particular relationship to each other. Any further processing of this signal which alters those relationships will undo some of the effect of the processing. Audio signals which have been processed to decrease their dynamic range are ***fragile***, in the sense that any subsequent processing may possibly increase their dynamic range.

Low bit-rate coders have as their main design criteria the reduction of data rate while maintaining the greatest possible audible fidelity to the original signal, without necessarily maintaining waveform fidelity. What is their effect on audio program material which has been processed for maximum modulation and loudness? Should audio processing be performed before or after the low bit-rate coder? Results are shown for musical signals with four different types of audio processing and three low bit-rate coders.

Broadcasters use signal processing to decrease the audio dynamic range to prevent overmodulation, increase coverage area, and have a louder signal. If an audio system is used to transmit such processed audio it may inadvertently increase the peak levels by any of the following mechanisms:

a. Low-pass phase shift alters the relationships of harmonics to their fundamentals for high frequencies and causes overshoots, or high-pass phase shift alters the relationship of the fundamental to the harmonics for low frequencies and causes waveform tilt.

b. Gibb's Phenomenon - If the bandwidth of the transmission channel is less than that of the signal, and the signal is transient in nature, the removal of harmonics results in overshoots, even if the phase response is linear.

c. Low bit rate coding adds quantization uncertainty at high frequencies. This uncertainty can result in undershoots and overshoots.

In order to measure overshoots caused by coder quantization, it is first necessary to eliminate overshoots caused by the first two effects.

MEASUREMENT METHOD

To measure overshoots in a system under test, it was necessary to develop a rational methodology for measuring peak levels throughout the duration of a processed musical selection. Only the peak level at a single instant in time has an unambiguous definition. For this study, peak levels were measured by a conventional analog peak meter circuit comprised of an absolute value circuit followed by a fast attack, slow decay peak detector circuit, with analog output to the data acquisition device. It is important that the peak meter should be repeatable to within 0.1 dB in order to accurately measure overshoots corresponding to increased modulation of 1%. The attack time should be fast enough to acquire a 15 kHz sine wave to within 0.1 dB in a half cycle, and then decay less than 0.1 dB in 0.25 seconds. The objective is to accurately follow the peak envelope with samples frequent enough to show the dynamics of the music, such as the beat and the attack transients of notes.

Very short transients may escape measurement by a conventional peak meter, and transients with a very large dynamic range tend to cause overshoots in the absolute value circuit. Digital oscilloscopes with peak capture capability are excellent for insuring that no peak escapes detection, although they tend to be limited in terms of their record length. The oscilloscope measurements presented in this paper were performed by sampling at 1 megasample/sec for 50 seconds with a record length of 2000. Each measurement therefore represents both the positive and negative peak value measured within each time window, which is .005 seconds in length. If the peak amplitude is at full level at the beginning of a measurement interval and falls to zero before the end of that interval, and remains there, then the time record will show full level for the first sample and zero for the second sample. This type of peak measurement results in a record of peak level which can vary much more dramatically from sample to sample than the analog peak meter because the analog meter incorporates a hold which is necessary in order for the voltmeter to settle.

Figure 1 below is an example of the extremely compressed dynamic range found in an atypical but not unique example of modern recorded music. The peak and rms levels for the second minute of the CD recording of "Hot Wire" by KIX were simultaneously measured, and then post-processed to give a running record of how the peak/rms ratio varied with time.

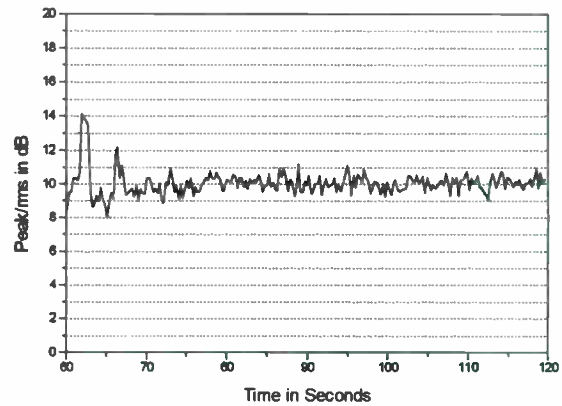

Figure 1: Peak/rms Ratio for "Hot Wire"

With the exception of two transients, the peak/rms ratio of the second minute of this selection remains between 9 and 11 dB.

This can be seen more readily in Figure 2 below, which is a histogram of the peak/rms ratios throughout the same minute of program material.

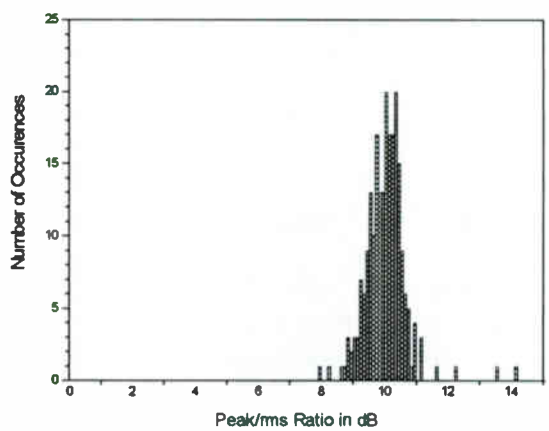

Figure 2: Distribution of Peak/rms in "Hot Wire"

The tight clustering of peak levels in the range of 9 to 11 dB can be seen, and two samples each representing the two transients to 12 and 14 dB peak/rms ratio.

AUDIO PROCESSING

Audio is processed for broadcasting in order to control its peak levels and to maximize its loudness. Whenever the term "audio processing" is used in this paper, it refers to this type of signal processing. Designers of these processors combine slow broad-band compression, faster multi-band compressors, phase scrambling, and peak clippers, to deliver maximum reduction of dynamic range with minimum degradation of sound quality. The phase scrambling, compression, and limiting keep the average levels more nearly constant while the peak clipper imposes a hard maximum on peak levels. The result of this is that the peak-to-average or peak-to-rms ratio tends to remain nearly constant with aggressive processing.

A model of the most aggressive possible processing is a full limiter followed by low-pass filtering. The limiting reduces the audio to a rectangular waveform with a 0 dB peak/rms ratio, and with information contained only in the zero crossings. The low-pass filtering is necessary to constrain the spectrum to the bandpass of the transmission or storage medium. However, the low-pass filtering increases the peak/rms ratio. This type of processing when applied to speech results in a peak/rms ratio of 5.1 dB.[2] Commercially available audio processing units reduce the peak/rms ratio of music from the range of 14-20 dB[3] to the range of 8-12 dB[4], with relatively little degradation in sound quality. Only a modest amount of processing is necessary to cause most of this reduction in dynamic range, and when additional processing is applied the perceived sound quality degrades very rapidly with each small improvement in reduction of dynamic range.

A waveform with very low peak-to-rms ratio necessarily has a certain phase relationship between its various harmonic components. Any alteration of that phase relationship will tend to cause components that originally interfered with each other to add constructively. The result is an overshoot. Any system that is not linear phase will cause these overshoots.[1]

EFFECT OF HIGH-PASS FILTERING

It has been reported that phase shift due to the low frequency cutoff of systems, such as STLs and exciters, in the audio broadcast signal path cause overshoots with highly processed low frequency material. If the audio signal contains substantial low frequency content, and the processing limits those low frequencies until they are roughly rectangular, then the tilt in the flat top of the waveform will result in increased peak levels. In order to pass a 50 Hz square wave with less than 1% increase in peak level, a low frequency bandwidth of 0.16 Hz is needed.[5]

An experiment was performed to analyze the effect of high-pass filtering on processed audio. This experiment tests the magnitude of this effect for musical signals.

The measurements were performed with a popular audio processor adjusted to perform the maximum possible processing at low frequencies. The distribution of peak levels, after this processing was applied, during 50 seconds of a pop recording[8] with substantial low frequency content is shown below in Figure 3. The processor was DC coupled into the measurement system.

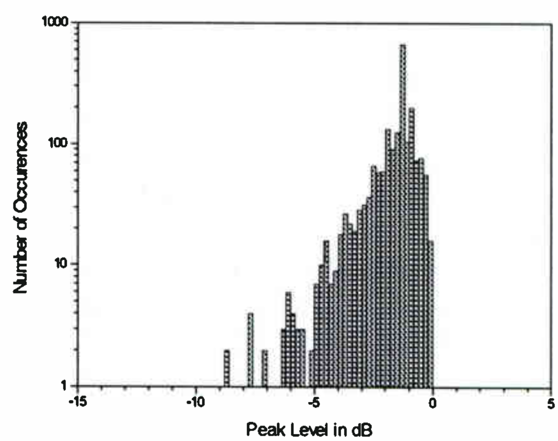

Figure 3: Peak Levels after Processing and with Low Frequency Extension to DC

The peak levels are normalized to 0 dB for this measurement, which is the reference for the two measurements which follow.

The second measurement was of the peak levels in the same 50 seconds of music, but high-pass filtered at 10 Hz. The distribution of peak levels is shown below in figure 4.

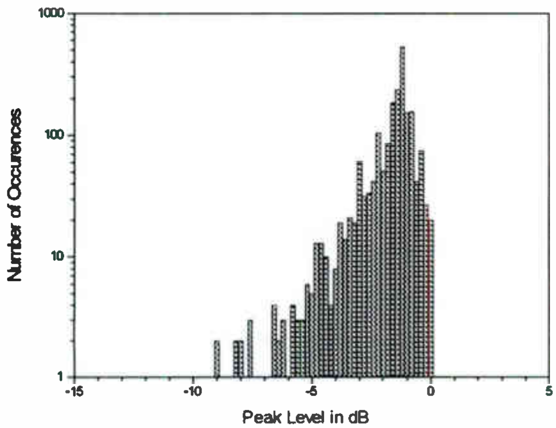

Figure 4: Peak Levels after Audio Processing and High-Pass at 10 Hz

The distribution of peak levels is very similar, but with an 0.1 dB increase in peak levels indicated. This measurement only has a resolution of 0.2 dB so this increase is probably not significant.

The last experiment uses the same 50 seconds of music as in the previous two, but high-pass filtered at 20 Hz. The results are shown below in Figure 5.

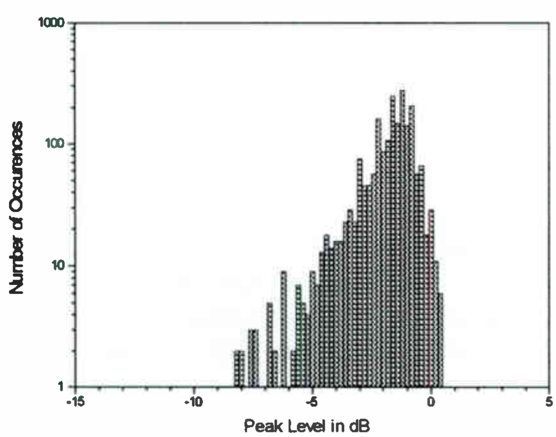

Figure 5: Peak Levels after Audio Processing and High-Pass at 20 Hz

Now the distribution of peak levels can be seen to have changed shape, and the peak levels have increased by 0.5 dB relative to the DC coupled condition. Although the overshoots with a 50 Hz squarewave and 10 Hz bandwidth would be 2.2 dB, it was found in this series of experiments that low frequency extension down to only 10 Hz was necessary to keep overshoots due to this effect on real-world program material less than 0.1 dB.

LOW-PASS FILTERING AND GIBB'S PHENOMENON

The vast majority of musical program material today has been digitized at a sample rate of 44.1 kHz, and consequently is limited in bandwidth to a maximum of about 20 kHz. Audio which has been processed for FM broadcast is limited by the processor to 15 kHz bandwidth. These two bandwidth reductions reduce the probability of overshoots caused by high frequency bandwidth restriction in the coder.

All three of the coders evaluated in this report use delta-sigma Analog to Digital converters operating at an oversampling ratio of 64. This allows the use of a simple linear phase anti-alias filter and eliminates the possibility of overshoots due to the filter itself. Likewise, overshoots due to Gibb's Phenomenon were not observed.

LOW BIT-RATE CODING

Low bit-rate coders function by dividing the audio spectrum into bands, analyzing the signal to determine which components are judged to be audible, and assigning an appropriate number of bits to quantize the signal within each band. Reductions of data rate of 4:1 to 8:1 with minimal loss of fidelity are possible at the current state of the art. The focus of low bit-rate coders is <u>spectrum reproduction</u>, not <u>waveform reproduction</u>. Different coders use different means for division of the spectrum, decision of audibility of spectral components, and allocate the available bits each according to their own criteria. Because the acuity of human hearing is substantially less at high frequencies (7 kHz to 20 kHz), fewer bits are used to reproduce that part of the audio spectrum. This leads to measurable quantization uncertainty at the highest audio frequencies (15 kHz for broadcast).

The use of pre-emphasis exacerbates the problem. The perceptual model used in low bit-rate coders assumes that the reproducer has flat response. If de-emphasis is used after the coder, then low frequency quantization noise which was masked by a louder high frequency sound may become audible when the high frequency level is reduced by the de-emphasis. In order to prevent this, one coder de-emphasizes pre-emphasized audio in the digital domain before the bit-rate reduction process. After the audio has been decoded, the pre-emphasis is reapplied with exact complementarity to the de-emphasis. This assures that the spectrum of the audio presented to the coder is "flat", and that its perceptual model is applied in the correct domain.

Re-application of the pre-emphasis after the coder causes the amplitude uncertainty to be much greater at high frequencies since the emphasis amounts to approximately 17 dB at 15 kHz. This is a system problem. If pre-emphasized audio is applied to the low bit-rate coder, then the eventual de-emphasis may cause otherwise inaudible coder errors to become audible. If the pre-emphasis is applied after the coder, then the high frequency overshoots of the coder will be exaggerated

All of the measurements on coders presented in this paper are for data rates of 182 - 192 kBits/sec per channel. This data rate was available for the three coders tested and is representative of the data rate needed for the highest quality. These coders reproduce an audio bandwidth to either 15 kHz or 20 kHz. The coders were adjusted as closely as possible to unity gain from analog input to analog output.

The following sequence of measurements illustrates the change in the distribution of peaks for an audio system which includes audio processing to control the peak levels, followed by a low bit rate coder.

Each of the following three figures shows the peak levels in the first 50 seconds of the song "Baby Baby" by pop vocalist Amy Grant[9]. The graphs are time records that show the greatest value, positive or negative, in each .005 second interval. The music is instrumentally dense and every part of the audio spectrum is occupied. Figure 6 shows the peak levels for the unprocessed music directly from CD.

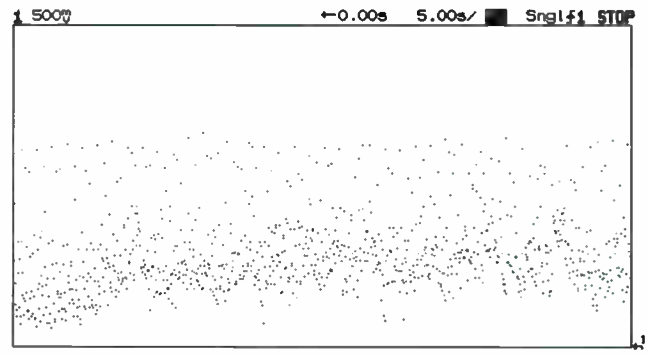

Figure 6: Unprocessed Music

The 2000 measured peak levels can be seen to be distributed over a wide range with most clustered around a level about one third of the amplitude of the maximum peak level.

The same 50 seconds of music was applied to an audio processor with dispersion, compression, and limiting contained within a 2-box system. The results are shown below in figure 7.

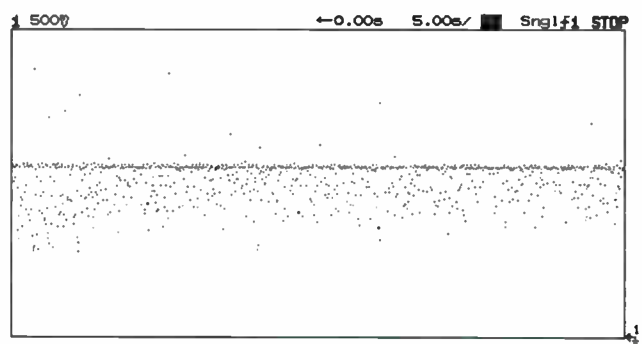

Figure 7: Processed Music

The peak levels can be seen to be well controlled to a specific level, with the exception of 14 incidents which exceed that level by as much as 4.5 dB. This particular processor is not successful at absolutely limiting the peak levels but does in general compress and limit the signal in such a way that the peak level is at or below the defined maximum nearly all of the time.

When this signal with tightly controlled peak levels is applied to a low bit-rate coder/decoder operating at 192 kBits/sec, the peak levels are modified as shown below in Figure 8.

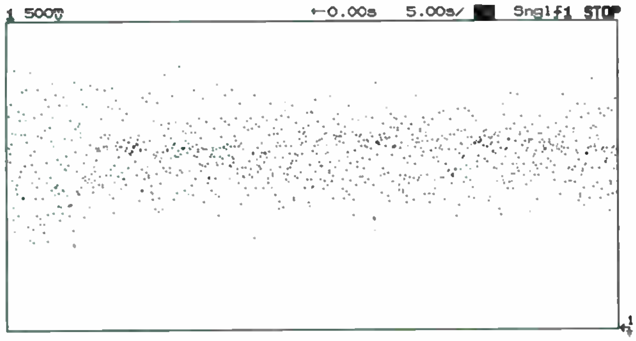

Figure 8: Processed Music after Coding

The previously well controlled peak levels can be seen to have been dispersed both above and below the previously well defined maximum. That original ceiling is still visible as a poorly defined line in the center of the cloud of peak levels. These errors in peak level are essentially overshoots (although they do not meet the functional definition given to a linear overshoot). Some of these overshoots are greater than 3 dB.

Additional experiments were performed with an audio processor which contains phase scrambling, compression, limiting, pre-emphasis, and a stereo generator within a single unit. For these measurements the stereo generator was bypassed and the processed audio was measured either directly or after application to a low bit-rate coder.

The low bit-rate coder used was Dolby AC-2 as implemented within the Dolby DSTL®. This product is equipped with complementary de-emphasis before the encoder and (re)pre-emphasis after the decoder. In addition, a high frequency limiter is implemented within the decoder to reduce any high frequency overshoots which may occur due to coding quantization uncertainty and to the post coding pre-emphasis. This limiter acts only when high frequency overshoots occur and has minimal subjective effect.

Figure 9 below shows the distribution of peak levels during the first 50 seconds of the vocal selection "Baby Baby" by Amy Grant[9], measured at the output of the audio processor with pre-emphasis.

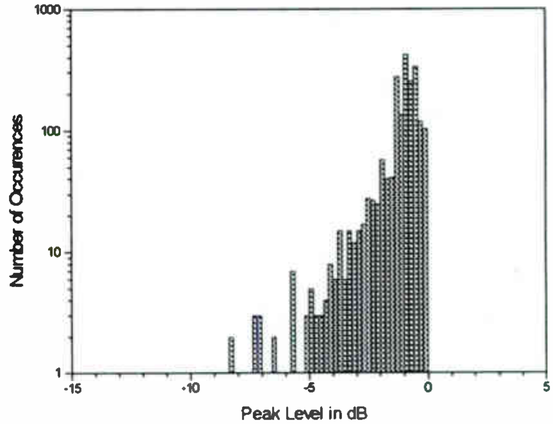

Figure 9: Peak Levels after Processing

The distribution of peak levels can be seen to have been very tightly constrained with no peaks greater than the arbitrary zero level chosen for this graph. The great majority of peaks are within 3 dB of full level.

The processed audio was then applied to the DSTL with the limiter disabled. The resulting distribution of peak levels is shown below in Figure 10.

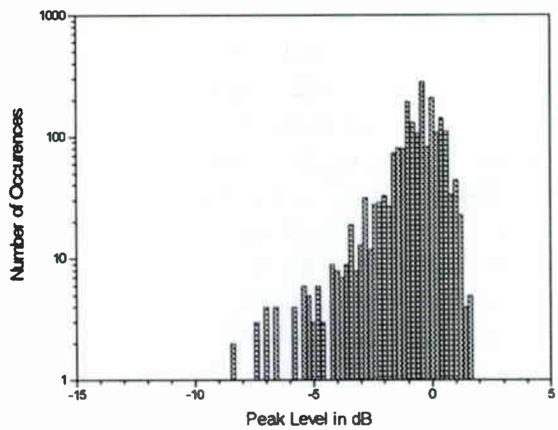

Figure 10: Distribution after Processing and Coding

The distribution of peak levels can be seen to have broadened and overshoots up to about 1.6 dB ± 0.2 dB above the original peak level have been added. Other coders tested produce similar or greater overshoots when followed by pre-emphasis.

The overshoot limiter in the DSTL acts only on high frequency overshoots and its effect is shown in Figure 11 below, which is the distribution of peak levels for the same processed musical selection but with the limiter enabled.

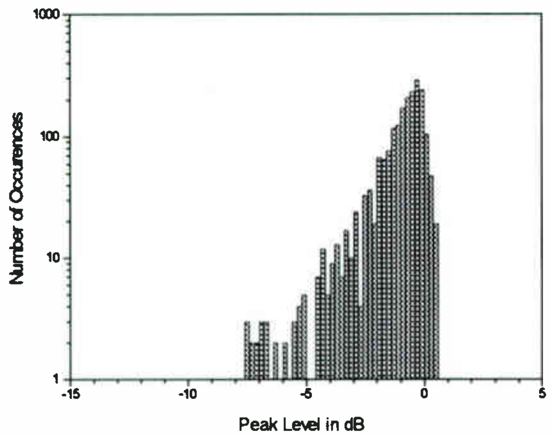

Figure 11: Peak Levels after Audio Processing and Low Bit-Rate Coding with Limiter

The distribution can be seen to have been narrowed and the overshoots constrained to less than 0.5 dB.

CONCLUSION

It has been shown that audio signal processing can be successfully used to reduce dynamic range and to control peak levels for purposes of broadcasting. The distribution of peak levels measured over time with a test signal or with music is a useful way of visualizing the effect of that processing. The effect of low frequency overshoots due to lack of extended low frequency bandwidth was investigated and it was found that low frequency bandwidth of 10 Hz was sufficient to constrain overshoots to within 0.1 dB for all of the musical sources and audio processors evaluated.

Some low bit rate coders may increase the peak/rms ratio of some musical material, especially if that material has been compressed to a low peak/rms ratio and if its spectrum contains substantial high frequency content due to pre-emphasis. A limiter integrated into the coder is shown to substantially reduce these overshoots.

If audio processing is applied prior to low bit-rate coding, overshoots may occur in the coder due to quantization uncertainty at high frequencies. A limiter either integrated into the coder, or applied further on in the transmission chain can be used to control those overshoots. If the processing is applied after the low bit-rate coder, that processing can conceivably increase the amplitude of previously inaudible quantization noise from the coder, until it is above the threshold of audibility. Both pre-processing and post-processing can be made to work well, but the users should be aware of the potential problems.

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MANAGING IN BROADCAST ENGINEERING

Monday, March 21, 1994

Moderator:
Margaret Bryant, WMAQ-AM, Chicago, IL

AN INTRODUCTION TO ENVIRONMENTAL REGULATIONS AFFECTING BROADCAST FACILITIES
Daniel Boone
Camp, Dresser, & McKee
Pittsburgh, PA

***THE ADA AND BROADCAST FACILITY DESIGN**
Frank Rees, AIA
Rees Associates
Irving, TX

***BACK TO SCHOOL: CONTINUING EDUCATION POSITIONS BROADCAST ENGINEERS TO HANDLE TOMORROW'S TECHNOLOGY TODAY**
Corey Carbonara
Baylor University
Waco, TX

*Papers not available at the time of publication

AN INTRODUCTION TO ENVIRONMENTAL REGULATIONS AFFECTING BROADCAST FACILITIES

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Camp, Dresser, & McKee Inc.
Pittsburgh, Pennsylvania

Abstract

The operation of broadcast facilities may include activities that require compliance with environmental regulations. Potentially applicable federal regulatory programs include underground storage tanks, hazardous material use, waste management, water quality, air quality and toxic substance management. This paper introduces these programs and summarizes their requirements for typical broadcast facilities. This introduction will serve as a foundation for operations engineers and personnel to determine regulatory requirements and comply accordingly on a facility specific basis.

Introduction

Regulations designed to protect the environment have been in existence since at least the 1950's. In the last ten years, however, there has been a substantial increase in the number and scope of these regulations. There is now the potential for citizen lawsuits, substantial fines and criminal prosecution by regulatory agencies against businesses and industries which do not comply with environmental regulatory programs. While large corporations typically have environmental lawyers and engineers responsible for tracking and complying with the regulatory programs, smaller businesses such as broadcast facilities may not be aware of environmental regulations or have the resources to comply with them.

This paper introduces the federal environmental regulatory programs and requirements which apply to typical broadcast facilities. Individual facilities may have unique operations which require compliance with additional environmental regulations not described here. Many state and local regulatory agencies also have environmental regulations which must be complied with, in addition to the federal regulations. This paper is intended to provide an overview of the types of environmental regulations which may impact broadcast facilities and serve as a general reference for facility engineers and operations personnel.

The major federal environmental regulations which are described in this paper fall under the categories of Underground Storage Tanks, Hazardous Materials Use, Waste Management, Water Quality, Air Quality and Toxic Substance Management.

Underground Storage Tanks

Underground storage tank (UST) regulations were developed to protect the environment from damage related to releases of hazardous materials resulting from spills or leaks. These regulations apply to any tank which is used to store hazardous material and has at least 10 percent of its capacity below ground. They include requirements for design, construction and installation; operation and maintenance; leak detection; release reporting and investigation; corrective action; closure and financial responsibility.

The diesel USTs present at many broadcast facilities which store fuel for emergency generators must comply with these regulations. Those that were installed by the Federal Emergency Management Agency (FEMA) under the Broadcast Station Protection Program are in the process of being brought into compliance with the UST regulations or are being replaced by aboveground storage tanks. Those that were installed by broadcast facilities are the responsibility of the station and must be brought into compliance with current regulations. These regulations are typically implemented by the local health department or fire department.

Hazardous Materials Use

Facilities which use hazardous materials are required to take measures to protect the public and their employees from the risks associated with these materials. Hazardous materials which may be used at broadcast facilities and are subject to these requirements include industrial cleaners and solvents, chiller chemicals and diesel fuel. Employees are protected under the Occupational Safety and Health Administration (OSHA) Hazard Communication Standard. This standard is enforced by the federal or state OSHA. SARA Title III regulations are designed to protect the public from the risks associated with the release of hazardous materials from facilities. Because Title III regulations are triggered by use of hazardous materials in quantities in excess of 10,000 pounds, it is unlikely that most broadcast facilities would be required to comply with them.

The hazard communication standard requires that the hazards associated with use of hazardous materials be evaluated and that purchasers of these materials and their employees are informed about the hazards and how to minimize them. The means through which this information is communicated is the Material Safety Data Sheet (MSDS). A manufacturer of a hazardous material must supply a MSDS with its sale, and the facility using the hazardous material must keep the MSDS on site and available to all employees who may be exposed to the material. Warning labels must also be provided on the hazardous material container, stating the identity of the material, the hazards associated and the manufacturer's name and address. Any employee who uses these materials or is exposed to them must be trained upon hire or any time a new material is introduced. The facility using these hazardous materials is responsible for the development of a written hazard communication program which details the procedures developed and implemented for maintenance of MSDS files, container labeling and employee training in accordance with these requirements.

Waste Management

Wastes generated from the use of hazardous materials should be managed in a manner that minimizes the risk of release of these wastes to the environment. Water quality protection regulations prohibit the disposal of most hazardous materials to wastewater treatment plants through the sewer system. Releases of these wastes directly to the environment are also prohibited and may result in facilities having to pay for the investigation and clean up of related soil, water or groundwater contamination. These requirements are typically enforced by the local or state regulatory agencies.

In addition to these requirements, wastes which are listed or hazardous by characteristic under the Resource Conservation and Recovery Act (RCRA) must comply with specific federal regulations for their storage, transportation and disposal. Most broadcast facilities will not use materials that generate hazardous waste. The exception might be some cleaning solvents and photographic laboratory wastes. Information regarding the classification of wastes and proper management of wastes created from the use of hazardous materials may be found in the MSDS for that material. State or local health agencies are typically responsible for implementing these regulations.

Water Quality

The Clean Water Act (CWA) regulates the discharge of pollutants into all navigable waters of the United States and is intended to improve and maintain the quality of the nation's waterways. The CWA has several major regulatory programs which regulate direct and indirect discharges, as well as the potential for releases to surface waters. Typical activities at broadcast facilities which would trigger CWA compliance include discharge of wastewater to public wastewater treatment plants, exposure of industrial materials to rainfall or snow, and storage of petroleum products.

The discharge of treated wastewater from publicly owned treatment works (POTW) is regulated under the CWA through discharge permits. Facilities which discharge wastewater to the POTW sewer system must comply with pretreatment requirements which protect the POTW ability to operate and comply with their permit. Although mandated by federal regulation, specific pretreatment requirements are developed and enforced locally by each POTW. These may include general and specific wastewater discharge limitations or prohibitions (such as flow volume and rate, pH or oil and grease concentrations), wastewater quality and flow monitoring and reporting requirements. Most broadcast facilities only generate wastewater as a result of support operations such as heating, equipment cooling and facility maintenance. Because these discharges may not be particularly toxic or large in volume, POTW may either be unaware of their existence or consider them insignificant in terms of impact on their CWA compliance. Negligence on the part of the POTW, however, does not exclude broadcast facilities from complying with CWA regulations.

In order to protect water quality from impacts related to pollution in stormwater, the CWA stormwater permitting program was recently established. Under this program, broadcast facilities which store industrial materials such as chemicals, wastes and packaging material outside and exposed to rainfall, must obtain a stormwater permit and comply with its requirements. These typically include stormwater discharge monitoring and reporting and development and implementation of a stormwater pollution prevention plan. The stormwater permit program is implemented by the state water quality agency or the EPA.

Broadcast facilities which store petroleum products in total quantities greater than 42,000 gallons below ground or 1,320 gallons aboveground (or 600 gallons for a single aboveground tank) may be subject to spill prevention requirements under the CWA. If a release from such a tank could reasonably be expected to reach surface waters, a spill prevention control and countermeasure (SPCC) plan must be prepared and implemented. The SPCC plan includes information about the facility, its operations and petroleum product storage. Specific procedures for avoiding spills and responding in the event a spill occurs must be developed and implemented. The SPCC plan must be certified by a professional engineer, kept on site for EPA review if requested and updated every three years or when conditions at the facility change. If spills do occur in excess of regulated quantities, they must be reported to the EPA.

Air Quality

The Clean Air Act (CAA) was developed to protect public health and welfare from air pollution. Although the CAA requires the EPA to set air quality standards, the responsibility for implementing CAA regulations is assigned to state and local agencies. These regulations apply to stationary sources such as equipment exhaust stacks and mobile sources such as automobiles.

Under the CAA, air emission sources must be permitted and meet design, construction and operation standards. The permits may specify emission limitations depending on the air quality in the region of the source and on the types of pollutants expected to be emitted. Emission sources at broadcast facilities which may be regulated include fuel storage tanks, boiler stacks and solvent exhaust. The location of the facility will determine if these requirements will be implemented by local air quality agencies or the state. Larger metropolitan areas typically have local air quality agencies and additional regulatory requirements. Because motor vehicle exhaust contributes a substantial portion to air polluting discharges, the CAA also regulates automobiles and other mobile air emission sources in locations where air quality doesn't meet federal standards (non-attainment areas). These regulations affect broadcast facilities located in non-attainment areas which have greater than 100 employees or fleets of more than 10 vehicles which are centrally fueled. Reductions in mobile air emissions from employee vehicles are required through car pooling or alternate transportation. Motor vehicle fleets must reduce air emissions through operational practices such as fuel substitution or vehicle modifications.

Toxic Substances

The Toxic Substances Control Act (TSCA) regulates the manufacture, use, distribution and disposal of a group of substances which present risks to public health and the environment. PCBs are the TSCA substance that are most likely to be present at broadcast facilities and trigger TSCA compliance requirements. Although the manufacture and use of PCBs was prohibited under TSCA, provisions were made to allow their use in a totally enclosed manner provided specific conditions are met. Broadcast facilities which own PCB-filled electric transformers, capacitors, rectifiers and other electrical equipment must comply with these requirements.

The requirements for PCB-filled equipment are intended to prevent the release of PCBs to the environment or exposure of the public. Their details vary according to the concentration of PCBs present. In general, however, PCB-filled equipment must be in good condition, with no leaks or major servicing required. It must be located away from food products, ventilation ducts and combustibles to avoid exposure if releases did occur. PCB-filled equipment must also be marked, labeled according to the regulations, and inspected regularly. Recordkeeping and reporting of its use and maintenance must be completed. Once the equipment is taken out of service, specific requirements for its storage and disposal must be complied with. These requirements also apply to PCB-contaminated materials such as soil impacted by a PCB release.

Summary

Although broadcast facilities do not have and industrial classification, many of the activities necessary to support their operation trigger requirements for compliance with federal environmental regulatory programs. Few of these programs are implemented by the EPA; most often state and even local regulatory agencies are responsible for developing and enforcing the regulations. In order to avoid the potential liabilities associated with non-compliance, broadcast facilities need to assess their operations, identify which major regulatory programs apply and determine specific regulatory requirements at the local or state level. Compliance with these requirements in the most cost-effective manner then becomes the operations engineer and personnel's challenge.

HDTV STATION ISSUES, PART II: UHF TRANSMISSION

Monday, March 21, 1994

Moderator:
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FULL BAND LINEARITY CORRECTION AND OTHER REQUIREMENTS FOR ATV TRANSMITTER SYSTEMS
Nat Ostroff
Comark Communications, Inc.
Colmar, PA

***ADVANTAGES OF THE KLYSTRODE® IOT IN HDTV SERVICE**
Chris Johnson, Heinz Bohlen and Thomas Stephenson
Varian Associates
San Carlos, CA

***KEEPING AND IMPROVING THE NTSC TRANSMITTER (VHF & UHF) AS YOU TRANSITION TO HDTV**
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ITS Corporation
McMurray, PA

***DIGITAL AMPLITUDE MODULATION: UHF TV TRANSMITTER FINAL TESTS**
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Acrodyne Industries, Inc.
Blue Bell, PA

PRE-CORRECTION TECHNIQUES USED IN THE DIGITAL TRANSMITTER
Jesus Diaz-Reganon
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*Papers not available at the time of publication

FULL BAND LINEARITY CORRECTION AND OTHER REQUIREMENTS FOR ATV TRANSMITTER SYSTEMS

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Colmar, Pennsylvania

COMARK *has provided and supported a "**DUAL USE**"™ IOT UHF transmitter for the required Field Testing of a newly proposed Digital HDTV (D-HDTV) ATV standard. This paper will discuss the linearity requirements for the high power transmission of this new standard.*

*This paper deals with the need for, and development of, analog "FULL BAND" linearity correction in "**DUAL USE**"™ and ATV only applications. There will be a discussion of the technologies being introduced, and results achieved. A primary focus will be the implication for future "D-HDTV" compatible products.*

I. Introduction

At the time of this writing, the "Grand Alliance" is in the process of defining the transmission modulation method. The decision will be made after a "bake-off" of systems at the Advanced Television Test Center (ATTC). Under consideration at this time are 32 state Quadrature Amplitude Modulation (32 QAM), as well as 4 & 6 level Vestigial Sideband (4 VSB & 6 VSB). **COMARK** will be investigating modulation methods testing on the Inductive Output Tube (IOT) fitted equipment in Charlotte, N.C. towards the end of January, 1994, and the results obtained there will be shared when available.

II. History

Comark introduced the world's first inductive output tube (IOT) based UHF television transmitter in 1986[1]. By 1990, a number of transmitters using the Eimac inductive output tube, called a Klystrode®, were in broadcast service. In 1991, a second generation, more user friendly IOT, was brought to the market by EEV and Comark and it was placed in broadcast service[2]. In 1992, Comark introduced the third generation inductive output tube transmitter, the "IOX" series. It is some of the design considerations for the "IOX" that will be discussed.

III. System Requirements

The small size, high efficiency, Class AB operation and linearity of the IOT has made it the "device of choice" for Advanced Television (ATV) service. The integration of two (2) main operational considerations has led to the development of an advanced "Full Band" linearity correction scheme. Those two (2) considerations which were integrated are:

1. Maximum system flexibility with no compromise to conventional television performance

2. Optimized HDTV compatibility

The IOT is a superior amplifying device operating in Class AB. The gain of the tube is above 20 db and its transfer characteristic is nearly linear, with well behaved turn on and saturation regions. Early in IOT development efforts it became clear that the tube's linearity was sufficient to permit common amplification operation. Common amplification implies full specification compliance, while the commonly used term of multiplex implies emergency substandard operation.

Common amplification in the IOT is possible because its transfer curve is easily corrected. The uncorrected third order intermodulation performance of the IOT is close to translator standards of -50 db. With normal correction applied, IM levels of -56 db are easily achieved. With state-of-the-art correction on both the visual and aural carriers, IM levels better than -62 db are achieved and IM levels of -60 db can be guaranteed.

An aural carrier corrector[3] (Figure 1) provides amplitude and phase modulation to the aural carrier to correct the cross modulation that the aural carrier receives from the visual carrier in common amplification applications. Without aural carrier correction, the common amplification based transmitter will not meet FCC Specification 73.682(c)(3) for stereo pilot protection or FCC Specifications 73.1570(a) and 73.1570(b)(3) for maximum modulation levels. Comark developed and patented an Aural Carrier Corrector for this purpose.

IV. D-HDTV/ATV Compatibility

The "IOX" series transmitter line not only meets uncompromised conventional television standards, but it also is compatible with future digital transmission technologies.

While no digital transmission standard has yet been selected, we have drawn some general conclusions about what will be required from the transmitter system, based on the results of testing with early ATV systems.

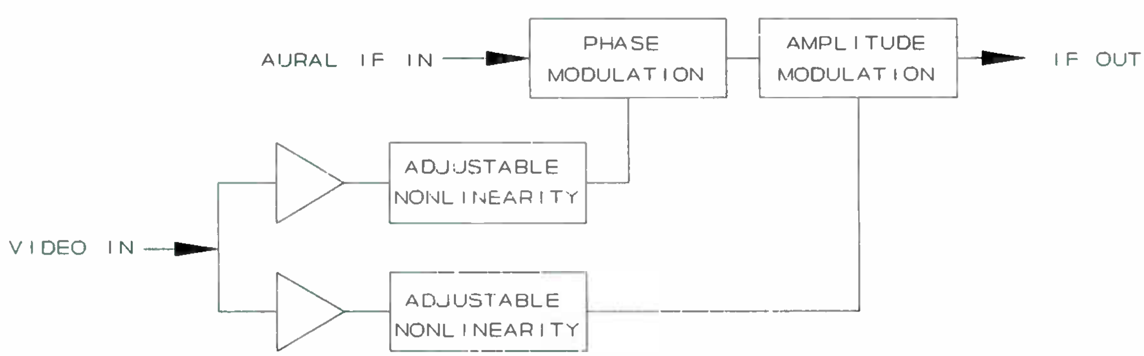

Figure 1 - Simplified Block Diagram
Aural Carrier Corrector - Patent #5,198,904

Early ATV transmissions were conducted around 16 QAM and 4 VSB formats. Previous papers describe some of these systems in detail[4,5]. Generally it can be stated that ATV RF signals are similar to white noise. The signals are characterized by their average power and peak to average power ratio in addition to their occupied bandwidth.

Developments in D-HDTV/ATV and the "Grand Alliance", has moved the data rate requirements on the transmitted signal higher. Thus, 16 QAM has moved to 32 QAM and 4 VSB to 6 VSB. The effect of this increase is to place tighter restrictions on the data's level and position. The correct position of data on the RF carrier can be corrected by including features in the digital systems to correct for transmitter amplitude and phase non-linearity.

However, this correction may not correct for the intermodulation generated in the non-linear power amplifiers in a transmitter. This intermodulation has the effect of adding noise to the transmitted signal and reducing its effective range.

As well, the ATV transmitter system is presently being defined as beginning at the 41-47 MHz IF input, and it only makes sense that the ATV transmitter system be "transparent".

Figure 2 is illustrative of the conventional analog techniques used to linearize a common amplification signal before the "IOX" series was produced. In this technique, a series of adjustable non-linear elements are inserted into the low level IF signal path.

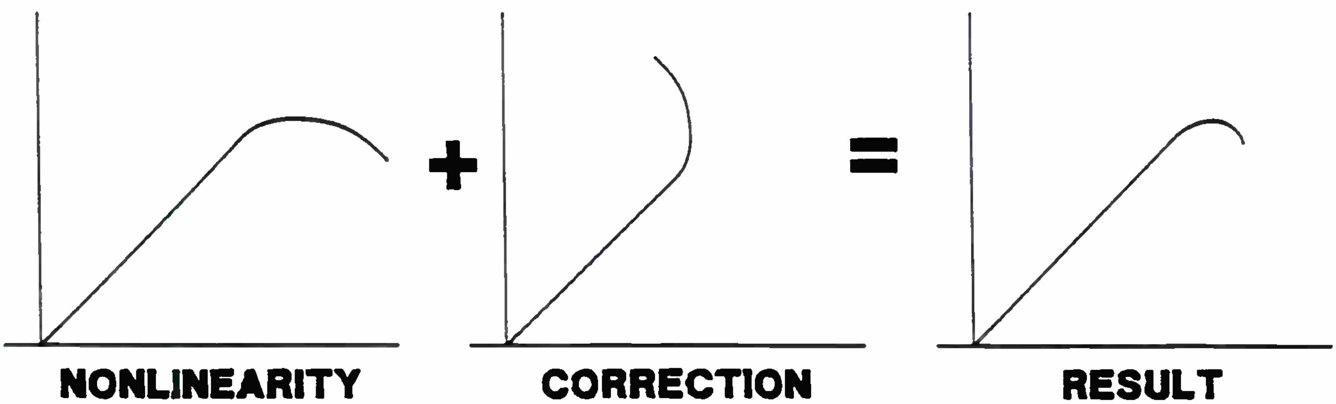

Figure 2 - Conventional Linearity Correction

These elements are adjusted to produce a nulling effect on the third order intermodulation product at ± 920 kHZ from the visual carrier in conventional NTSC systems. While the non-linear elements used in the technique are wideband, the effect of concentrating their effects on the 920 kHZ IM makes the overall technique narrow band in its final effect.

This is true because the matching of sideband amplitude and phase at 920 kHZ, in order to produce cancellation, ignores the amplitude and phase variations of the power amplifier chain at other frequencies across the band. Figure 3 illustrates the narrow band nature of conventional linearity correction. Note the increase in IM levels as the IM moves from the visual to the aural carrier frequency.

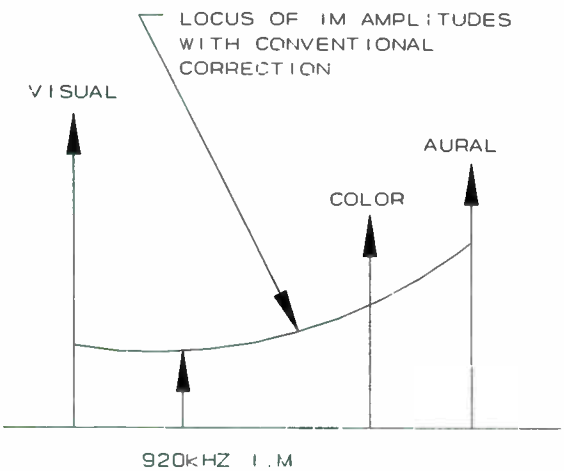

Figure 3 - Performance of Conventional Linearity Correction across the TV Channel

The requirement on the "IOX" series transmitter to be fully ATV compatible forced the designers to develop a linearity correction system that provides continuous correction over the entire channel[6]. This full band linearity correction takes the amplitude and phase variations of the power amplifier chain into account. In conventional operation, the TV signal benefits from lower distortion at all frequencies. In ATV, the transmitter provides vastly improved performance over conventional linearity correction techniques. Figure 4 shows the effects of a "Full Band" linearity corrector.

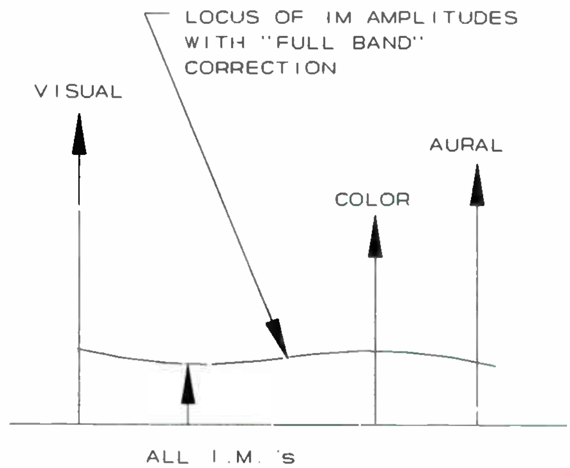

Figure 4 - Performance of "Full Band" Linearity Correction across the TV Channel

One of the prerequisites for the success of the full band linearity corrector was the minimization of unnecessary non-linearities in the power amplifier chain itself[7]. This was accomplished by providing full Class A linear amplifiers up to the output IOT. Thus, the full bandwidth corrector only had to be adjusted for the output stage non-linearity. This avoided serious increases in complexity and adjustment difficulty.

Figure 5 illustrates the measured effect of conventional correction on a 16 QAM signal transmitted from an IOT transmitter. Note the out-of-band intermodulation levels. This correction provides a 10 db improvement in the energy spurred out of the channel. This out-of-band energy is also indicative of what is produced in-band. The in-band IM energy adds noise to the data and degrades the range of the transmitted signal as previously discussed. The testing which will be done in January, 1994 will provide results with the incorporation of "Full Band" linearity correction, and current simulations indicate that an additional 6-10 dB of improvement can be expected. This falls close to that which has been specified for future ATV systems out-of band requirements.

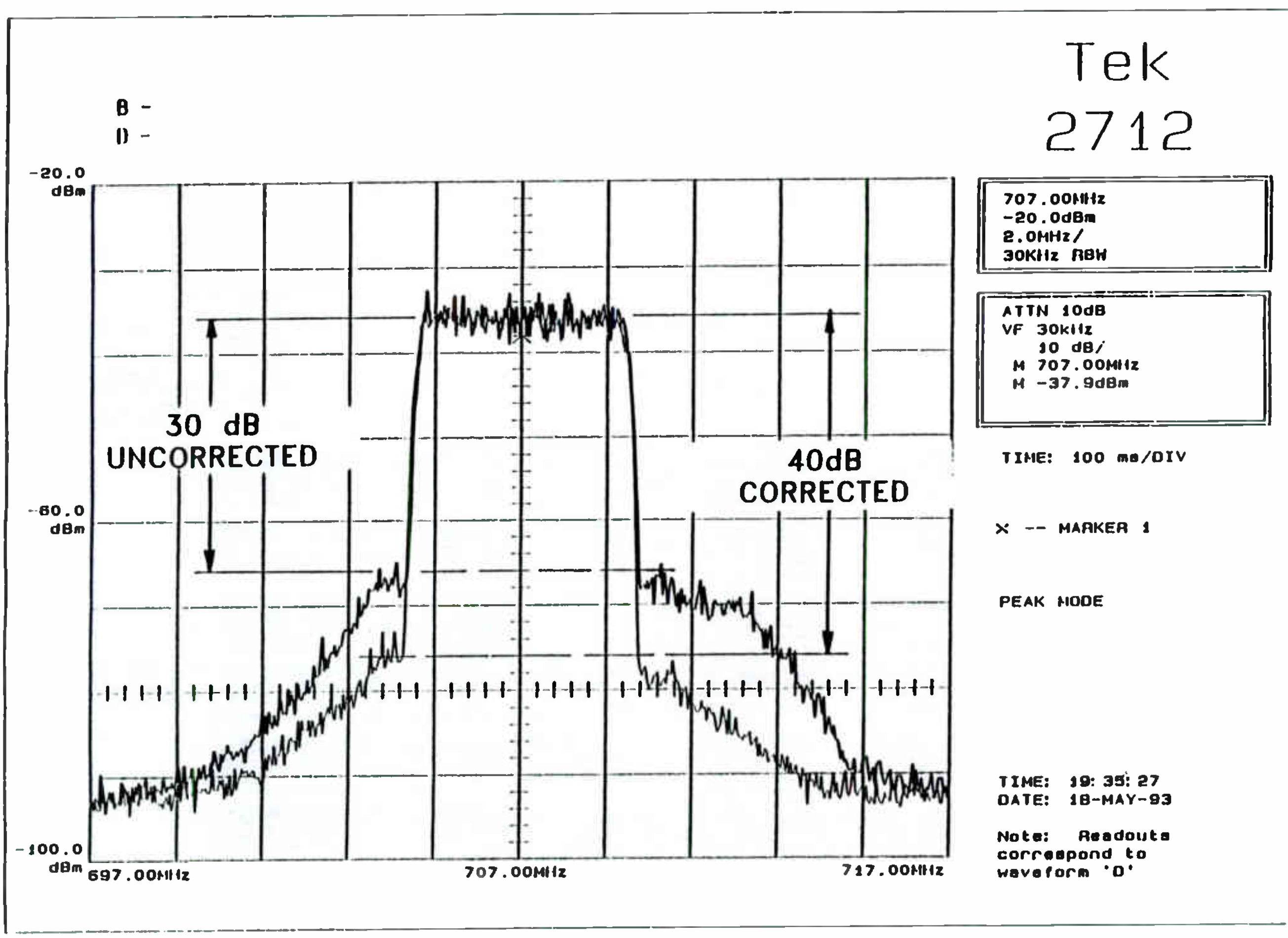

Figure 5 - Conventional Linearity Correction with/without Correction

Comark has decided to equip the "IOX" series with Class A solid-state drivers. This decision has been made based on the linearity and simplicity benefits achieved with Class A over Class AB. In addition, the effect of Class AB drivers is to require more complicated correction and to reduce the effectiveness of the overall correction result. The linearity advantages of using Class A drivers is demonstrated by the following simulated results. Figure 6 is a plot of the symbol error rate as measured from a fully NTSC corrected, television transmitter using cascaded Class AB stages transmitting 16 QAM digital data.

The figure shows data versus theoretical symbol error rates (SER) over a range of carrier to noise ratios (C/N). Note that at 1 x 10^{-3} errors, there is a 7 db difference between theoretical and measured carrier to noise ratios necessary in order to achieve the error rate. This data indicates that the cascaded Class AB NTSC transmitter would require 7 db more output power to meet the SER of 1 x 10^{-3}. The error rate of 1 x 10^{-3} was chosen since it is generally recognized as the minimum acceptable error rate to maintain stable picture reception. Higher error rates will result in rapid loss of the picture.

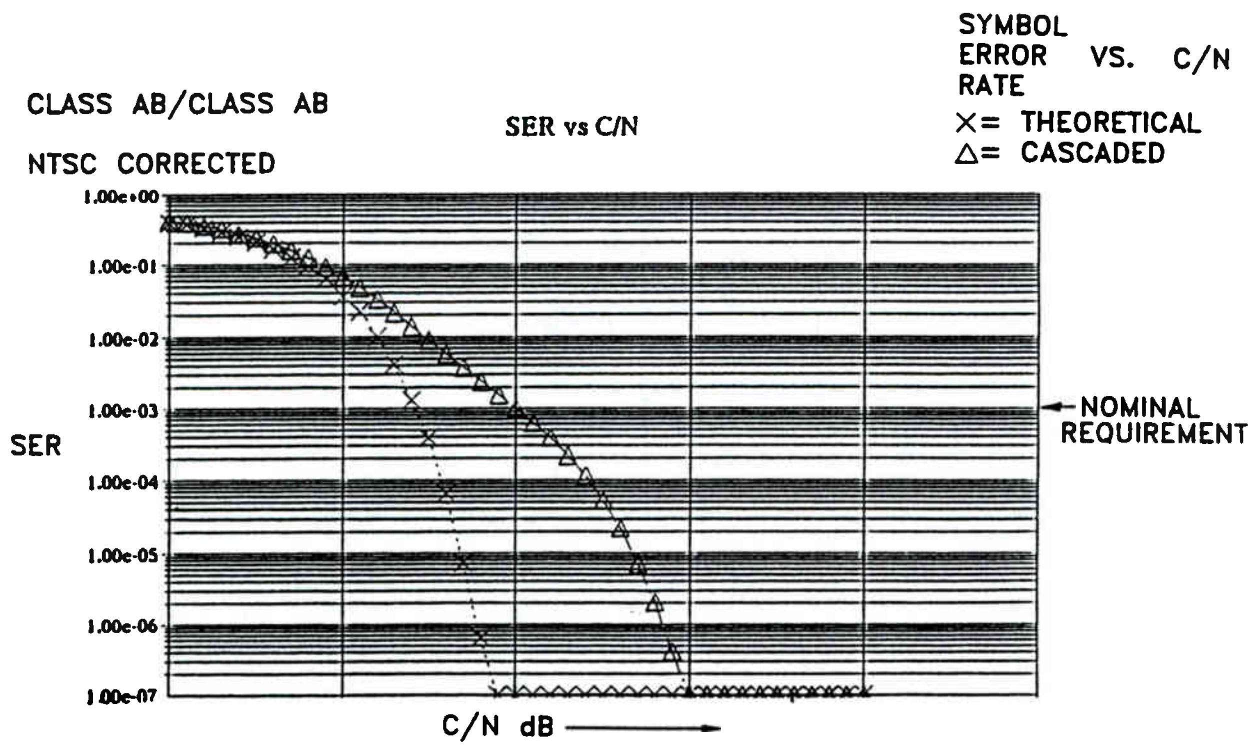

Figure 6 - Symbol Error Rates (SER) from Transmitter with Cascaded Class AB Power Stages (Correction Optimized for NTSC)

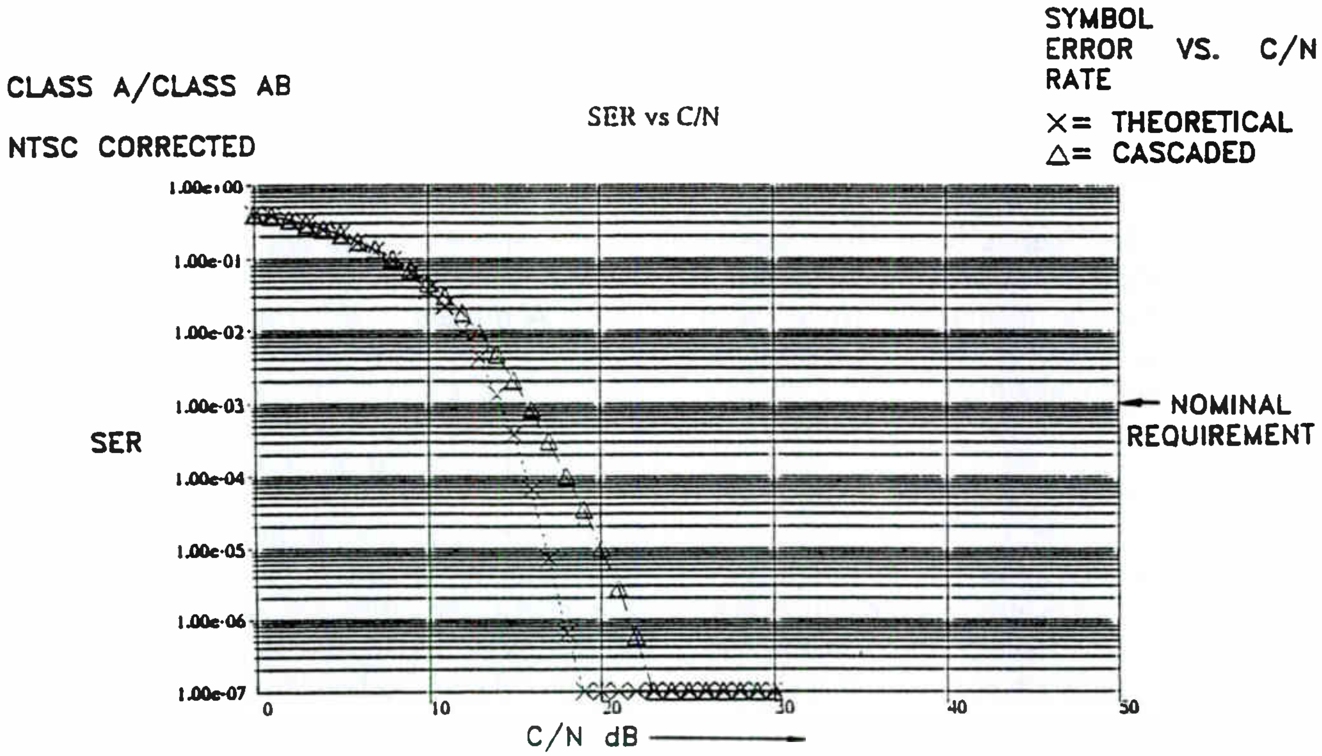

Figure 7 - Symbol Error Rates (SER) from Transmitter with Class A Driver (Correction Optimized for NTSC)

Figure 7 illustrates the effect of Class A drivers in an NTSC corrected IOT transmitter. At the SER of 1×10^{-3}, the deviation from theoretical is approximately 2 db.

Thus, the use of Class A drivers reduced the intermodulation generated in cascaded Class AB drivers and improves the C/N required for a SER of 1×10^{-3} by 5 db.

A 5 db difference means that a Class A driven IOT transmitter, transmitting 16 QAM, could have 1.78 times the range of a Class AB driven transmitter for the same output power. In other words, a Class A driven IOT could achieve the same range as a Class AB driven transmitter with one-third (1/3) the output power of the Class AB driven system.

This data was simulated using 16 QAM. Using 32 QAM we would expect the differential to become even greater.

While this data is only illustrative of the principle involved, a 5 db improvement in C/N by using Class A drive systems for digital transmission was significant reason to choose Class A drivers for the "IOX" series transmitter line.

Thus, the "IOX" series transmitter meets its design objective of providing optimized digital HDTV performance while also providing uncompromised conventional television transmission.

V. Other Considerations

Another consideration for future ATV systems revolves around the possible need for output filter shaping. Because the IOT output is impedance dependent, any filtering should be accomplished by the incorporation of "Balanced" or "Constant Impedance" devices.

Not only do these provide good out-of-band performance, but may as well provide for future requirements of channel combining equipment. Figure 8 is a diagrammatic representation of one "Balanced" filter approach.

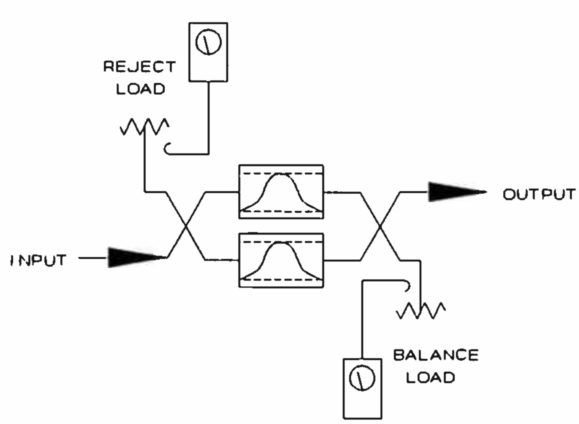

Figure 8 - "Balanced" Filter Diagram

VI. Conclusions

This paper discussed the design considerations for choosing the systems used in **COMARK**'s new "IOX" series of transmitters.

Presentations were provided showing why the IOX transmitters use IOT devices and Class A drivers in common amplification. It was shown how this choice allows for high levels of system flexibility without compromising conventional TV transmission performance.

This paper also presented data indicating that Class A driven systems provide significant advantages over Class AB driven systems in the areas of linearity correction and digital transmission fidelity.

Data was presented comparing the Class AB driven system to a Class A driven system. This data showed a significant advantage was realized by the superior linearity of the Class A driven system.

In conclusion, the design choices made during the development of the "IOX" series and presented in this paper have resulted in an improved **DUAL USE™** transmitter providing superior NTSC performance today and ready for D-HDTV tomorrow.

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PRE-CORRECTION TECHNIQUES USED IN THE DIGITAL TRANSMITTER

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ABSTRACT

This paper describes a modulator and the digital processor used to correct the amplifier's non-linear behavior. In this way, the AM-AM and AM-PM typical distortions of the QAM transmitters can be pre corrected. This paper also describes the pre distortion techniques used for HDTV power amplifiers, as well as an analysis of the results. Comparison is made between the effectiveness of non-linear correction at the Digital level or the IR level.

32 QAM VS NTSC COVERAGE

By using the data of the coverage area analysis in the Digicipher™[1] HDTV System Description, submitted by General Instruments to the ATTC on August 22, 1991, the ratio for 56 miles area QAM is:

$$10 \log \left(\frac{\text{Peak Sync Power}_{NTSC}}{\text{Avg. Power}_{32\,QAM}} \right) = 18.3 \text{ dB}$$

Digital transmission presents a Peak/Average power that depends on the type of modulation and base band filter characteristics. For the case of 32 QAM and a raised cosine filter (alpha = 0.1) the peak to average ratio is 7.7 dB.

The distribution density of the instantaneous Peak power levels is indicated in Figure 1. With the fives peaks corresponding with the constellation points of a 32 QAM modulation, see Figure 2.

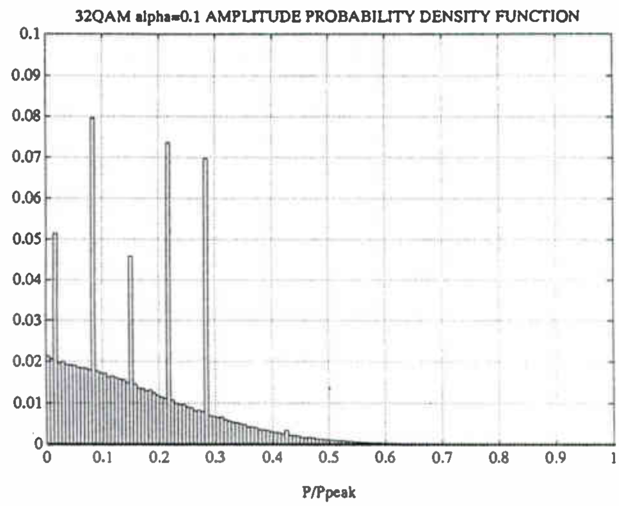

Figure 1. 32 QAM alpha=0.1 Amplitude Probability Density Function

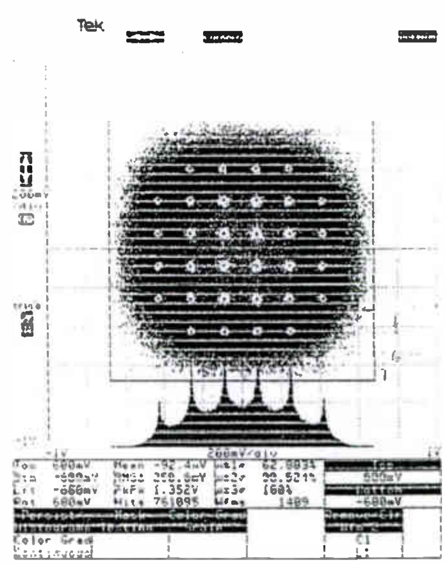

Figure 2. Five Peaks Corresponding with the Constellation Points of a 32 QAM Modulation

In the histogram of Figure 1, it can be appreciated that the probability of peaks exceeding in more than 5dB, the average power is less than 1%. There are some proposals about clipping these peaks, so that a 2dB can be gained in the Peak to Average ratio, but considering the 1dB lost in the C/N ratio needed in the receiver, then only 1dB is the final improvement, with remarkable inconveniences of pre correction capabilities due to the side bands generation of the clipping action.

With the previous considerations and not including other factors, like vertical polarization of the antenna, better noise figure of the receiver's front end, etc., in favor of the digital transmission, the 32 QAM Tx Peak Power vs NTSC Peak Sync Power is -10dB, as shown in Figure 3.

For example, the area now covered by a 50kW P.E.P. NTSC transmitter can be covered by a 5kW P.E.P. 32 QAM transmitter, working at less than 1kW average power.

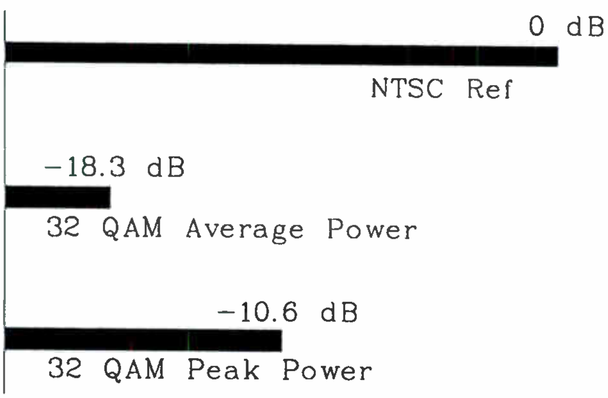

Figure 3. 32 QAM Transmitter's Peak Power Vs NTSC Peak Sync Power

SOLID STATE AMPLIFIERS

For Building a Solid State Transmitter with 5kW peak power output stage the adequate type of amplifier is AB class. This type of configuration has more non linearities for low level signals than the A class and are very difficult to linearize at IF:

The non linearities of the power amplifier produce two undesired effects:

Out of Channel Radiation

In analog NTSC transmitters, the out of band intermodulation products of the different vision color and sound carriers appear at a few frequency locations and can be eliminated using a combination of traps and band pass filters.

The 32 QAM signal after the band shaping has a flat spectrum similar to a band limited gaussian noise. Thus, the non linear behavior of the amplifiers creates out of band intermodulation products with a continuous spectrum and can not be eliminated using traps.

In a digital 32 QAM transmitter, the allowed level of out of band radiation must be expressed in terms of spectral power density.

Distortion of the Transmitted Signal Increasing the Carrier to Noise Ratio (C/N) Necessary to Achieve the Desired Error Probability at the Receiver

The effect of the compression of the amplifier is a position shift of the transmitted symbols in the constellation diagram decreasing the free distance between adjacent points and thus increasing the error probability, as shown in Figures 4 and 5.

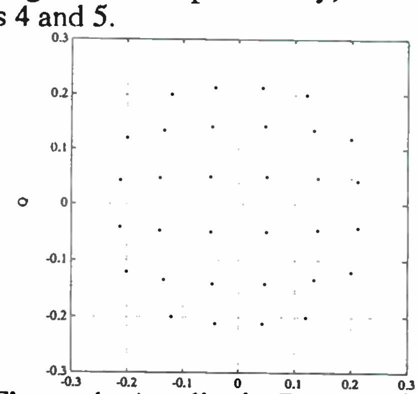

Figure 4. Amplitude Compression

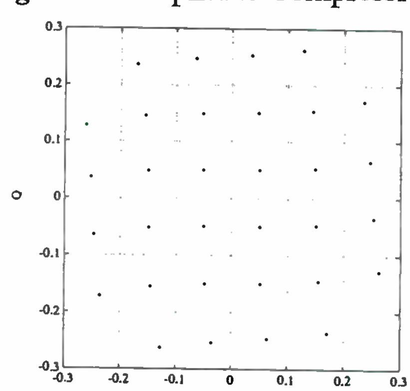

Figure 5. Phase Compression

The solution to this problem is to use digital pre corrections before I/Q modulators. In this way, amplitude and phase non linearities of the amplifiers (AM-AM AM-PM conversion) are pre corrected.

This base band pre correction takes place in the driver unit of the transmitter.

Because the average signal power is constant in a 32 QAM transmitter, there is nothing similar to the APL dependent distortion of the NTSC transmitters.

The block diagram of a 32 QAM transmitter with this kind of digital pre corrections is shown in Figures 6 and 7.

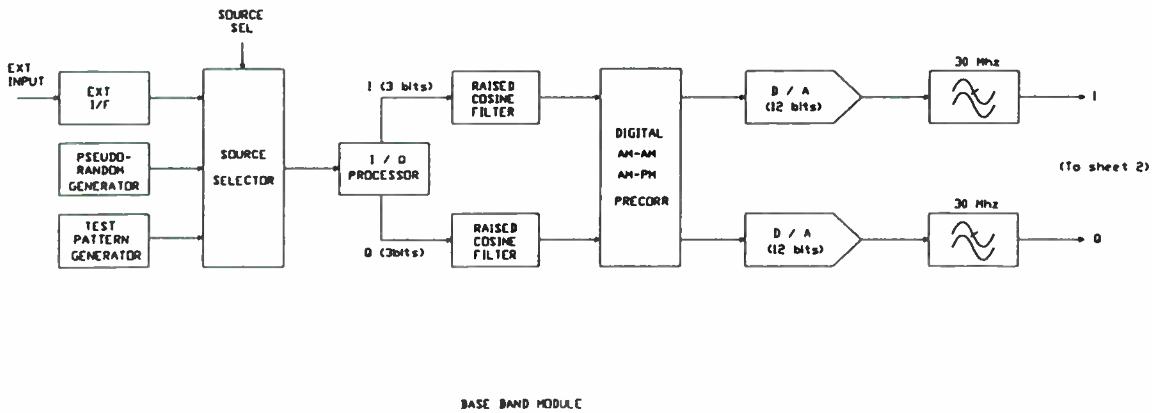

Figure 6. 32 QAM Experimental TV Transmitter Block Diagram

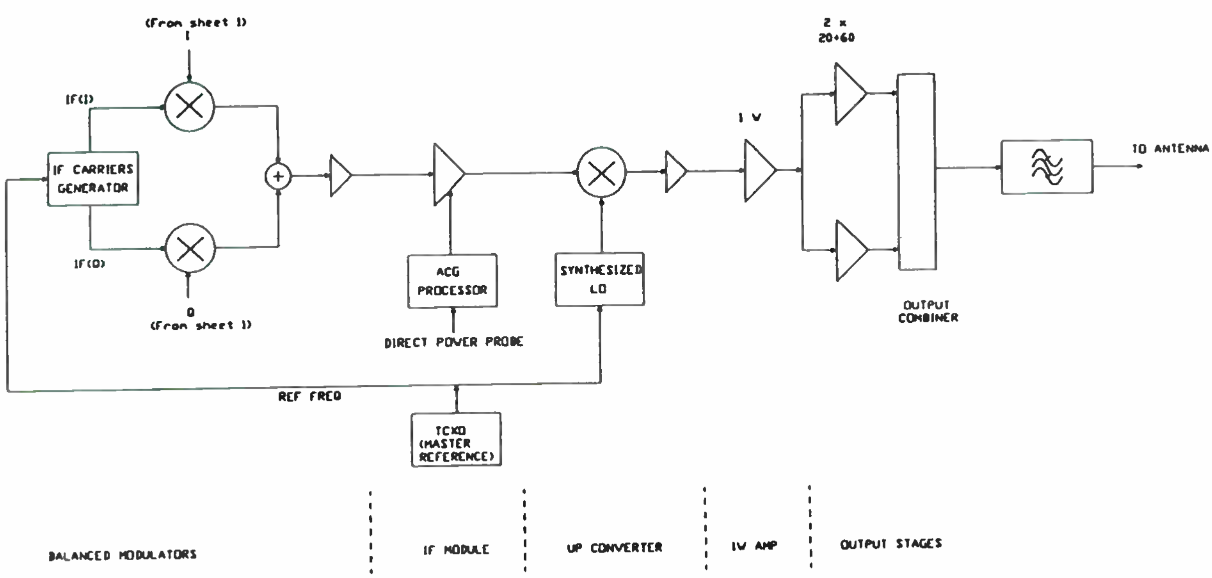

Figure 7. 32 QAM Experimental TV Transmitter Block Diagram

DESCRIPTION OF THE DIGITAL AM-AM AM-PM CORRECTIONS

The outputs of the raised cosine digital filters are the I and Q signals once they have been sampled at four times the symbol frequency. Each pair of samples goes through a table in which another pair of output samples are assigned.

To make up this table, the transfer function of the power amplifier in phase and module has been characterized. In this way, each pair of I, Q values, that is to say, each carrier vector, will have its values modified depending on its module.

To characterize the power amplifier, power ramps with the adequate I, Q amplitude and duty cycle are used. These ramps make the envelope of the output signal vary linearly in time.

The transmitted I, Q can be retrieved by demodulating the output signal; in this way we can calculate the AM-AM AM-PM distortions that must be corrected.

As an example of the transmitter's non linearity characterization procedure, these are the data obtained from the study of a 28 Watts peak power amplifier.

Figure 8 shows the envelope of the transmitters output signal when the input signal is a carrier which amplitude is modulated by a ramp. The AM-AM compression of the transmitter can be also measured directly on this curve.

To obtain this graph and the next one, a sampling oscilloscope has been employed.

Figure 9 shows the characterization of the AM-PM conversion. The oscilloscope's screen shows four traces; the first one (starting from left) represents the envelope of the output signal once the amplitude compression has been corrected. The traces labeled I, Q are the components in quadrature, retrieved by means of a QAM demodulator specially made for this purpose. From these signal values, we can directly determine the transmitter's phase characteristic according to the instantaneous amplitude of the carrier (AM-PM distortion).

The last trace is the composition in X-Y mode of the I-Q detected signals. The AM-PM conversion can be distinguished clearly as the deviation of a straight line.

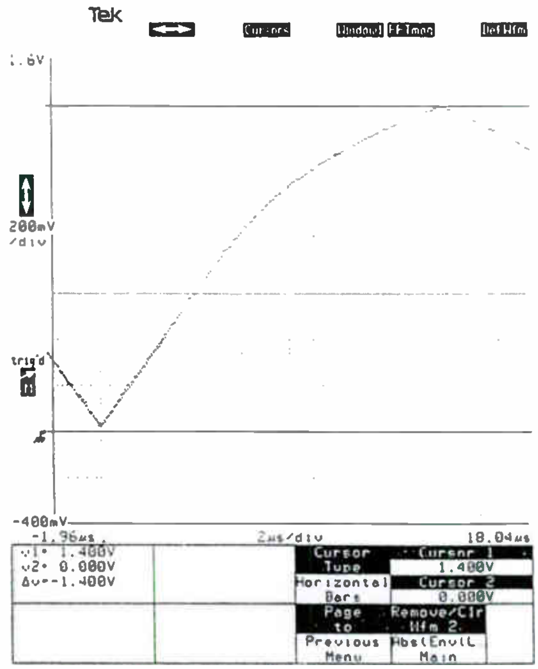

Figure 8. The Transmitter's Output Signal When the Input Signal is a Carrier which Amplitude is Modulated by a Ramp

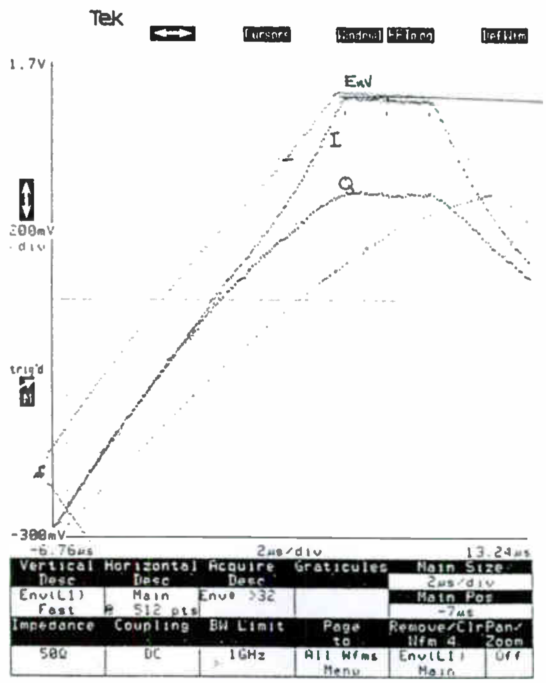

Figure 9. Characterization of the AM-PM Conversion

Figure 10 represents the curve obtained from the AM-AM distortion of the 28 W amplifier and the non linear responses that are applied to baseband signal in order to correct it.

Figure 11 represents the same as the previous one but applied to the AM-PM distortion.

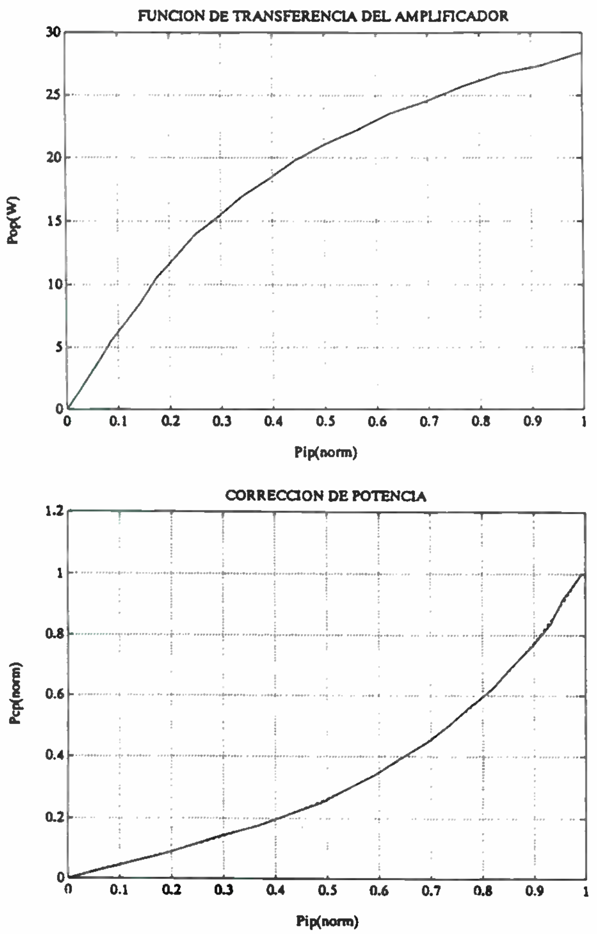

Figure 10. AM-AM distortion

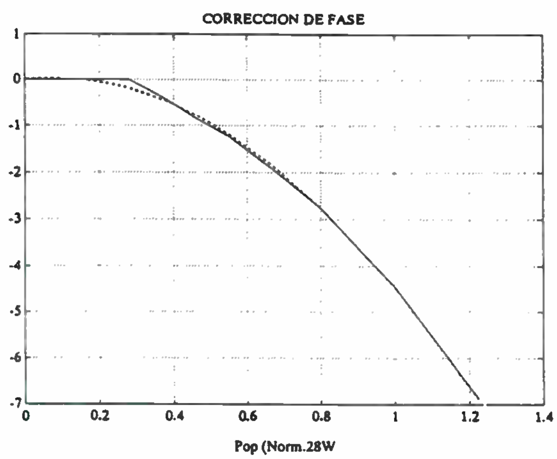

Figure 11. AM-PM distortion

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DIGITAL MODULATION FOR TELEVISION BROADCASTING

Monday, March 21, 1994

Moderator:
Louis Libin, National Broadcasting Company, New York, NY

STUDY ON ADVANCED DIGITAL MODULATION METHOD: OFDM AND AW-CDM
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***EUROBITS—WHITHER DIGITAL EUROPE?**
George Waters
European Broadcasting Union
Geneva, Switzerland

TECHNICAL AND QUALITY ISSUES FOR EUROPEAN DIGITAL TELEVISION
David Wood
European Broadcasting Union
Geneva, Switzerland

EVALUATION OF MULTI-CARRIER TRANSMISSION SYSTEMS FOR ADVANCED TELEVISION IN NORTH AMERICA
Yiyan Wu
Communications Research Centre
Ottawa, Ontario, Canada

***THE GRAND ALLIANCE TRANSMISSION SYSTEM**
Presenter to be Determined

*Papers not available at the time of publication

STUDY ON ADVANCED DIGITAL MODULATION METHOD: OFDM AND AW-CDM

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ABSTRACT

This paper describes NHK's most recent research and development on two advanced digital modulation methods. One is Orthogonal Frequency Division Multiplexing (OFDM) to overcome multipath propagation and the other is Adaptive Weighted Code Division Multiplexing (AW–CDM) to make graceful degradation feasible. Through simulations and experiments using prototype models, some of their features have been investigated.

1. INTRODUCTION

OFDM is a useful digital modulation method for terrestrial digital broadcasting systems, both for digital TV broadcasting and digital sound broadcasting. The OFDM signal consists of a large number of digitally modulated carriers, which are orthogonal to each other, within a certain length of the signal period. OFDM shows excellent performance, especially in multipath environments, and has many other merits, namely, comparatively high spectrum efficiency, ruggedness against interference signals, applicability for single frequency networks (SFN), and suitability for digital signal processing. OFDM has been studied mainly in the telecommunications field[1][2], and has been adopted as a modulation technique for the digital audio broadcasting (DAB) proposed by EBU[3][4].

NHK is committed to research and development of OFDM, considering it to be an important technique for terrestrial digital broadcasting systems such as DAB, and especially for mobile reception.

Of particular interest now is how to ensure graceful degradation, which improves the breakdown characteristics of digital transmission at its threshold. One way considered to achieve graceful degradation is to first encode video and audio signal into hierarchical data, then weight each to give it a different priority, and, lastly, to multiplex them for transmission. OFDM can multiplex weighted hierarchical data employing different modulation, e.g., BPSK, QPSK, 16QAM or 32QAM, for different carriers.

Another attractive possibility being considered for hierarchical transmission is AW–CDM, which employs a spread–spectrum technique. Simulation and development of an adaptive equalizer is in progress because AW–CDM requires one for terrestrial use, which involves multipath propagation.

2. OFDM

OFDM is a kind of multi–carrier modulation. The most common modulation method for each carrier is QPSK, but multi–level modulations such as 16QAM or 32QAM can also be used.

Figure 1 shows an OFDM transmission symbol, which consists of an effective symbol period and a guard interval. The guard interval is a signal period used to reduce the influence of multipath signals. The signal waveform of the guard interval is a cyclic repetition of the effective symbol period's.

The shape of the power spectral density of OFDM is almost rectangular, as shown in Figure 2.

A prototype DQPSK–OFDM modem (prototype model I) has been developed for experiments on digital sound broadcasting and field experiments have been carried out. A multi–level OFDM modem for digital TV broadcasting (prototype model II) is now being developed.

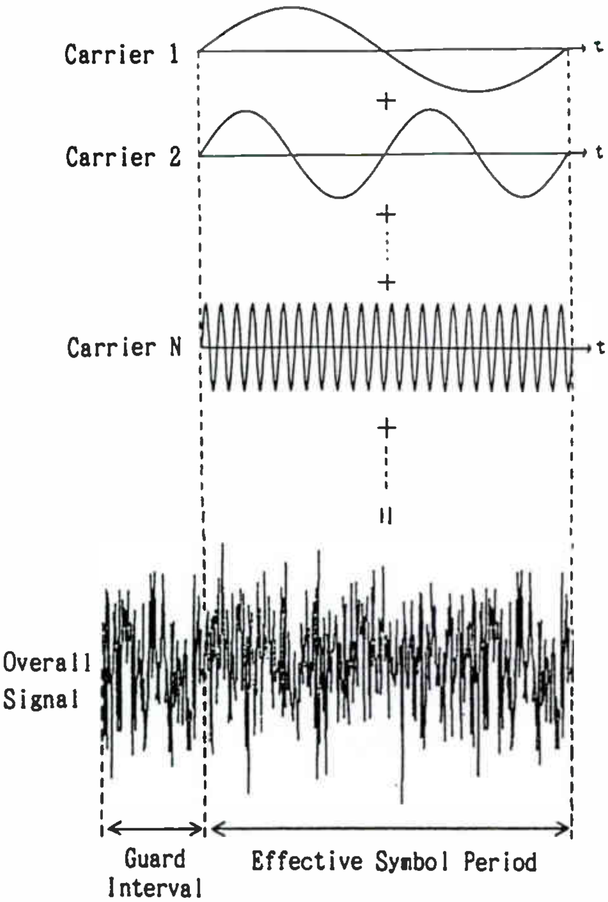

Figure 1 Baseband Signal Waveform of OFDM

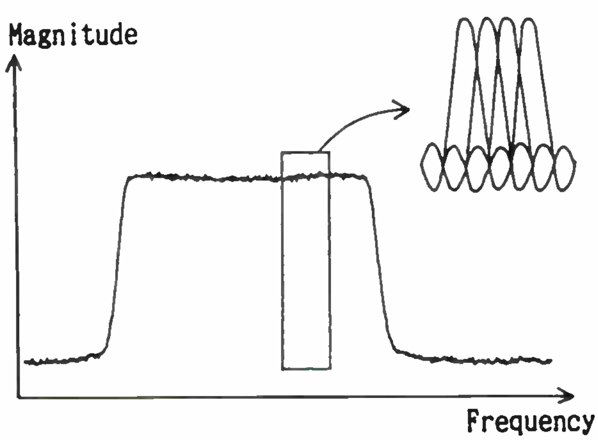

Figure 2 Spectrum of OFDM

2-1. FEATURES OF OFDM

a. The performance degradation due to multipath signals is much less than that of single-carrier modulation systems because of the long symbol length and the addition of the guard interval.

b. The maximum bit rate attainable within a given frequency bandwidth is rather close to the Nyquist rate, therefore the spectral efficiency is comparatively good.

c. The interference from OFDM to other services is small because the signal waveform resembles white Gaussian noise, as shown in Figure 1.

d. Modulation and demodulation by FFT (Fast Fourier Transform) are possible.

e. Interleaving in the frequency domain is easy, and a system that is very robust against selective fading can be constructed by the combination of OFDM and error-correcting codes.

f. The side-lobe level of the power spectral density curve diminishes sharply and the power leakage into adjacent channels is small.

g. Hierarchical transmission or graceful degradation can easily be incorporated.

h. All the frequency intervals between adjacent carriers are the same. Thus, if the transmission channel has non-linear characteristics, the OFDM signal is subject to performance degradation due to intermodulation. Therefore, a power amplifier for OFDM signals must be used within its sufficiently linear range.

2-2. PROTOTYPE MODEL I

To verify the fundamental transmission characteristics of OFDM, a DQPSK-OFDM modem (prototype model I) using the parameters shown in Table 1 has been developed. This equipment permits transmission tests for both digital sound broadcasting and terrestrial digital TV broadcasting. This section describes some results of indoor experiments.

Table 1 Parameters of Prototype Model I

frequency bandwidth: 3.5 MHz
effective symbol length: 128 μs
guard interval length: 32 μs
number of carriers: 448
modulation for each carrier: DQPSK
frame length: 48 ms
gross bit rate: 5.544 Mbit/s

Figure 3 shows the bit error rate (BER) performance when both Gaussian noise and a single ghost signal (multipath signal) whose delay time is 2 µs and DU ratio is 8 dB are added to the desired signal simultaneously. The bold lines show the results of indoor experiments and the dotted lines show the results of computer simulations. No equalization is introduced in either the experiments or the simulations. If an implementation margin of about 1 dB is allowed, the experimental results are in accord with the simulation results. When ghost signals exist, the BER performance of a single-carrier modulation such as QPSK degrades badly without equalization. However the degradation of OFDM due to ghost signals is much smaller than single carrier systems. The superiority of OFDM over a conventional single carrier approach in a multipath environment is evident from this figure.

Adding a single ghost with constant DU ratio to the desired signal with the ghost delay time varied yields the BER curves shown in Figure 4. Within the range of ghost delay time where the ghost signal of the adjacent symbol does not come into the FFT window of the demodulator (below 28 µs in Figure 4), the degradation of BER performance is not serious because there is no inter-symbol interference. If the ghost delay time exceeds the maximum value within the range, the BER increases.

Figure 5 shows the BER performance in the case where the channel characteristics contain non-linearity. The horizontal axis indicates the 3rd-order intermodulation level (IM3) of the amplifier used in the indoor experiment. The amplifier is a general-purpose solid-state amplifier whose gain is about 25 dB. The IM3 values were measured by 2-tone method for different input levels to the amplifier, where the frequency difference between the two tone signals was identical to the frequency

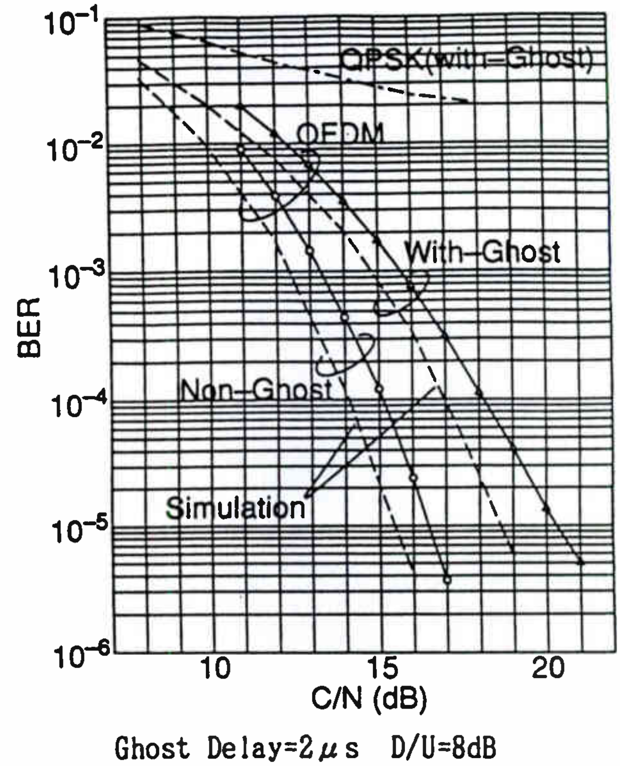

Figure 3 BER Performance under Single Ghost and Gaussian Noise (1)

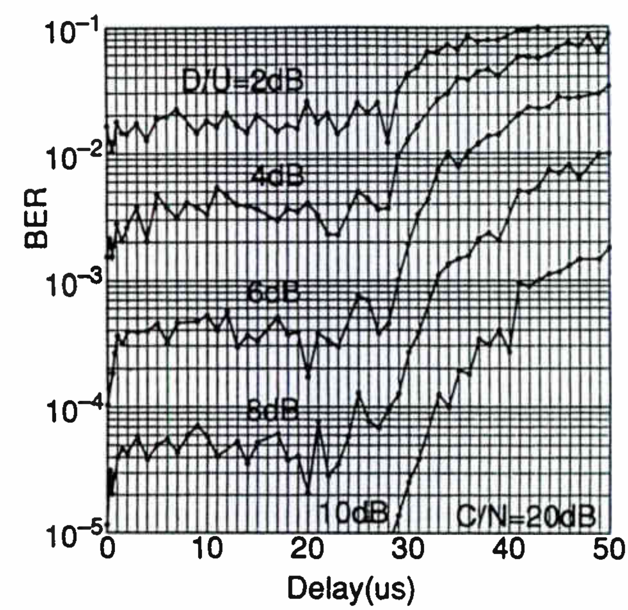

Figure 4 BER Performance under Single Ghost and Gaussian Noise (2)

interval between the OFDM carriers. The BERs in Figure 5 were measured for different input levels of OFDM to the amplifier. From Figure 5, it can be seen that, in the case of this general-purpose amplifier, the BER is always below 10^{-3} if the CN ratio is above 15 dB and the IM3 is below –20 dB. The BER value of 10^{-3} can be reduced to a negligible value by using an appropriate error-correcting code with a coding rate of 70% to 80%. Therefore, in terrestrial digital broadcasting systems where sufficiently large power amplifiers can easily be employed depending on the requirements, the influence of non-linearity on the performance of OFDM is not considered to be a significant problem.

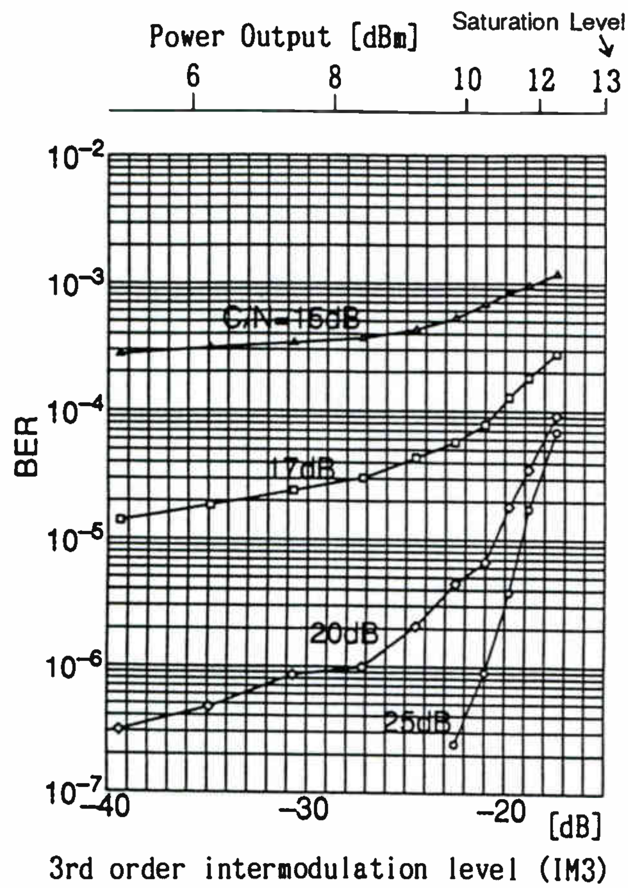

Figure 5 BER Performance in a Non-Linear Channel

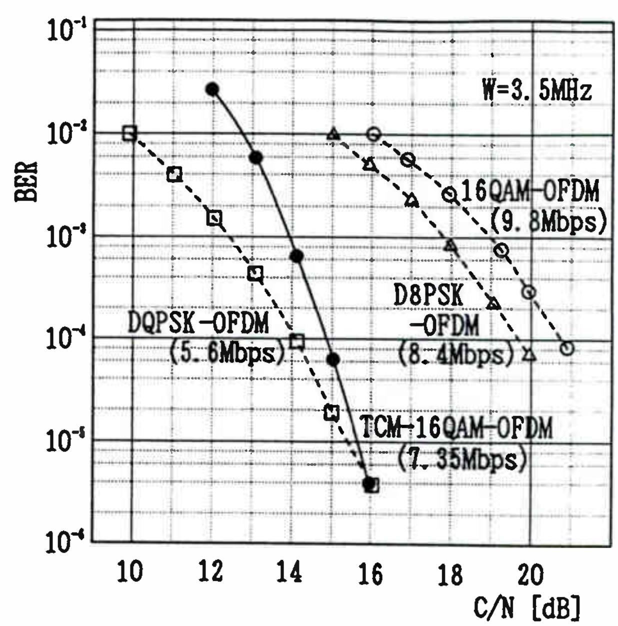

Figure 6 BER Performance in a Gaussian Channel and Effective Bit Rates of the OFDM Systems

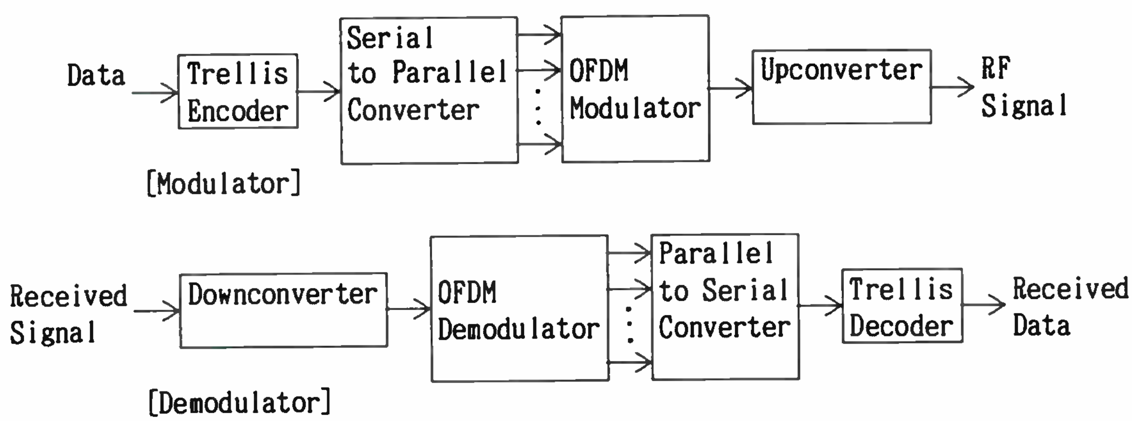

Figure 7 Conceptual Configuration of the Trellis Coded OFDM Modulator and Demodulator

2-3. MULTI-LEVEL OFDM AND PROTOTYPE MODEL II

To apply OFDM to digital television broadcasting, it is necessary to increase the bit rate in the same bandwidth, and multi-level OFDM such as 16QAM-OFDM is suitable for this. Figure 6 shows results for computer simulation of BER performance in a Gaussian channel and the effective bit rates of multi-level OFDM systems. Here, DQPSK-OFDM, D8PSK-OFDM and 16QAM-OFDM stand for systems in which each carrier is modulated with DQPSK, differentially modulated 8-phase PSK and 16QAM, respectively, and TCM-16QAM-OFDM is a combination of Trellis Coded Modulation and 16QAM-OFDM[5]. The configuration of the trellis coded OFDM modem is given in Figure 7. In 16QAM-OFDM and TCM-16QAM-OFDM, the data that provide phase and amplitude references for each carrier are sent periodically from the transmitter to the receiver, and the demodulation is carried out based on the reference data. The OFDM parameters are the same as in Table 1 except for the bit rate and the modulation method for each carrier. TCM-16QAM-OFDM can transmit about 1.3 times as many data bits as DQPSK-OFDM and shows better BER performance than DQPSK-OFDM if the CN ratio is above 16 dB.

Production of a multi-level OFDM modem (prototype model II) is now underway, using the parameters shown in Table 2. Both indoor and field experiments are planned for 1994.

Table 2 Parameters of Prototype Model II

frequency bandwidth: 6 MHz
effective symbol length: 100 μs
guard interval length: 12.5 μs
number of carriers (nominal): 600
modulation for each carrier: 16QAM or 32QAM
frame length: 45 ms
gross bit rate: 16.64~24.51 Mbit/s

3. AW-CDM

Portable receivers such as "pocket television" require special consideration in terms of digital broadcasting, because they operate at low CN ratios due to limited aerial and receiving capabilities. Hierarchical transmission using a spread-spectrum is one answer for portable reception. Spread-spectrum AW-CDM enables an extremely wide range of hierarchical transmission in a restricted power because of the homogeneous spectrum it maintains in spite of weighting. AW-CDM may fill the digital transmission gap without using a "gap filler".

AW-CDM is an outgrowth of the code division multiple access (CDMA) digital cellular system[6] developed by QUALCOMM, Inc. Research of it was started at the end of 1991, when, coincidentally, Prof. W. F. Schreiber presented the idea[7] of introducing spread-spectrum methods into digital broadcasting at SMPTE technical conference in Los Angeles. While no concrete example of a spread-spectrum digital broadcasting system has been shown, we have successfully demonstrated his idea using experimental AW-CDM equipment based on CDMA.

The AW-CDM experimental equipment is a 64-channel multiplexing system which uses 64 binary orthogonal codes based on Walsh sequences W_0-W_{63}, each of which is one row of a Hadamard matrix, as "carriers". Figure 8 shows the transmission symbols and spectrum of AW-CDM. The 64 modulated and weighted sequences, W_0'-W_{63}', are combined and scrambled by a pseudo-noise (PN) sequence, so the signal of a channel k is $W_k' \times PN$. The spectrum of each channel is homogeneously spread over the full bandwidth.

Although hierarchical transmission is feasible in AW-CDM by only weighting the magnitude of each code, an adaptive equalizer is required for higher performance.

3-1. FEATURES OF AW-CDM

a. Wide-range and/or quasi-continuous, graceful degradation

Different subchannels can be given arbitrarily different weights. In spite of subchannel weighting, the spectrum remains homogeneous because the spectrum of every subchannel homogeneously spreads over the full bandwidth. This means that extreme

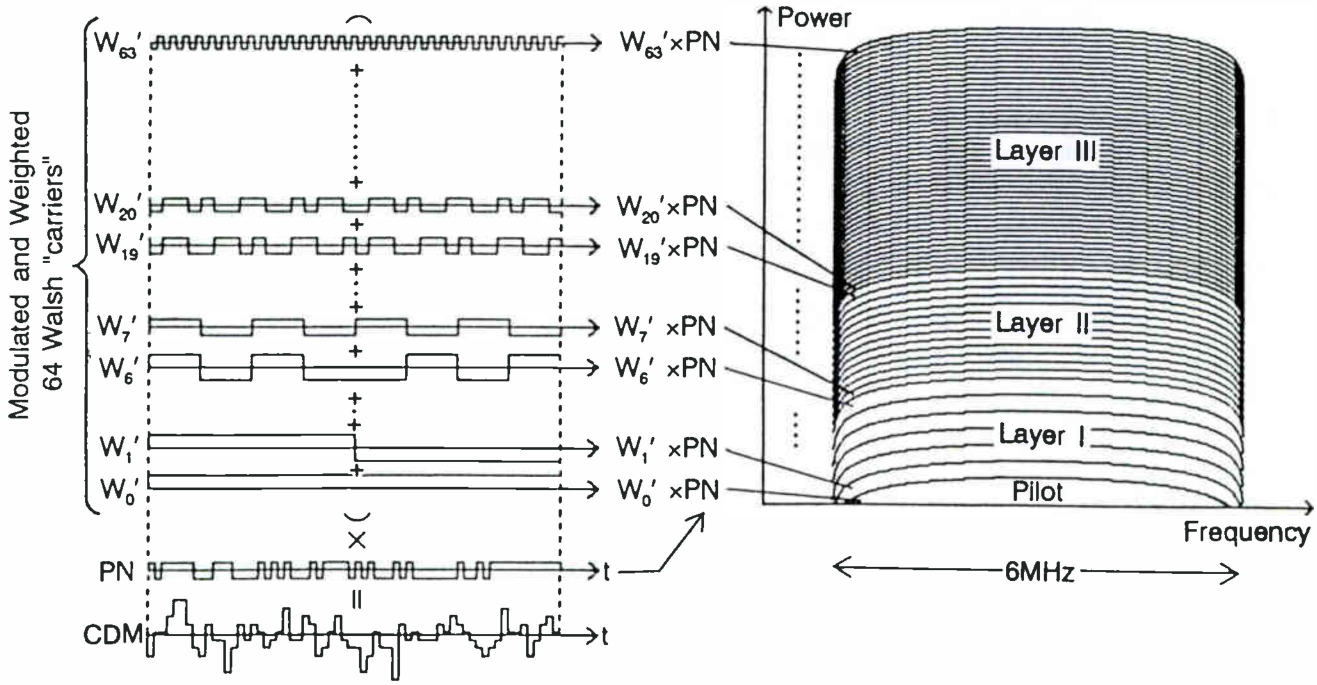

Figure 8 Transmission Symbols and Spectrum of AW–CDM

weighting (wide–range graceful degradation) is feasible within a restricted power. Also, different spatial frequencies can be given an optimum distribution and image quality can be varied directly with the CN ratio (quasi–continuous graceful degradation).

b. Variable data rate

Even if a number of subchannels are not used, there is no problem. This means that data of variable rate can be transmitted directly, without inserting dummy data. This is a favorable characteristic for transmitting digital data of moving pictures.

c. Coherent detection

AW–CDM uses a PN sequence as a reference "pilot" signal. The PN sequence is orthogonally multiplexed together with other subchannels into the full bandwidth. In the receiver, the impulse response of the transmission channel is periodically obtained through correlation with the reference PN sequence. Synchronous pulses are directly obtained from this periodically appearing impulse response. Synchronous pulses are rapid enough to regenerate the carrier even for mobile reception. Coherent detection has a 3 dB advantage in CN ratio over delay detection, which is widely employed in mobile reception.

d. Adaptive equalization

The transmission channel impulse response is obtained periodically through the correlator in the receiver. For adaptive equalization, all that is necessary is to calculate tap coefficients and to load them to the transversal filter.

e. Feasibility of data rate increasing

Higher data rate transmission requires nothing more than simply changing the input signal binary to multiple. There is no need to modify the coherent detector and adaptive equalizer. Increasing the data rate enables AW–CDM to transmit HDTV in the 6 MHz band.

3–2. SCHEME OF EXPERIMENTAL EQUIPMENT

The experimental AW–CDM equipment is shown schematically in Figure 9. In the transmitter, video and audio signals are hierarchically coded into 3 layers and these layers are multiplexed into a pair of 64 bit/frame time division multiplex (TDM) data streams. Each layer is weighted to have a different priority. Although hierarchical transmission is possible using this TDM stage, it is very difficult to hierarchically detect in mobile channels.

Each bit of the data streams is orthogonally spread

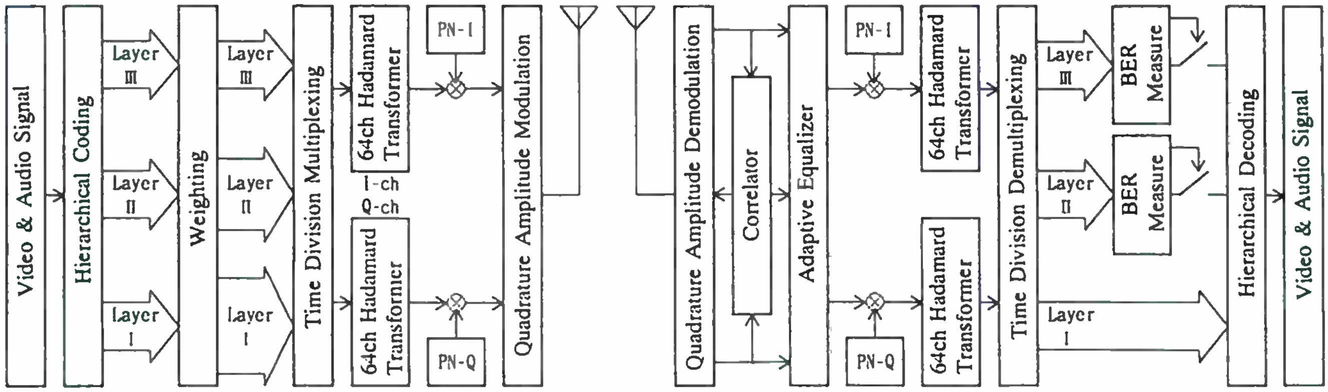

Figure 9 Experimental AW-CDM Equipment

into the full channel bandwidth (6 MHz) and summed into a pair of data streams through a pair of 64ch Hadamard transformers and PN scramblers. These streams are filtered and quadrature-amplitude-modulated.

In the receiver, the received signal is coherently detected, filtered and adaptively equalized into a pair of data streams. These streams are demodulated into a pair of 64 bit/frame TDM data streams through a pair of PN descramblers and 64ch Hadamard transformers. The quality of each demultiplexed layer is evaluated and chosen to be used or not, depending on its BER. Only retrievable data is decoded into video and audio signals. Figure 10 shows measured BER performance in a Gaussian channel with a single ghost. As in the figure, the required CN ratios of layers III, II and I for BER<10^{-5} without ghost (multipath) are 13.2 dB, 8.3 dB and 0 dB, respectively; and with a single ghost they can perform at BER<10^{-5} for ghosts of D/U=15 dB, 10 dB and 0 dB, respectively. Parameters of the experimental equipment are given in Table 3. The data rate and level weighting allocation to each layer can easily be changed. In the same bandwidth, various types of performance, e.g., wider-range or quasi-continuous graceful degradation, can be freely selected.

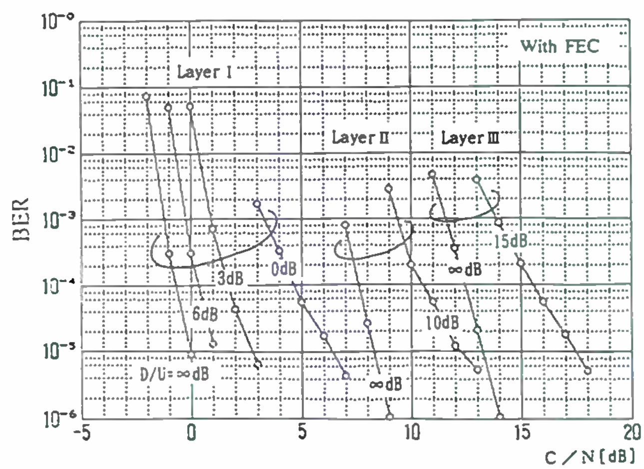

Figure 10 Measured BER Performance of AW-CDM in Gaussian Channel with Single Ghost

Table 3 Parameters of Experimental AW-CDM Equipment

BANDWIDTH	LAYER	DATA RATE EXCEPT FEC	FEC RATE	DATA RATE WITH FEC	BIT ASSIGN IN 1 FRAME	LEVEL WEIGHT	CNR FOR BER≦10^{-5}
6 MHz	I	384 kbps	1/2	768 kbps	5	+6.2 dB	≧ 0 dB
	II	1.3824 Mbps	3/4	1.8432 Mbps	12	0 dB	≧ 8.3 dB
	III	5.9136 Mbps	7/8	6.7584 Mbps	44	-3 dB	≧13.2 dB

3-3. ADAPTIVE EQUALIZER

The adaptive equalizer is illustrated schematically in Figure 11. For high-speed performance, a feed-forward structure is adopted. Impulse response of the transmission channel is automatically obtained from two correlators. Adaptive equalization is simply achieved by loading two pairs of 1024 tap finite impulse response (FIR) filters with the obtained impulse response data. From Figure 12, showing simulated BER performance of AW-CDM (layer II) before and after equalization, BER performance is expected to be drastically improved.

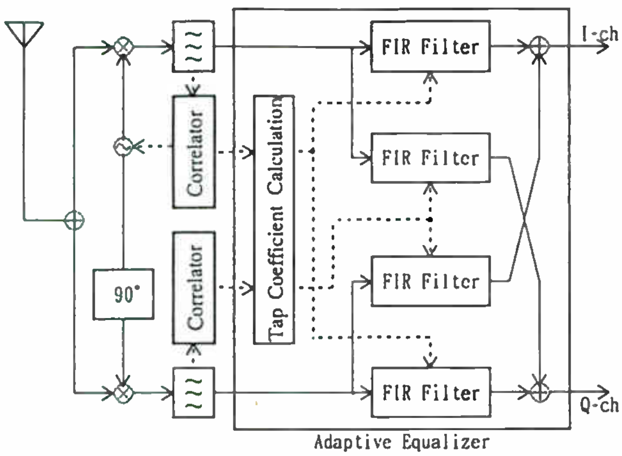

Figure 11 Scheme of Adaptive Equalizer for AW-CDM

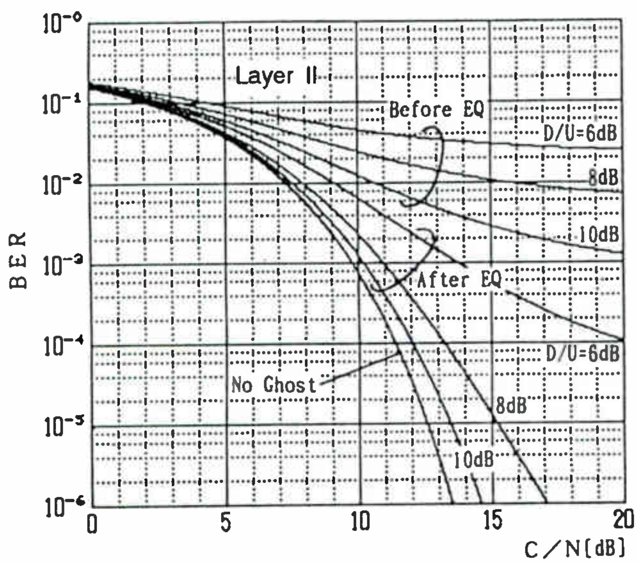

Figure 12 Simulated BER Performance of AW-CDM Employing Adaptive Equalizer (EQ)

4. CONCLUSION

We have outlined NHK's most recent research and development in OFDM and AW-CDM.

Realization of a digital broadcasting system should be the result of integrating various technologies. We aim to produce an optimal performance, based on the results of both laboratory and field transmission tests of these modulation methods.

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TECHNICAL AND QUALITY ISSUES FOR EUROPEAN DIGITAL TELEVISION

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ABSTRACT

The development of digital television broadcasting systems in Europe is proceeding rapidly and efficiently. A market-driven outlook is guiding the work, which is leading to a series of systems for different delivery media. The initial systems are likely to be multi-channel satellite and cable, followed later by terrestrial systems. The paper outlines the main elements of the satellite and cable system, which use MPEG2 image and sound coding, and suggests the future directions the studies will take in terrestrial systems. Some comments on the value of open architecture for digital receivers are made.

1. INTRODUCTION

In Europe, the development and standardization of digital television broadcasting is a major combined private and public sector initiative. Currently, over 110 European organizations have formed a voluntary grouping, to develop digital television broadcasting systems and standards. This is termed the DVB project (or group).

The DVB (Digital Video Broadcasting) project has organized itself into a committee structure which will allow it to achieve its objectives. The European Broadcasting Union provides a Project Office for the activity, which is financially supported by all signatory organizations.

There is great strength of opinion that digital television standards must be market-led. They must be developed, from the start, not just on the basis of what technologists think is possible, but also on the finance available for programme making, and what the customer is likely to be willing to pay for. The starting point for the European systems are thus Commercial Committees, who reach a common accord on what the systems need to do, to have the best chance of commercial success, and for what retail price region. This is then translated into practical terms by the technologists in technical committees.

Certainly not the only one to do so, the European Broadcasting Union sees its job in this major enterprise as encouraging directions which will result in the highest technical quality and lowest cost for the general public.

Thus, a particularly attractive feature of the plan is the coordinated development of digital television systems for different transports; satellite, terrestrial, CATV, SMATV. In future, also hopefully VOD systems could be included. The intention is to maximize commonality between all different transport media.

2. THE OVERALL STRATEGY

There are differences in the needs and constraints associated with different transport media. There are different bandwidths, different error environments, and different power limitations.

Satellite transponders have relatively wide bandwidths, low available power, and few multipath problems. RF systems with constant or near constant amplitude are needed.

Terrestrial channels are narrow bandwidth, higher power, and susceptible to multipath. Cable channels have traditionally the same bandwidths as are used for terrestrial broadcasting, but there are differences in the environment. Within the world of cable, large scale installations (CATV) have different constraints to localized systems (SMATV). For example, for SMATV systems, costs needs to be kept very low.

The basic means by which bridges will be built between all types of delivery media is to use, as far as possible, a common baseband coding and multiplexing system, together with application-specific channel coding and modulation, tailored to the particular transport environment.

3. THE MPEG2 SYSTEM

The basis for the vision and sound coding coding will be the MPEG2 system. The MPEG2 system begins with a series of compression tools. The tools are combined or concatenated to form systems. The MPEG2 vision coding is actually a family of systems, which each use some or all of the tools, on some or all of a series of source formats.

The tools are techniqúes for motion-adaptive hybrid DCT coding. The source formats are for conventional and high definition scanning algorithms.

Some key elements of the MPEG2 matrix are shown in Table 1. The y-axis gives the four Levels which are associated with different scanning formats. There are two HDTV levels, High-1440 (using up to 1440 samples per line), and High (using up to 1920 samples per line). The number of active lines can be up to 1152. There is a Conventional Scanning level, also capable of wide aspect ratio incidentally, which can thus provide EDTV and SDTV. Finally there is a quarter picture level for LDTV, Limited Definition Television.

There is not a simple relationship between the Levels and final picture quality, because the compression system is allowed to operate over a range of bit rates at each Level, and different sets of compression tools can be used along the x-axis. The systems perform differently at different bit rates, and with different sets of tools. As a generality, as the bit-rate is reduced, the proportion of scenes which will contain unwanted impairments increases. For many of the tools, as the number of tools increases, the quality possible, at a given bit rate, increases. For other tools however, an overhead is needed, which can reduce the quality at a given bit rate.

Thus the term "Level" gives a general guideline on quality, but does not define if precisely. The quality is set by a combination of the source format, the tools used, and the operating bit rate.

There are five profiles: Simple, Main, SNR scalable, Spatial Scalable, and High. In each case, the decoder makes use of a progressively greater number of tools. This means, in each case, that the decoder needs to be more sophisticated, and thus may cost more. In each case, this has to be traded off against the features that the tools achieve.

There is an arranged degree of compatibility in the matrix. A given decoder must be compatible (able to decode signals) with levels and profiles to its left and below it.

There is a ceiling on the storage requirements associated with each Profile/Level combination. The bit-rates quoted in Table 1 are the ceilings on operating bit-rates for the Level/Profile combination concerned. The boxes marked with an x are not foreseen for use.

4. THE SATELLITE SYSTEM

The most likely candidate for vision coding for the first digital multi-channel satellite services in Europe is the Main Level/Main Profile combination. This can provide conventional scanning pictures with an operating bit-rate between 3 and 15 Mbit/s. In practice, it will be unnecessary to operate the the system at its maximum bit rate of 15Mbit/s, as quality improvements will be largely imperceptible after about 10 Mbit/s.

In general, the level of artifacts which occur at about 5 Mbit/s will make the system about equivalent to the quality available with PAL and SECAM. Taking the bit-rate up to 9 or 10 Mbit/s will allow near 4:2:2 studio conventional quality. Either 4:3 or 16:9 aspect ratio is possible. One quite likely scenario is to use the satellite transponder to carry about six conventional quality channels.

The sound coding will be based on the MPEG audio system.

The system will use the MPEG-2 multiplex structure, as will all transports. The MPEG2 multiplex consists of concatenated fixed length 'Transport Stream' packets. Each packet is 188 bytes in length. The Packet has a four byte header, which describes its contents, etc., the first byte of which is a sync byte. This may be followed by an optional adaptation field, and the rest of the packet is payload.

The payload can include image signals, sound signals, or indeed any other kind of data signals, developed in response to future needs.

The channel coding and modulation system will be as follows.

* Single carrier Grey-coded QPSK modulation with a roll-off of 0.35.

* Punctured convolutional coding, basis 1/2, constraint length 7, offering an open choice of convolutional rates among 1/2, 2/3, 3/4, 5/6, or 7/8. The receiver finds the particular rate by trial and error.

* Convolutional interleaving (Forney approach) with depth I = 12. Sync bytes are routed via the 0 delay path.

* Shortened Reed Solomon code, with a total code word of RS(204,188) for a complete MPEG2 packet. The capacity of the code is T=8, which means that up to 8 bytes per MPEG2 packet can be corrected.

* Target error rate after FEC is 10(-10) to 10(-11). This corresponds to less than one error event per hour

* Sync word is inverted every eight Transport packets

Table 2 shows examples of bit rates versus transponder bandwidths. The satellite transponders will be able to carry the order of 30-40 Mbit/s depending on the choice of convolutional code protection used. The selection of parameters will be made depending on the service operators' aspirations for the dish size of the user, given the power level of the satellite. Generally, the European view is that a 60 cm dish is the maximum size acceptable, and smaller is preferable.

Beyond the immediate interest in digital multi-channel satellite broadcasting at conventional quality is a longer term interest in HDTV broadcasting by satellite.

A single transponder would have the capacity to broadcast a relatively high quality HDTV signal, for example a Main Level/High 1440 Profile signal operating at about 34 Mbit/s. However, more sophisticated solutions are also being investigated.

The HDSAT project is a pan-European collaborative project which is developing a very high quality HDTV digital broadcasting system, for use in the recently allocated 20 GHz band, and a corresponding cable transport system for these signals. The system will also be a candidate for use in the 11/12 GHz region. HDSAT is studying layered modulation systems. We hope compatibility will be found between the two modulation systems, with a common denominator of the DVB's QPSK layer as the lower HDSAT layer.

The HDSAT approach, using a layered modulation scheme will give the service a degree of graceful degradation, which would be particularly useful for broadcasting in the 20 GHz band.

The MPEG2 system specification does not specifically provide for layered coding in the multiplex, but it should be possible to arrange this externally and transparently to MPEG2.

5. THE CABLE SYSTEM

Efforts are being made to develop and standardise cable systems. The first cable system developed is intended to provide onward transportation by cable networks of the satellite signals. The system is designed for 8MHz cable networks. It will have the following features.

* Nominally 64 QAM modulation, but can be also 16, or 32 QAM, and provision for future expansion to 256 QAM will be made.

* Roll-off 15% (under review). For a roll-off of 15%, in an 8 MHz channel, the limits on the useful bit-rate after the MPEG2 demultiplexer are 27.83Mbit/s for 16 QAM, 34.78 Mbit/s/s for 32 QAM, and 41,74 Mbit/s/s for 64 QAM. Reference to Table 1 will show how these match will the satellite transmitted bit-rates. For example, if the satellite service used a 33 MHz transponder, with a 2/3 convolutional code, the final useful bit-rate would be 31.6 Mbit/s before RS and 34.368 Mbit/s after RS. This could be carried directly either by a 32 QAM system with an occupied bandwidth of about 7.9 MHz, or by 64 QAM in an occupied bandwidth of 6.8 MHz.

* RS Code as used for satellite system is retained

* No convolutional coding is used.

* Convolutional interleaving, I = 12

The C/N required for noise free pictures will be significantly lower than for analogue signals.

TABLE 1: THE MPEG2 IMAGE CODING FAMILY
(November 1993)

High Level 1920 S/L 1152 lines	X	up to 80 Mbit/s	X	X	up to 100 Mbit/s
High-1140 Level 1440 S/L 1152 lines	X	up to 60 Mbit/s	X	up to 60 Mbit/s	up to 80 Mbit/s
Main Level 720 S/L 576 lines	up to 15 Mbit/s	up to 15 Mbit/s	up to 15 Mbit/s	X	up to 20 Mbit/s
Low Level 352 S/L 288 lines	X	up to 4 Mbit/s	up to 4 Mbit/s	X	X
	Simple Profile No B-frames 4:2:0 Not scalable	**Main Profile** B-Frames 4:2:0 Not Scalable	**SNR Scalable** Profile B-frames 4:2:0 SNR scalable	**Spatially Scalable** Profile B-frames 4:2:0 SNR scalable Spatial scalable	**High Profile** B-frames 4:2:0 or 4.2:2 SNR scalable, Spatial scalable

Notes:

1. Profile/level combinations marked X are not defined as compliance points
2. A maximum of one SNR enhancement layer (in addition to the base layer) is allowed in SNR Scalable, Spatially Scalable, and High Profiles
3. A maximum of one spatially scalable enhancement layer (in addition to base layer and SNR enhancement layer) is allowed in Spatially Scalable and High Profiles

Extensions to 256 QAM will probably require echo-correction systems which, in turn, will require a training signal to be transmitted. The option of using the MPEG2 multiplex Network Information Table for this purpose is being studied currently.

First tests of 64 QAM on some SMATV systems have shown that it gives promising results, and thus may be used for both CATV and SMATV.

6. THE TERRESTRIAL SYSTEM

The development of digital terrestrial broadcasting in Europe is at an earlier stage than the satellite system. As mentioned in earlier papers, there is much interest in Europe in developing a multilevel terrestrial broadcasting system, that can be received simultaneously in different viewing environments. The initial study suggested that a system which allows HDTV reception by rooftop antenna, and portable reception of conventional quality signals by rabbit ear antenna, would be a very attractive proposition.

At the current time, the most likely MPEG2 member capable of providing this, in the terrestrial environment, is the High-1440/ Spatial Scalable member. This can be arranged to provide an upper HDTV layer, and two lower layers. One is a conventional quality level, via the Spatial scalability, and the other an intermediate layer via the SNR scalability. This would thus provide a three- step graceful degradation characteristic, and the option of conventional quality reception, at the same time as HDTV reception.

There is also considerable interest in COFDM techniques to reduce the effects of multi-path, and to enable the use of single-frequency networks.

It should be possible to report more definitively about the technical prospects for such a system in about one year's time. Major demonstrations of digital terrestrial multi-layer schemes are planned for mid 1995 in Europe.

However, there are currently some reservations about the coverage prospects in Central Europe for HDTV-based multi-layer systems, and indeed about the coverage prospects for services to portable receivers, at least while existing PAL and SECAM transmissions are being radiated. There are thus many studies still to be made.

7. THE RECEIVER ARCHITECTURE

One of the commercial lessons in new technology may come from the PC story. There are tens of millions of MS-DOS PCs in use throughout the world, despite the fact that it operates like a series of after-thoughts. It is outstandingly successful. For the consumer, an extremely competitive market keeps prices down. Vast fortunes are being made in software because the market is so large.

More or less anyone can set themselves up to make and market PCs, because the system is essentially open, and the majority of systems have the same open architecture. It is this latter factor which has had the single largest effect on the popularity and growth of the PC population.

Open architecture means having a common data bus format, and a common way of assembling system elements. One of the advantages of open architecture is that it allows retro-fitting and upgrading, but this is not the most important advantage. The real advantage is that it brings competition for the same market. Each system module can be sourced according to cost or other criteria. Open architecture is a democratic system.

The author believes (although not all manufacturers share this view) that the option of open architecture for digital television receivers should be available. It should be possible to design a system with a common data bus at the multiplex level, which allows different application specific modulation modules to be used, or indeed different video and audio decoding modules to be used. The demodulation modules might include those for satellites, cable, terrestrial, VOD ADSL, etc. The source decoding modules might encompass the different quality levels. It has to be said however, that many manufacturers in Europe argue that the short term impact on receiver cost of this approach may make it impractical.

8. CONCLUSIONS

The initiative to develop digital television broadcasting in Europe now has much momemtum and support across all parts of the European programme-making and consumer electronics industry.

The systems developed for satellite broadcasting and for corresponding cable transport are

BW (at -3 dB) [MHz]	BW' (at -1 dB) [MHz]	R_s (for BW/R_s=1.27) [Mbaud]	R_u (for QPSK + 1/2 convol.) [Mbit/s]	R_u (for QPSK + 2/3 convol.) [Mbit/s]	R_u (for QPSK + 3/4 convol.) [Mbit/s]	R_u (for QPSK + 5/6 convol.) [Mbit/s]	R_u (for QPSK + 7/8 convol.) [Mbit/s]
54	48.6	42.5	39.2	52.2	58.8	65.3	68.5
46	41.4	36.2	33.4	44.5	50.0	55.6	58.4
40	36.0	31.5	29.0	38.7	43.5	48.4	50.8
36	32.4	28.3	26.1	34.8	39.1	43.5	45.6
33	29.7	26.0	24.0	31.9	35.9	39.9	41.9
30	27.0	23.6	21.7	29.0	32.6	36.2	38.1
27	24.3	21.3	19.6	26.2	29.4	32.7	34.4
26	23.4	20.5	18.9	25.2	28.3	31.5	33.1

R_u stands for the useful bit rate after the MPEG-2 MUX (see Fig.1)
R_s (symbol rate) corresponds to the bi-lateral Nyquist bandwidth of the modulated signal

Table 2: Examples of bit-rates versus transponder bandwidths

relatively straightforward ones, in terms of modulation. This will help their early introduction. They are nevertheless capable of the transparent transport of MPEG2 image and sound systems, and indeed any other data services that can be devised in the years ahead.

We believe that we have succeeded in our goal of invoking maximum commonality, and we are to use the most sophisticated coding image system available today, and a worldwide common multiplex system.

We hope that the satellite and cable systems will become official European Standards, via ETSI, in the coming months.

Services will start as soon as hardware is available, and are likely to begin in 1995.

9. ACKNOWLEDGMENTS

The work described in this paper was done by the DVB project, but the opinions expressed are solely those of the author.

The DVB Project is led by Peter Kahl of the German Federal Misnistry of Posts and Telecommunications. The Chairman of the DVB Technical Module is Prof. Ulrich Reimers of the Technical University of Brunswick, Germany. Both exhibit exceptional leadership qualities, and they have the thanks of the European media industry. Many other Chairmen and Engineers, (and even Accountants) have contributed significantly to the succes of the project.

10. REFERENCES

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EVALUATION OF MULTI-CARRIER TRANSMISSION SYSTEMS FOR ADVANCED TELEVISION IN NORTH AMERICA

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1. Introduction

Multi-carrier modulation (MCM), such as orthogonal frequency division multiplexing (OFDM), has been successfully demonstrated in digital audio broadcasting (DAB) for wide-band data transmission. In DAB, each MCM carrier is modulated using Differential Quadrature Phase Shift Keying (DQPSK) for mobile reception. There are many projects under development in Europe to extend the DAB success to digital television terrestrial broadcasting (DTTB), where higher order QAM modulation (16/64QAM) could be used to broadcast digital television over 7 or 8 MHz bandwidth [1]. Work is also being carried out at the Communications Research Centre (CRC) to investigate DTTB using OFDM in a 6 MHz bandwidth and to compare its performances to other techniques such as VSB and QAM [2].

ATV transmission using multi-carrier modulation may provide some advantages over single-carrier systems, such as

- flexibility in allocation of data capacity to different services;
- possiblity of single frequency network operation, on-channel repeaters or gap fillers [3];
- better performance under multipath distortion.

This report presents some preliminary specifications and results for multi-carrier transmission systems believed suitable for 6 MHz channels in an NTSC environment.

2. System Description

OFDM transmission systems for advanced television transmission in a 6 MHz channel have been simulated at the CRC. Their preliminary parameters are listed in Table-1 as System−1, System−2 which was proposed by the CCETT(France) and System−3 which was only used for the test on co-channel interference and is not proposed as a potential ATV transmission system.

System−1 has a higher data throughput of 19.99 Mbps (R-S code redundancy included), a reduced guard interval of $21.3\mu s$ and a sampling rate of 6 MHz. System−2 has a data throughput of 19.096 Mbps, a larger guard interval of $32\mu s$ and a sampling rate of 8 MHz [1]. Both systems could be used to provide an Advanced Television (ATV) service over-the-air. Trellis coded 64QAM with rate 4/6 coding is implemented, but trellis coded 32QAM with rate 4/5 coding is another option. Conversion from the OFDM to the higher data rate modulation selected for cable distribution would be done at the cable head-end.

[1] $21.3\mu s$ represents a path differencial of about 6.4 km and $32\mu s$ represents about 9.6 km.

3. COFDM System Performances under Combined Impairments of Random Noise and Multipath Distortion

Computer simulations were conducted to evaluate the trellis coded OFDM system performances under combined impairments of random noise and multipath distortions. Trellis code was used for these simulations, but other codes such as Turbo-code [5] could improve the performance. Figure 1 shows the bit error rate (BER) performances for the 64QAM COFDM system (System–2 in Table 1) with single multipath distortions for different desired-to-undesired ratios (D/U). One important advantage of a COFDM system is that its performance is independent of path delay variations, as long as the delay is within the guard interval. This is not the case for a single carrier modulation system using an adaptive equalizer; their performances degrade as the multipath delay increases. Figure 2 shows the BER performance simulation of System–2 for a Gaussian channel. The worst case scenario for a COFDM system with 0 dB multipath distortion is also presented in Figure 2. The CCETT hardware test result [4] is shown in Figure 2 to confirm the simulation results.

To simplify the simulation, only one level of channel coding, the inner code with trellis coded modulation (TCM), which is key to the system performance, was implemented. The outer code, the Reed-Solomon code, was not used. The system threshold of visibility (TOV) was estimated to be at BER $= 10^{-3}$.

4. OFDM System Interference into an NTSC Receiver

Laboratory tests were conducted at CRC to evaluate the impact of OFDM signal interference into an NTSC receiver, from a co-channel, upper and lower adjacent channel. An arbitrary function generator (AFG) was used to generate the System–3 (OFDM-16) and System–2 (OFDM-64) signals, as described in Table-1, as well as the OFDM-16H and OFDM-64H signals which use spectrum shaping techniques to open a spectrum hole at the NTSC vision carrier spectrum, at a cost of a throughput reduction of about 8%. This latter approach does not appear to be appropriate from a spectrum utilization point of view and the system would be sub-optimal after terminating NTSC services.

The TOV of the OFDM systems interference into an NTSC receiver are listed in Table-2 (a), (b) and (c). For comparison with single carrier modulation systems, the test results of the DigiCipher 32QAM system and a 4VSB system with a 0.7 dB pilot tone injection are also presented.

5. Interference into OFDM System

Because of the noise like nature of the OFDM signal, the ATV-to-ATV interference performance of a multi-carrier system is similar to that of a single carrier system.

The robustness of a multi-carrier system to NTSC interference will depend on the degree of error coding implemented and the amount of interleaving applied. Further studies are required to evaluate different options and to estimate their relative performance.

6. Other Impairments

A multi-carrier system should exhibit a much better performance compared to a single carrier system under impulse interference conditions since the interference will be averaged out over the entire OFDM block.

On the other hand, single tone interference might more severely affect a multi-carrier system. However, it can be handled by error-correction coding.

7. OFDM Signal Peak-to-Average Power Ratio

Peak-to-average power ratio tests were conducted on OFDM-16, OFDM-64, the DigiCipher 32QAM and 4VSB (this time with a pilot injection about 0.3 dB) signals. The measurements were performed on a digital osciloscope. The results are listed in Table-3. The corresponding test results from the Advanced Television Test Centre (ATTC) are also presented for comparison.

The results show that 99% of peaks fall within 5 dB of the average power for 32QAM and 4VSB and within 6.5 dB for OFDM systems.

8. OFDM System Sensitivity to the Phase Noise

No results are available on the robustness of the systems discussed above, but the experience with similar systems used for Digital Audio Broadcasting (DAB) would indicate that the phase noise should not pose a major problem with todays technology.

9. Conclusions

The performance COFDM systems is independent of path delay variations, as long as the delay is within the guard interval.

COFDM systems out-perform single carrier modulation systems for strong, long delays and multiple multipath distortion cases. They are good candidates for on-channel repeater, gap filler and single frequency network (SFN) implementations.

COFDM systems have comparable performance with single carrier systems for co-channel ATV interference into NTSC. But their peak-to-average power ratio is about 1.5 dB higher than that of single carrier systems.

Further studies are required to determine the best inner-code and the amount of interleaving needed for COFDM systems and to estimate their performance against NTSC interference and phase noise. These studies and the ones done on other proposed techniques, such as QAM and VSB, should provide very relevant information for selecting the best transmission system for DTTB in a 6 MHz channels.

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[4] ITU-R Document TG 11-3/TEMP/39-E, "COFDM demostrations", Geneva, Switzerland, October 1993.

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Figure 1: COFDM-64QAM BER Performances: Gaussian Channel and Single Echo

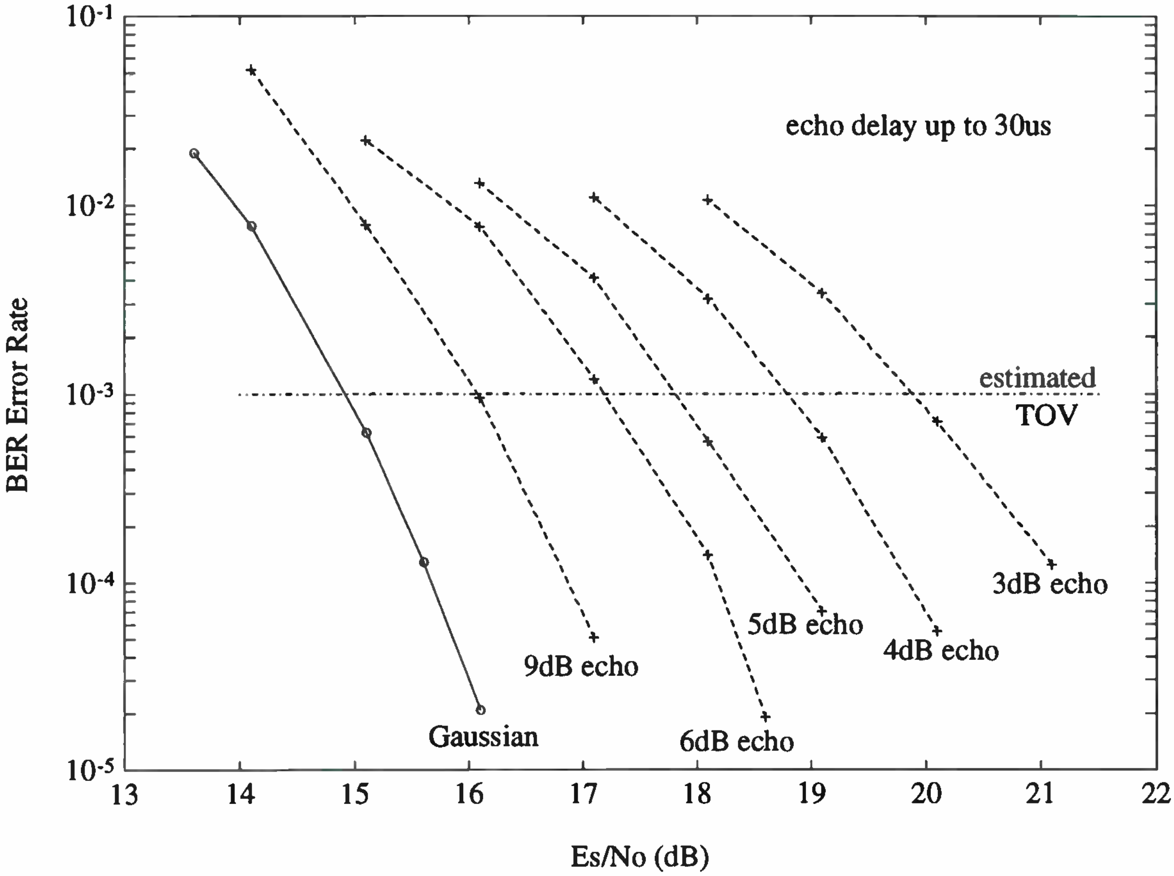
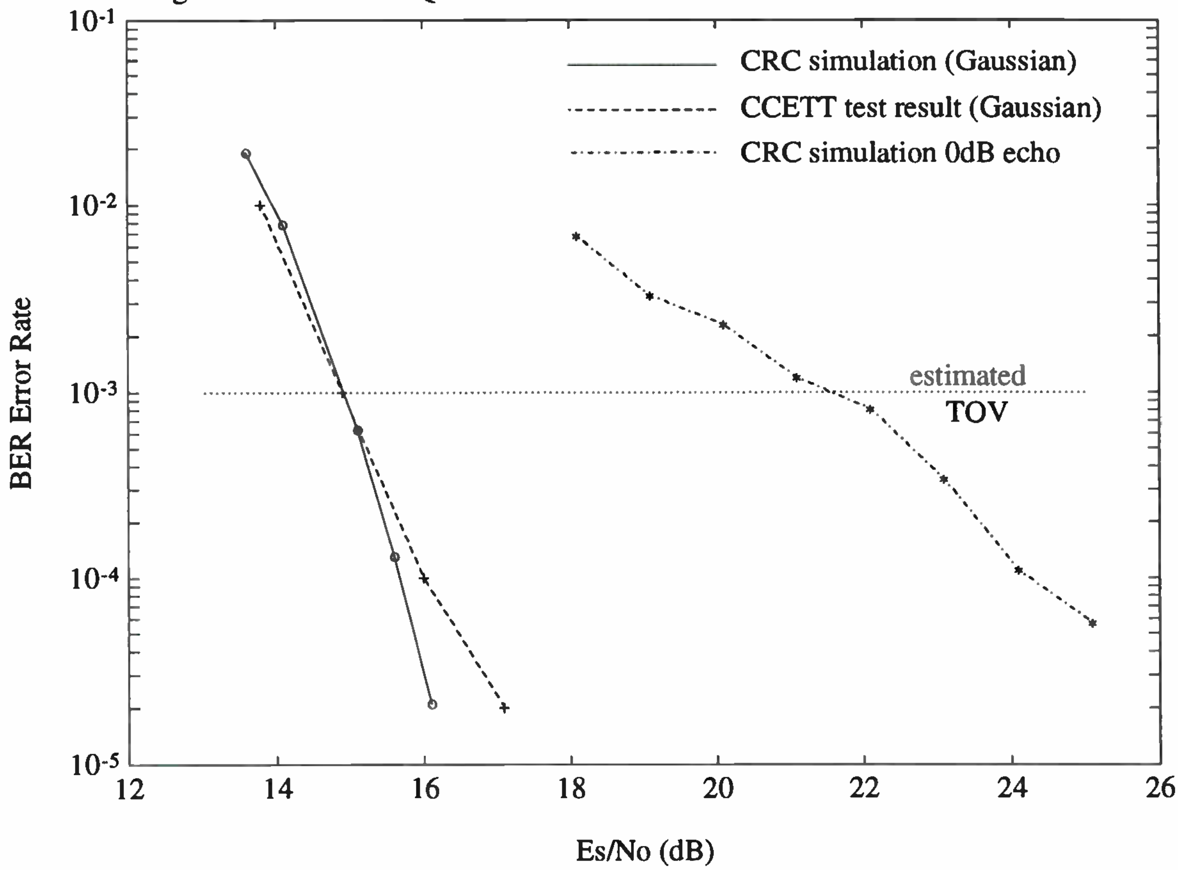


Figure 2: COFDM-64QAM BER Performances: Gaussian Channel and 0dB echo



System parameters	System–1	System–2	System–3
Channel bandwidth	6 MHz	6 MHz	6 MHz
actually used bandwidth	5.4375 MHz	5.5 MHz	5.09 MHz
FFT size	2048	2048	512
OFDM symbol length	341.333 μs	256 μs	100 μs
guard interval (multipath delay)	21.333 μs	32 μs	20 μs
total length of the symbol	362.666 μs	288 μs	120 μs
total number of subcarriers	1856	1408	512
carrier spacing	2.93 kHz	3.91 kHz	9.94 kHz
number of reference subcarriers	29 (1/64)	22 (1/64)	–
number of useful subcarriers	1827	1386	512
frame length	90.66 ms	72 ms	–
total number of symbols per frame	250	250	–
number of sync symbols per frame	2	2	–
number of useful symbols per frame	248	248	–
constellation of each subcarrier	64QAM$^{(*)}$	64QAM$^{(*)}$	16QAM
coding rate	4/6$^{(*)}$	4/6$^{(*)}$	1
gross spectral efficiency	4 bits/s/Hz	4 bits/s/Hz	4 bit/s/Hz
total data rate	19.99 Mbps	19.096 Mbps	16.97 Mbps

Table 1: System parameters for 6 MHz OFDM ATV transmission systems

(*) Trellis coded 32QAM with rate 4/5 coding has been investigated as a possible alternative.

Desired signal	NTSC signal C/I (dB) at TOV						
NTSC	white noise	4VSB .7 dB pilot	Digicipher 32QAM	OFDM-16 16QAM†	OFDM-64 64QAM‡	OFDM-16H 16QAM	OFDM-64H 64QAM
average of 3 test signals	46 dB	46.3 dB	48 dB	48.7 dB	47.7 dB	46 dB	46 dB
ATV relative to white noise	—	0.33 dB	2 dB	2.7 dB	1.7 dB	0 dB	0 dB

Table 2(a): TOVs of co-channel ATV interference into an NTSC receiver.

† System–3

‡ System–1 and 2.

Desired signal	NTSC signal C/I (dB) at TOV					
NTSC	4VSB .7 dB pilot	Digicipher 32QAM	OFDM-16 16QAM†	OFDM-64 64QAM‡	OFDM-16H 16QAM	OFDM-64H 64QAM
Average of 3 test signals	16.7 dB	1.0 dB	-0.67 dB	4.67 dB	-1.0 dB	4.67 dB

Table 2(b): TOVs of upper adjacent channel ATV interference into an NTSC receiver.

Desired signal	NTSC signal C/I (dB) at TOV					
NTSC	4VSB .7 dB pilot	Digicipher 32QAM	OFDM-16 16QAM†	OFDM-64 64QAM‡	OFDM-16H 16QAM	OFDM-64H 64QAM
Average of 3 test signals	-3.33 dB	-0.33 dB	-1.0 dB	0.67 dB	-1.33 dB	0.67 dB

Table 2(c): TOVs of lower adjacent channel ATV interference into an NTSC receiver.

Peak level above mean (dB)	32QAM Digicipher		4VSB .3 dB pilot		OFDM	
					OFDM-16	OFDM-64
	ATTC (%)	CRC (%)	ATTC (%)	CRC (%)	16QAM† (%)	64QAM‡ (%)
+2.0 dB	20.98	22.85	22.84	25.35	24.43	23.16
+3.0 dB	10.60	5.37	15.03	10.29	13.55	15.50
+4.0 dB	3.52	1.86	8.54	2.37	7.42	9.04
+4.5 dB	N/A	1.12	6.13	1.01	4.77	4.74
+5.0 dB	0.61	0.30	4.06	0.38	2.69	4.00
+5.5 dB	N/A	0.10	2.52	0.11	1.38	2.42
+6.0 dB	0.06	0.0075	1.41	0.005	0.86	1.22
+6.5 dB	0.007	0	0.71	0	0.46	0.77

Table 3: Peak-to-average power ratio

† System–3
‡ System–1 and 2.

MULTI-CHANNEL TV DELIVERY SYSTEMS

Monday, March 21, 1994

Moderator:
Jerry Butler, WETA, Washington, DC

DIRECTV™—A DIGITAL MULTI-CHANNEL DISTRIBUTION SYSTEM
John Hartwell
Sony Electronics Company, Inc.
San Jose, CA

***LARGE SCALE REDUNDANT LIBRARY SYSTEMS FOR INTEGRATED PROGRAM AND SPOT MATERIAL**
Philip Livingston
Panasonic Broadcast and Television Systems Company
Secaucus, NJ

***SATELLITE SCHEDULING AND UPLINK CONTROL FOR A MULTIPLE CARRIER PER TRANSPONDER OPERATION**
Scott Birdwell
Public Broadcasting Service
Alexandria, VA

MULTI-CHANNEL BROADCASTING AUTOMATION
George Fullerton
Louth Automation
Menlo Park, CA

***VIDEO ONRAMPS TO THE INFORMATION SUPERHIGHWAY**
Richard Mizer
Pacific Bell, Inc.
San Ramon, CA

MULTICHANNEL TRANSPORT OF VIDEO SIGNALS IN THE TELECOMMUNICATIONS EXCHANGE
Daniel Wirth
Bellcore
Livingston, NJ

*Papers not available at the time of publication

DIRECTV™—A DIGITAL MULTI-CHANNEL DISTRIBUTION SYSTEM

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Abstract

Multi-Channel video systems are not new to the broadcast community. However, broadcast facilities have traditionally been designed around the one-to-one relationship of the number of channels supported to the number of distribution systems in the facility, and the required number of operational staff. With the emergence of digital video compression technologies, the broadcast facilities of the future will be required to evolve into systems that support and deliver hundreds of channels efficiently and economically to an ever growing and divergent viewership. This paper is intended to explore this evolution using the *DirecTv*™ Castle Rock Broadcast Center system design as a case study.

Background

DirecTv™ utilizing two new generation high power satellites will deliver over 150 channels of programming to their customers. This national Direct Broadcast Satellite System is scheduled to start delivering programming to their customers in early 1994.

To support the goals of *DirecTv*™, Hughes Communications recognized that the requirements of 150 channels could not be supported economically using the present architectures of video distribution systems. Lacking a model to expand on, Hughes Communications, published a Request For Information identifying requirements believed to be vital to the success of the *DirecTv*™ program:

1. Support and distribution of an eventual 200-plus Channels of programming.
2. Technology that would support present high quality video and future Advanced Television signal formats.
3. System design to support the life expectancy of the *DirecTv*™ satellites, 15 to 20 years.
4. System design with automation integration allowing minimum staffing.
5. System to be designed and built over a 14 month period.

DirecTv™ Routing System Design

Several signal routing and switching systems have been utilized in the design of the *DirecTv*™ facility. Figure one depicts the five signal routing systems designed into the *DirecTv*™ facility:

1. TRS (Television Routing Subsystem) - A 512x512 270 Mbps digital routing matrix. The Television Routing System is responsible for all On Air switching and distribution of all program material used in the facility. In the implementation of this system, extensive use of embedded AES/EBU audio into the video data stream is used, simplifying the wiring of the facility.
2. MFDB CTL (Multi-Format Dub Control) - A 32x32 RS-422 control signal routing matrix. This system is responsible for the support of the Editing and Quality Assurance and Formatting (QAF) processes. The Edit and QAF processes share 28 tape transports, this routing system manages their control assignments.
3. SAPS (Servic Access Processing Subsystem) - 144x40 Multi-Level analog router for NTSC video and two levels of dual channel audio. This switching system is responsible for providing monitoring of all received analog signals external to the facility, and quick access to the thirty back-up signal processing chains.
4. DOWNLINK - A 192x16 Multi-Level analog router for NTSC video and two dual channels of audio. This switching system is for the support of monitoring functions in Broadcast Operation Control (BOC). Through the Downlink router, operators in BOC have the capability to view

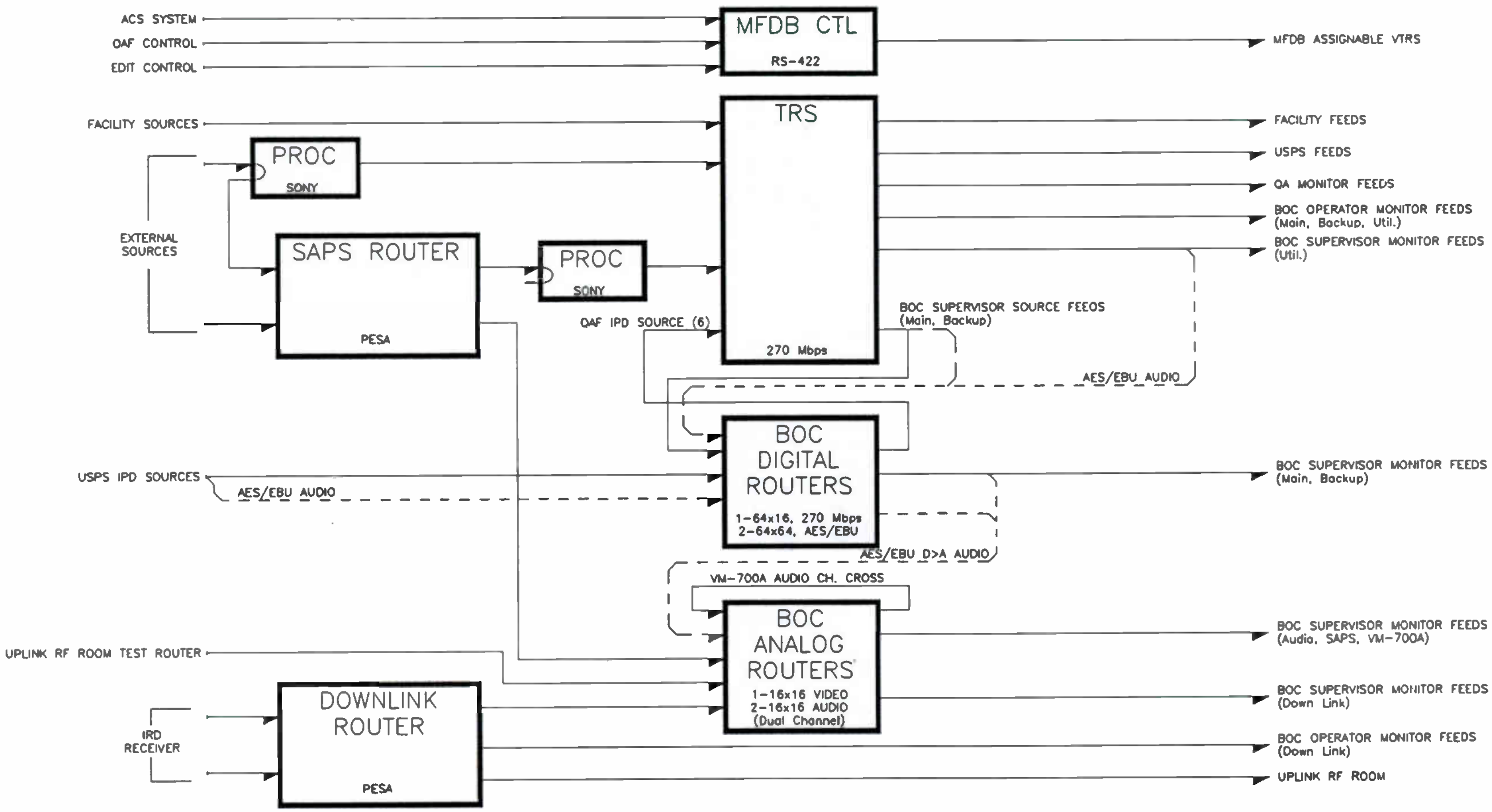

Figure 1 DirecTv™ Routing Systems

Downlinked programming distribute by *DirecTv*™.

5. BOC (Broadcast Operation Control) - One 64x16 270 Mbps digital routing matrix, two 64x64 AES/EBU routing matrix, and one Multi-Level 16x16 analog router for NTSC video and two dual channels of audio. This switching system provides the supervisors in BOC the ability to monitor, make technical measurements, and assign facility-wide monitoring paths for the digital compression equipment. The Multi-Level virtual control of this Multi-Format system is key to allowing the BOC supervisors to accomplish their assigned tasks.

TRS Router Design

The decision to make this ON-AIR routing system a component digital video matrix, was based on four factors:

1. Video Compression Compatability - The video compression system used by *DirecTv*™ is MPEG which is based on bit-rate reduction of a digital component CCIR-601 source.
2. Video Quality - Distributing programming from the CCIR-601 format early in the signal chain would eliminate any NTSC encoding/decoding anomalies.
3. Digital Tape Format - Digital video tape supporting CCIR-601, would require no operator set up on playback.
4. System Maintenance - Digital system designs eliminate all equipment required to equalize an analog signal carried over long lines.

The configuration of the TRS has been designed around the need to provide both primary and back-up signal paths to the Uplink Signal Processing System (USPS) that do not pass through common equipment frames. This is accomplished by splitting the TRS into two physical blocks. Shown in figure two, the primary source signals are contained on inputs 1 through 256, with the back-up signal sources on 257 through 512. Likewise the primary outputs of the TRS are similarly split with primary feeds from the TRS on outputs 1 to 256 and back-up feeds on outputs 257 through 512.

The distribution of program material through the TRS relies on the fact that audio and video are congruent. Up to four audio channels, two AES/EBU stereo audio pairs, are supported and routed with the video data as allowed and described in pending SMPTE standards.

In special cases, a spilt of the video and audio maybe required. For these instances separate signal paths are designed to split the audio and video sources.

MFDB Router Design

The Edit and QAF (Quality Assurance and Formatting) operations supported by this system are used to prepare house formatted tapes from distributor provided masters. The tasks performed by operators in these areas are to assure the maintaining of visual and aural quality of the recorded programming to be aired.

The configuration of the MFDB CTL system is designed to allow intermittently used equipment to be shared between the specialized task oriented control rooms. Accomplishing this requirement, this system provides for up to thirty-two video tape decks to be controlled and shared by the Edit room, six QAF rooms, the automated dubbing system, and three monitoring positions.

There are two important relationships to be met for the MFDB CTL routing system to achieve its operational requirements.

The first relation is the pairing of the control ports to outputs of the TRS system. Using Virtual crosspoint assignments, the router control system provides the switching of the MFDB CTL system's control source to the required VTR by the identification of the TRS source to the related TRS destination by the evaluation of two conditions;

1. Source video tape equipment is related to the destination that has selected it.
2. Target video tape equipment is related to the source selected for its input.

Once the control path is identified, it is automatically protected using a set of logical rules. Each of these rules provide the security so that control of video tape equipment will not be accidentally switched in the middle of an assigned task.

SAPS Router Design

The SAPS, Service Access Procssing Subsystem, routing system is a 140x40 multi-level router for analog audio and composite NTSC video. The SAPS system provides *DirecTv*™ with live and cable network turnaround programming. Signals received as inputs to the SAPS routing system are from a variety of reception equipment. Included in the reception equipment are satellite, fiber, and Off Air broadcast receivers. 116 of these received signals are converted to 270 Mbps component digital video before distribution through the TRS router.

The configuration of the SAPS routing system is designed to provide monitoring of the external signals as

PRIMARY PATHS

	INPUTS 512–449	INPUTS 448–385	INPUTS 384–321	INPUTS 320–257	INPUTS 256–193	INPUTS 192–129	INPUTS 128–65	INPUTS 64–1
OUTPUTS 1–64	DIGITAL ROUTER FRAME 71A STATION ADDRESS 01* -023 DVS-V6464	DIGITAL ROUTER FRAME 67A STATION ADDRESS 01* -022 DVS-V6464	DIGITAL ROUTER FRAME 63A STATION ADDRESS 01* -021 DVS-V6464	DIGITAL ROUTER FRAME 38A STATION ADDRESS 01* -020 DVS-V6464	DIGITAL ROUTER FRAME 42A STATION ADDRESS 01* -019 DVS-V6464	DIGITAL ROUTER FRAME 46A STATION ADDRESS 01* -018 DVS-V6464	DIGITAL ROUTER FRAME 21A STATION ADDRESS 01* -017 DVS-V6464	DIGITAL ROUTER FRAME 17A STATION ADDRESS 01* -016 DVS-V6464
OUTPUTS 65–128	DIGITAL ROUTER FRAME 71B STATION ADDRESS 01* -031 DVS-V6464	DIGITAL ROUTER FRAME 67B STATION ADDRESS 01* -030 DVS-V6464	DIGITAL ROUTER FRAME 63B STATION ADDRESS 01* -029 DVS-V6464	DIGITAL ROUTER FRAME 38B STATION ADDRESS 01* -028 DVS-V6464	DIGITAL ROUTER FRAME 42B STATION ADDRESS 01* -027 DVS-V6464	DIGITAL ROUTER FRAME 46B STATION ADDRESS 01* -026 DVS-V6464	DIGITAL ROUTER FRAME 21B STATION ADDRESS 01* -025 DVS-V6464	DIGITAL ROUTER FRAME 17B STATION ADDRESS 01* -024 DVS-V6464
OUTPUTS 129–192	DIGITAL ROUTER FRAME 70A STATION ADDRESS 02* -023 DVS-V6464	DIGITAL ROUTER FRAME 66A STATION ADDRESS 02* -022 DVS-V6464	DIGITAL ROUTER FRAME 62A STATION ADDRESS 02* -021 DVS-V6464	DIGITAL ROUTER FRAME 37A STATION ADDRESS 02* -020 DVS-V6464	DIGITAL ROUTER FRAME 41A STATION ADDRESS 02* -019 DVS-V6464	DIGITAL ROUTER FRAME 45A STATION ADDRESS 02* -018 DVS-V6464	DIGITAL ROUTER FRAME 20A STATION ADDRESS 02* -017 DVS-V6464	DIGITAL ROUTER FRAME 16A STATION ADDRESS 02* -016 DVS-V6464
OUTPUTS 193–256	DIGITAL ROUTER FRAME 70B STATION ADDRESS 02* -031 DVS-V6464	DIGITAL ROUTER FRAME 66B STATION ADDRESS 02* -030 DVS-V6464	DIGITAL ROUTER FRAME 62B STATION ADDRESS 02* -029 DVS-V6464	DIGITAL ROUTER FRAME 37B STATION ADDRESS 02* -028 DVS-V6464	DIGITAL ROUTER FRAME 41B STATION ADDRESS 02* -027 DVS-V6464	DIGITAL ROUTER FRAME 45B STATION ADDRESS 02* -026 DVS-V6464	DIGITAL ROUTER FRAME 20B STATION ADDRESS 02* -025 DVS-V6464	DIGITAL ROUTER FRAME 16B STATION ADDRESS 02* -024 DVS-V6464
OUTPUTS 257–320	DIGITAL ROUTER FRAME 69A STATION ADDRESS 03* -023 DVS-V6464	DIGITAL ROUTER FRAME 65A STATION ADDRESS 03* -022 DVS-V6464	DIGITAL ROUTER FRAME 61A STATION ADDRESS 03* -021 DVS-V6464	DIGITAL ROUTER FRAME 36A STATION ADDRESS 03* -020 DVS-V6464	DIGITAL ROUTER FRAME 40A STATION ADDRESS 03* -019 DVS-V6464	DIGITAL ROUTER FRAME 44A STATION ADDRESS 03* -018 DVS-V6464	DIGITAL ROUTER FRAME 19A STATION ADDRESS 03* -017 DVS-V6464	DIGITAL ROUTER FRAME 15A STATION ADDRESS 03* -016 DVS-V6464
OUTPUTS 321–384	DIGITAL ROUTER FRAME 69B STATION ADDRESS 03* -031 DVS-V6464	DIGITAL ROUTER FRAME 65B STATION ADDRESS 03* -030 DVS-V6464	DIGITAL ROUTER FRAME 61B STATION ADDRESS 03* -029 DVS-V6464	DIGITAL ROUTER FRAME 36B STATION ADDRESS 03* -028 DVS-V6464	DIGITAL ROUTER FRAME 40B STATION ADDRESS 03* -027 DVS-V6464	DIGITAL ROUTER FRAME 44B STATION ADDRESS 03* -026 DVS-V6464	DIGITAL ROUTER FRAME 19B STATION ADDRESS 03* -025 DVS-V6464	DIGITAL ROUTER FRAME 15B STATION ADDRESS 03* -024 DVS-V6464
OUTPUTS 385–448	DIGITAL ROUTER FRAME 68A STATION ADDRESS 04* -023 DVS-V6464	DIGITAL ROUTER FRAME 64A STATION ADDRESS 04* -022 DVS-V6464	DIGITAL ROUTER FRAME 60A STATION ADDRESS 04* -021 DVS-V6464	DIGITAL ROUTER FRAME 35A STATION ADDRESS 04* -020 DVS-V6464	DIGITAL ROUTER FRAME 39A STATION ADDRESS 04* -019 DVS-V6464	DIGITAL ROUTER FRAME 43A STATION ADDRESS 04* -018 DVS-V6464	DIGITAL ROUTER FRAME 18A STATION ADDRESS 04* -017 DVS-V6464	DIGITAL ROUTER FRAME 14A STATION ADDRESS 04* -016 DVS-V6464
OUTPUTS 449–512	DIGITAL ROUTER FRAME 68B STATION ADDRESS 04* -031 DVS-V6464	DIGITAL ROUTER FRAME 64B STATION ADDRESS 04* -030 DVS-V6464	DIGITAL ROUTER FRAME 60B STATION ADDRESS 04* -029 DVS-V6464	DIGITAL ROUTER FRAME 35B STATION ADDRESS 04* -028 DVS-V6464	DIGITAL ROUTER FRAME 39B STATION ADDRESS 04* -027 DVS-V6464	DIGITAL ROUTER FRAME 43B STATION ADDRESS 04* -026 DVS-V6464	DIGITAL ROUTER FRAME 18B STATION ADDRESS 04* -025 DVS-V6464	DIGITAL ROUTER FRAME 14B STATION ADDRESS 04* -024 DVS-V6464

*IDENTIFIES 'S-BUSS' LINE

Figure 2 DirecTv™ TRS Routing Matrix

received. In total there are 138 external signal paths into the SAPS routing system. A number of these signals are dedicated to SAPS processing equipment, with the balance available through the SAPS routing system to thirty additional sets of processing equipment chains. Once an external signal passes through a SAPS processing chain, the signal can be used as either a primary or back-up signal source for program distribution.

Down Link Router Design

The Down Link routing system is a 192x16 multi-level router for analog audio and composite NTSC video, it is designed to provide monitoring of the *DirecTv*™ viewer channels as received through the down link system. Each of the input signals to this router are also distributed to dedicated 8 inch color monitors mounted in one of four BOC monitor walls.

The output of the Down Link router is used for the monitoring of the Off Air signals at each operational position in BOC and a test routing system in each Up Link RF room. These feeds are displayed on waveform monitors and dedicated color monitors switched in unison with monitor feeds from the TRS.

BOC Router Design

The BOC routing system is a multi-format, multi-level system. Comprised of one 64x16 270 Mbps digital routing matrix, two 64x64 AES/EBU routing matrix, and one Multi-Level 16x16 analog router for NTSC video and two dual channels of audio. This switching system, shown in figure three, provides the supervisors in the BOC the ability to monitor, make technical measurements, and assign facility wide monitoring paths for the digital compression equipment. The Multi-Level virtual control of this system is key to allowing the BOC supervisors to accomplish their assigned tasks.

Broadcast Operation Control, BOC

The BOC is the area designed to support the On-Air operation of *DirecTv*™. Operational tasks assigned to the staff in this area are the monitoring of all *DirecTv*™ channels, acquisition and set up of external signals, problem identification, and supervision of program automation systems.

The BOC is divided into six operational areas, each supporting an operating engineer. The physical layout places the four operators side by side, with a supervisor elevated behind each set of two operators. All positions have good sight lines to four separate monitor walls located in front of each operator position. Depending on the data compression used, each operator could be supporting up to 48 channels, with the ability to be expanded to 64. Designed for a normal staffing of two operators with one supervisor supporting each satellite, the design also allows for fewer operators during non-peak times or when fewer channels are in use.

Each monitor on the BOC monitor walls is identified by the transponder number and multiplexed channel. This channel identification is mounted on each monitor just below the screen. Just above each set of two horizontal monitors is a Source Identification Display (SID) to display the TRS source currently distributed to the channel the monitor below it represents. When a multiplex channel is not in use the display will indicate 'OFFLINE'.

Located in the center rack of each monitor wall are three 19 inch color critical monitors, each having a SID panel above. These monitors represent, top to bottom, a channel's Down Link signal, main program source, and assigned back up program source. The selection of the channel monitored is through the operators Channel Selector Panel (CSP). The SID panels provide the operator the TRS physical to logical relationships of the signals shown.

At each operator's station, there are three control panels to control the routing of the signals to be monitored. Each control panel has special relationships to the monitoring paths, control of signals to be fed to the USPS, and remote control of local equipment.

The Channel Selector Panel, or CSP is physically two Sony BKS-R3204 router control panels operated as a 64 button per source panel. These buttons are assigned to select the three sources displayed on the 19 inch color monitors on the monitor wall. Each of these buttons are dynamically mapped so that each is married to an 8 inch color channel monitor on the monitor wall and the channel it displays. Using the dual color illumination of this panel the, operator is communicated the following information:

- GREEN The selected channel that is displayed on the 19 inch monitors.
- RED Indicates a fault for the channel assigned to the button. When this indication occurs the SID panel above the channel monitor on the monitor wall will also indicate a fault.
- AMBER The selected channel currently viewed on the 19 inch monitors has a fault.

The CSP is a major component of the BOC monitoring operation. Information that maps each button is used by the Alarm Select Panel (ASP) described below, the automation system, and a corresponding CSP panel in the supervisors area.

The Alarm Select Panel (ASP) provides the operator with indications of problems at other operator stations in

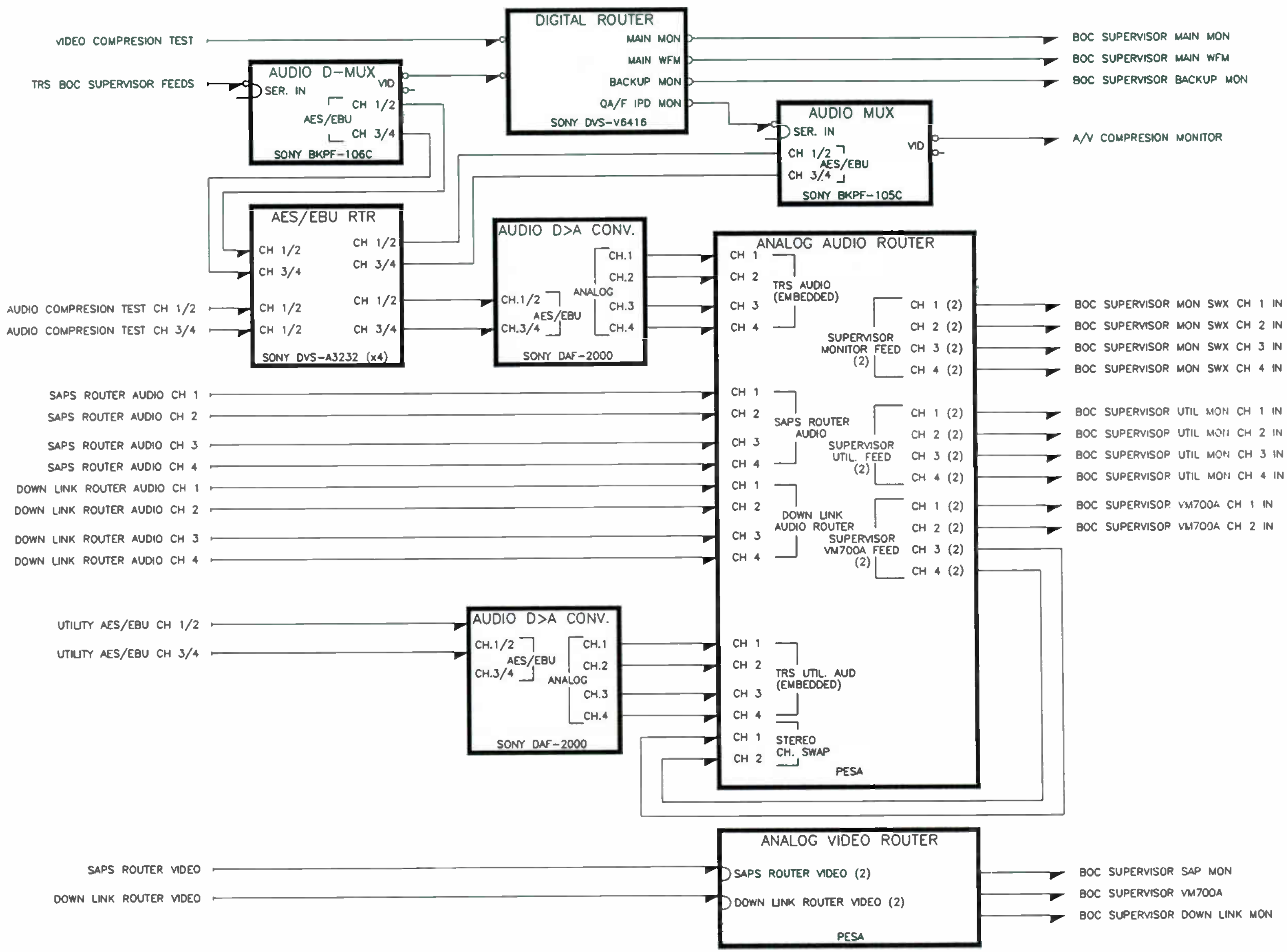

Figure 3 DirecTv™ BOC Routing System

BOC, selections to correct faults controlled at their station, and selections for local monitoring.

Across the top of the ASP are illuminating switches that display the condition of all operator stations in the BOC, and select the map of the local station's CSP. Normally the local CSP panel at the station is mapped to match the channel monitors on the monitor wall directly in front of the operator. When an alarm is noted at a BOC operator station any BOC engineer can select the alarming station and map the local CSP to match the station that is displaying the alarm.

To the left of the ASP are operator fault decision buttons. These buttons indicate the possible actions an operator could take in the event of a fault on the channel selected on the local CSP. When an action is available, the respective button is illuminated with green. When an action is taken the indicator is turned off until a decision once again becomes valid.

The remaining area of the ASP is for control of the local monitoring equipment at the operator station. The monitor selections allow the operator to hear the chosen audio in the area monitor speakers, and view the associated video on the local waveform monitor.

When a fault is detected by the automation system, the respective tally on the SID panel above the channel monitor will illuminate RED . A fault will also cause the associated channel monitor tally to flash and the previously described indications on the associated CSP and ASP control panels.

The last panel at the operator station is a X/Y router control panel. This panel is normally dedicated for the operator to select a feed for utility monitoring. Also made available through this panel are the routed destinations feeding all the monitors located in operators work area.

The supervisors area is designed to support and assist the BOC operators, system fault isolation, and system adjustments. To achieve these goals equipment is located in the supervisor's console to mimic the operator stations. Although no monitor wall is dedicated to the supervisor, sight lines to the operator station monitor walls are provided, and three local monitors are available at the supervisor's station for critical monitoring. Since the same monitors are required for other duties, the inputs to these monitors are routed through the BOC router system.

The monitors to view the Down Link, main program source, and back-up source are located to the left of center in the front section of the console and from left to right are; a 13 inch color critical monitor for the main program source, 8 inch fine pitched color monitor for the back up program source, and an 8 inch fined pitched color monitor for the Down Link. Just above these monitors are two SID panels. The panel above the main program source displays the TRS source name on the left and the TRS physical source input number on the right. On the panel above the back up source and Down Link, the left half of the display indicates the TRS source name for the back up signal source, and the right half indicates the BOC channel name.

Like the operator station, the same three control panel types are located near the supervisor for easy access and use. These three panel types are very similar in appearance and operation, but in implementing switches the BOC router in conjunction with the other *DirecTv*™ routing systems described earlier.

Router Control System

Controlling the routers is a custom router control system. This system was designed to control the routers in real time, and allow easy changes to their configurations. To accomplish this, distributed processing and modularized software was developed using configuration files defining the relationships between the different routing systems, automation, and operator control panels.

The control system defines a virtual single X/Y matrix, based on configuration files. Each routing system is then defined by the placement of it's crosspoints within this matrix. The limits of this matrix have been arbitrarily set to make the routing systems appear to be one 1000 x 1000 router.

Other configuration files are used to set the personality and logic of the relationships of requests and actions. The relationship of the MFDB router to the automatic protection of signal paths, as discussed earlier, is one example. In BOC the relationships of the CSP and ASP control panels, and multiple signal formats to be monitored are others.

Conclusion

The *DirecTv*™ facility represents the first of a new generation of multi-channel distribution systems. The design makes use of maturing technologies put into an environment that is familiar to experienced operators. As multi-channel facilities proliferate in the coming years, the experiences learned with *DirecTv*™ along with new emerging technologies, will move the next generation of facility designs away from these familiar architectures.

But for now, the combination of the routing systems designed into *DirecTv*™, provides the major infrastructure for the facility. As with all infrastructures, there is a complex matrix of relationships that allow the operational staff to perform their daily tasks. It was the goal of the design team to identify these relationships and address them in the design of this multi-channel distribution system. The success of their efforts will only be realized as the facility is stimulated by operators in the coming year.

MULTI-CHANNEL BROADCASTING AUTOMATION

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Automation systems play a pivotal role in controlling the much more complex environment of operating several channels from a central location. These channels, or feeds, can either be totally independent of each other (asynchronous), simulcast (synchronous) or a combination of both.

The management of large tape libraries and multiple As-Run logs; centralized feeding of sectional commercials; local control of commercial insertion; and the simultaneous control of many devices from different vendors, including multiple cart machines, can only work under the control of a sophisticated automation system. This system must maximize utilization of old and new equipment and perform many functions concurrently--functions such as automated recording, dubbing, compiling, and random playout.

CURRENT ENVIRONMENT

During the past decade, emerging business pressures have changed the way broadcasting is managed.

Major challenges, such as declining advertising income, coupled with the increased emphasis on local news, have added overhead without the revenue to offset the cost. The decline in spot revenue is principally due to the increasing number of participants competing for their share. This has been mainly the result of an active cable market, but the new era of expanded services such as DBS and the entrance of the TELCOS makes the market even more competitive.

Historically, the action from broadcasting management has met this challenge by reducing costs in all functional areas: sales, production, traffic, etc. Nowhere has this cost-cutting pressure been felt more keenly than in engineering.

Automation has played a key role in providing engineering management with tools to cut costs and increase productivity. Installation of robotic cameras, cart machines, satellite control systems, and master control automation have all contributed to meeting business targets and objectives.

Total station automation systems are now available to tie all of the automation subsystems together. These systems allow for more efficient use of a station's large investment in equipment and applications. They virtually eliminate lost revenue from blown or clipped spots, while at the same time they streamline operations. Total station automation also provides flexibility for last second changes. It supports insertion of local commercials, and offers new functionality, such as the ability to do on-line editing.

But the most important role for broadcasting automation is just emerging, and that is to provide a smooth migration from today's single-channel operations to the future environment of multiple-channel offerings from a central facility.

FUTURE ENVIRONMENT

Past history shows that multi-channel broadcasting has been more prevalent in the international broadcasting community. This is due primarily to a more pragmatic business approach by government regulating agencies. In the U.S., competitive factors as well as environmental changes are now accelerating plans to add additional channels that utilize the overhead in equipment, facilities, and human resources.

The business climate affecting these changes is as follows:

- A small station(s) in geographic proximity to a larger (host) station, that has gone dark or is in financial difficulty, could be profitable if totally supported by the host facility either through purchase, lease-back, or partnership.

  In this scenario, the host location would provide all services, including insertion of local commercials. Programming could be synchronous, asynchronous or a combination of both, all controlled by an automation system's separate playlists and As-Run logs.

- Established stations adding new channels as business opportunities arise. These could be foreign language, sports, news, or cable channels negotiated through "must carry" provisions. With each additional offering, the economies of scale are enhanced for all of the parties.

- The contemplated FCC ruling of awarding a second channel for HDTV broadcasting. The best guess is that initially the NTSC and HDTV channels will operate mostly asynchronously, with a gradual evolution to simulcast events, and then on to a final target of HDTV only.

LOCAL INSERTION

The common thread that runs through all of the scenarios is the need to facilitate localization of the material to the viewing audience. Material such as commercials, promos, news and programs should be deliverable to the targeted market just as if the facility were physically located there.

The networks also play a role in offering localization of commercial materials from their national feeds. The requirement exists to synchronously switch to sectional spot breaks, an application that is similar to a host stations supporting nearby remote stations.

An example of this kind of operation is when a nationally transmitted commercial from a tire manufacturer features snow tires in New England and rain tires in Florida, both during the same break. To offer this type of multi-channel service to national customers requires the automation system to be frame-accurate at all times for the synchronous insertion process. Recently, a West Coast network automated its three regional feeds, allowing for five sectional commercial breakaways within each region.

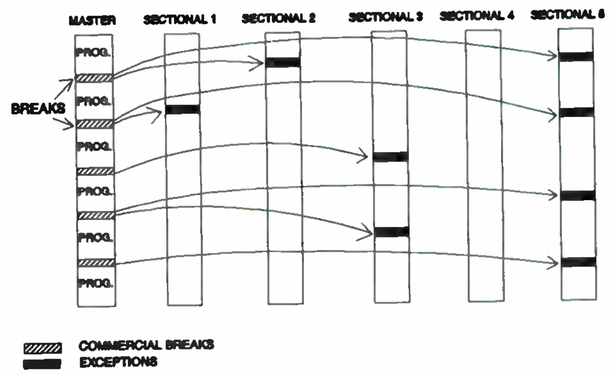

FIGURE A - SECTIONAL SPOT BREAKAWAYS

In this scenario (Figure A), the automation system automatically pre-compiles the commercials from cart machines to external record VTRs. Since the system is frame-accurate, last-second changes can be made without recompiling the whole tape. Changes are made by insert-editing over the previously recorded spot.

PRE-COMPILING BREAKS

Pre-compiling from cart machines is an important option to consider in supporting multi-channel broadcasting. Pre-compiling allows maximum utilization of multi-spot tapes resident in a cart machine.

Currently, some cart machine vendors provide multi-spot software for on-air applications. These require duplicate tapes to be prepared and stored in the cart machine in case of on-air conflicts. This software is not suitable for multi-channel on-air operation because of the complexity of managing the elevator, VTRs, bins, and carts while running multiple lists simultaneously. With the proper automation system software, compiling is performed hours prior to air, allowing time for conflicts to be resolved in the cart machine (two spots on a master multi-spot tape). If a spot is not immediately available, the system will skip to the next spot (leaving space for the insert), and then automatically return and record it when available.

Since minimal tape handling is important in managing multiple channel operations, pre-compiling allows much more material to be resident in the cart machine. Unlike the direct-to-air multi-spot system, the entire tape can contain spots (shuttle time is not an issue), and duplicate tapes are not necessary. The combination of these two advantages allows approximately four times as many spots to be in the cart machine than with the direct-to-air version.

With pre-compiling, the cart machine becomes a massive storage library containing all commercials and promos for all channels. Based on bin capacity, a cart machine could contain up to 20,000 spots.

TAPE MANAGEMENT

A multi-spot version of a tape management system, supplied by the automation system vendor, should be utilized to provide the information base for all the automation processes. This would include a large centralized database located on the network for easy access to various users and applications.

Again, to minimize tape handling on the front end, dubbing should be automatically transferred from the one inch source tapes directly into multi-spot cassettes in the cart machine. From that initial process, tapes should never have to be handled again.

The tape management system would identify each tape with bar-coding and spot the individual commercials with timecode user bits. This process would be coordinated over the network from a Traffic System Dub list.

THE SYSTEM

In a complete system, a networked workstation would run an "automated compile list" to sequentially construct the breaks for a prescribed amount of time. These compile lists would be electronically communicated from Traffic, and the actual compiling process would require minimal supervision in engineering.

The compiled breaks, supporting each channel, could record into VTRs in the library cart machine, or to another cart machine dedicated to satellite recording and playout. This playout cart machine could hold larger tapes for storing and playing segmented program material. In either case, the completed break tapes are stored in bins until needed to support that particular channel at air time (Figure B).

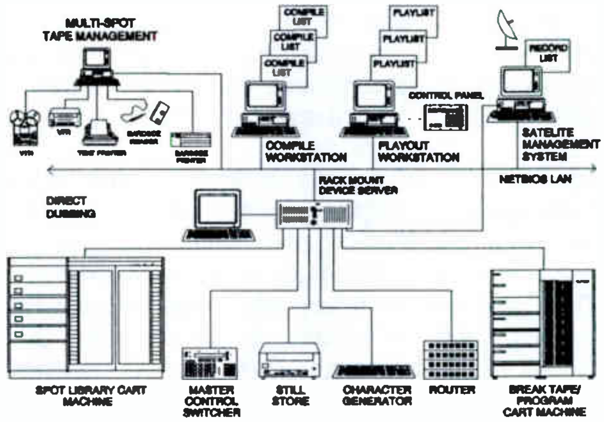

FIGURE B - MULTI-CHANNEL AUTOMATION - NO TAPE HANDLING

This system will also interface to external applications, such as automated satellite control systems and media delivery applications. Products that automatically segment program material can be recorded directly into the playout cart machine. Spot commercials delivered by satellite can also be recorded directly into the spot library cart machine. Multiple concurrent record lists are managed by the automation system. The information from all external sources is entered into the automation system's central database.

Multiple playlists simultaneously play the breaks, promos and programs to air. If last second changes are required during playout, a recorded spot on the compiled break tapes can be overridden with a new spot by editing (cut and paste) the active playlist.

A key benefit in this type of system is that the tapes are never removed from either cart machine, with the possible exception of archiving program tapes for future use.

OTHER SYSTEMS

Advances in technology are presenting alternate options for automating multi-channel operations.

Some of these options are available today and others are imminent.

Omnicart

At NAB last year, the first highly integrated cart machine designed specifically for multi-channel support was announced. This product, Odetics' Omnicart, is capable of performing the entire previous application scenario internally. As with the previous systems, a single location can use multiple programming and commercial feeds to address specific regions within a market.

Advanced multi-channel software, from Louth Automation, manages recording, program replay and spot creation utilizing pre-compiling. The system contains two elevators, up to eleven internal VTRs, and can be totally integrated with external devices such as master control switchers and routers. This system can run up to eight concurrent lists; record, compile or playout. One of the key ingredients is the ability to pre-compile break tapes prior to air.

Video Disk Products

The evolution of systems that support multi-channel broadcasting will rely heavily on new video disk technology. Advances in digital compression are starting to revolutionize the industry.

Initially, TV quality video disk products will be limited in storage capacity and consequently better suited for on-air presentation rather than for mass storage of material. Disk products are now entering the market that can economically store one to four hours of commercial material and promos.

It is predicted that eventually disk technology will replace tape in broadcasting as disk costs go down and storage capability increases. For the foreseeable future, though, both tape and disk technology will co-exist in broadcasting.

Instead of making installed cart machines obsolete, disk products could extend the useful life of these cart machines in multi-channel applications by converting them to tape storage library devices from on-air spot players.

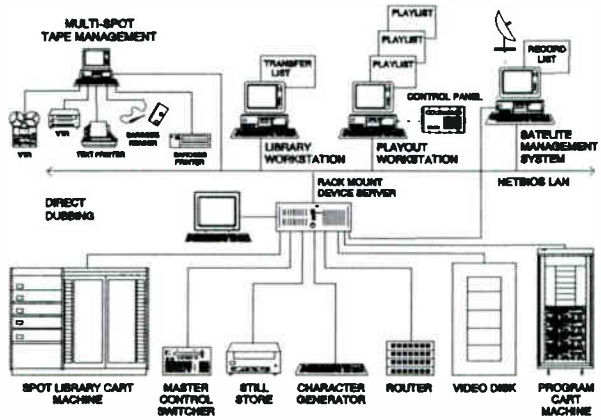

FIGURE C - MULTI-CHANNEL AUTOMATION - DISK/TAPE

Disk Buffer

In this system (Figure C), the disk performs the role of an on-air buffer. On demand, active spots are transferred from the library to the disk to support breaks for several hours of broadcasting. Spots to be used frequently would remain in the disk and new material would be transferred by the automation system as ordered by Traffic. All commercials are randomly played out from the disk by the automation system playlist.

Program material could be stored and played out from external VTRs, the dedicated playout cart machine or, if the capacity exists, directly from the tape library cart machine. All of these options are manageable by a multi-channel automation system.

The disk product eliminates the need to pre-compile and provides real-time playout for multiple channels. It also helps reduce a problem inherent in pre-compiling - tape and VTR wear.

All Disk System

The final scenario (Figure D) depicts multi-channel support in an all-disk environment. The program material could also be stored and played on disk or could stay on tape. This disk product would perform both the library and playout functions. Multiple "spigots" would allow multiple lists to be played out concurrently. The buffer, if needed to perform the on-air role, could be located internally. This system totally eliminates the problems of VTR and tape wear.

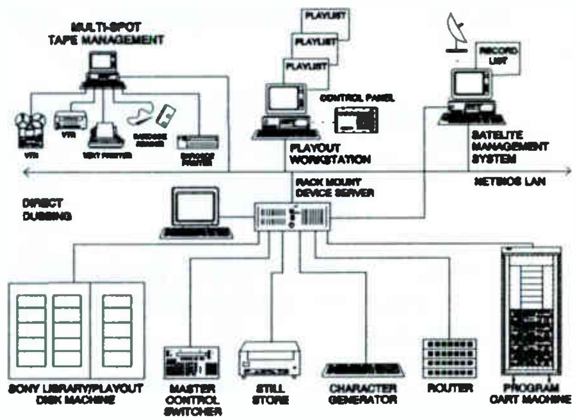

FIGURE D - MULTI-CHANNEL AUTOMATION - DISK

TECHNOLOGY

Automation technology must take into consideration that no two broadcasting environments are the same. This is certainly true in the multi-channel application environment. The technology should facilitate ease of customization for each end-user. This customization should be quickly deliverable to the end user with the ability to extend the system in the future as new devices and applications are added.

Software is the only cost efficient way to deliver automation systems that are easily customized for each user.

SOFTWARE DEVELOPMENT

The best development technology to produce customizable software is Object Oriented Programming (OOP). Once a software object is designed and developed to perform a function (device drivers, network management, etc.), it is reusable and extendable to support the same family of products, with minor changes, for different models and vendors.

An example of an actual OOP application is shown in Figure E ("VTR Objects"). All VTRs perform the same functions. Once the software object has been created and tested to perform these functions, it can be used to support all types of VTRs. The only additional development needed is to address the physical differences supplied with each vendor's protocol. The standard functional object, along with the physical characteristics, is then incorporated into a new VTR object.

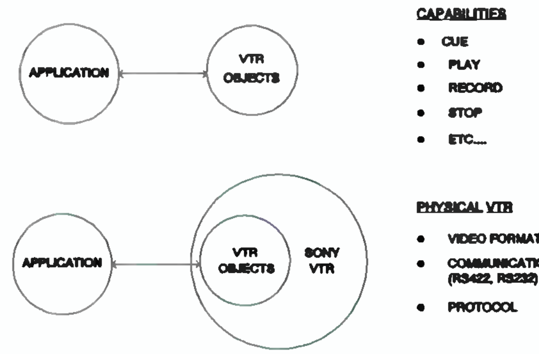

FIGURE E - VTR OBJECTS

CLIENT/SERVER ARCHITECTURE

To maximize the investment in an automation system, the technology should also allow multiple clients to access its benefits. It should provide a link to external applications that require device control, with an easy to execute development procedure.

This can best be accomplished by a versatile implementation of Client/Server architecture. In this structure, the workload is distributed across the network as needed. The Server (computer device controller) can do real-time work while the clients (workstations on the network) can do various background tasks such as editing playlist, media verification, and As-Run logging.

The clients, which reside on the network, should be available wherever and to whomever they are needed in the facility. The status of each device supported by the Server should be broadcast to each client at every frame.

In addition to performing the master control function, the clients can host individual applications supplied by other vendors (newsroom systems, satellite control systems, etc.) or custom applications (user interfaces, other platforms, etc.) developed by the end user or an integrator.

APPLICATION PROGRAM INTERFACE

This automation system and application co-residency is made possible by the Automation vendor providing an Application Program Interface (API) that acts as a bridge to the external application (Figure F). The API should be operating system and programming language independent and easy to use. It resides on a client whose purpose is to provide the "hooks" that allow this linkage. The end result is a tight integration of both systems that provides full device control to the external application.

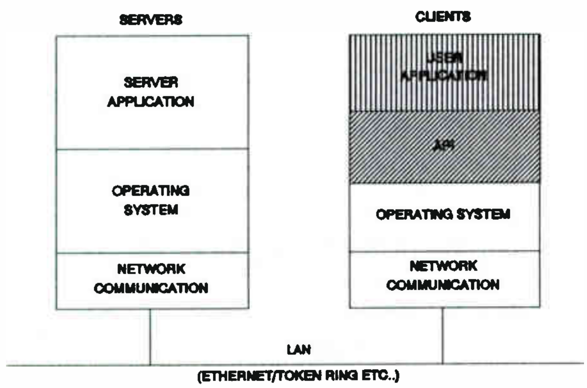

FIGURE F - AUTOMATION SOFTWARE API UTILIZATION

DEVICE INTEGRATION

All devices controlled by the Server, including cart machines and video disk Servers, should be integrated with the Device Server to support multi-channel broadcasting. Cart machine subsystems are difficult to integrate if the cart machine operating system is different than the automation system's overall operating software. To fully support multiple concurrent lists, the automation system's operating software should be capable of displacing the vendor's native software, making it a peer with the other devices under control of the Server. At the same time, the cart machine must maintain and, in many cases, extend the functions and benefits offered by the original manufacturer.

EXTENDIBILITY

Finally, the automation system technology should provide almost unlimited extendibility to support new devices and applications.

The system should be designed to be "future-proof" by being able to control future devices such as disk based systems and new applications like pay-per-view services.

It must support the user's next generation of requirements.

SUMMARY

The technology to automate all of the applications and systems described in this paper are developed and available today.

Broadcast engineering can no longer rely on manually intensive operations to deliver the services needed to remain competitive in this changing environment.

Leading edge automation systems provide the framework to expand to multi-channel operations without adding the cost of new facilities and resources.

MULTICHANNEL TRANSPORT OF VIDEO SIGNALS IN THE TELECOMMUNICATIONS EXCHANGE

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Livingston, New Jersey

ABSTRACT

There are several video and associated audio multichannel transport systems available to the telecommunications industry capable of distributing television and other video signals. Principal telephone company *Hub* locations, such as the larger Television Operating Centers (TOCs or TVOCs) in many major cities and customer serving central offices in other areas, provide local access to broadcasters, cable television operators and other parties interest in transversing the local exchange within a local exchange company (LEC) or for transport to the interexchange carrier(s) (IC) of choice. The architecture schemes may vary for each application, coupled with the quality of service required. For broadcast television quality signals and in some cases television signals approaching broadcast quality, multichannel video service is primarily transported by the LECs on fiber optic facilities that can be arranged within the analog or digital systems transport domain. However, it is important to point out that commingling the two disciplines may produce questionable quality results, compounded by variations of circuit design complexities. Multichannel distribution can be configured as point-to-point or point-to multipoint within a single or bi-directional path. This paper will discuss the various types of architectures associated with multichannel video distribution arrangements, with simple to complex analog and digital system examples.

BACKGROUND

Terrestrial and satellite transmission systems [both] offer intermediate solutions to cross country transport of video service. Prior to the AT&T Divestiture of the Bell System in 1984, terrestrial microwave radio and coaxial cable facilities provided coast to coast transport of broadcast television service and was commonly used by the television networks. Satellite transmission became more and more popular in the early 1980's and by 1984, at the time of this divestiture, national and international satellite transport of television broadcasting was rapidly embraced as the primary mode of transport. The older Bell System terrestrial network was divided into 22 separate operating companies, within seven regions throughout the United States. Some believed that this division may have jeopardized television service, as confidence levels wained by the fragmented and older cross country telecommunications terrestrial links.

Since this "Quickening", the Regional Bell Operating Companies (RBOCs) shifted to fiber optic technologies, adding greater stability to terrestrial systems and facilities. They have taken great strides by adapting to the economic demands of this important service, making available the necessary interfaces and state of the art modes of transmission. Credences are being restored as quality television terrestrial transport is again being demonstrated within and between LATA boundries, intra-interstate, coast to coast. Television transported on terrestrial analog/fiber and digital/fiber optic systems is safe. The service is reliable, which also extends to several other video service offerings approaching broadcast quality, i.e. cable transport, business television, video teleconferencing.

The telephone companies have migrated into broad band delivery systems and are providing voice, data and video transmission services within their serving areas. This has been accomplished with fiber optics and hybrid technologies. Most video services provided terrestrially today are carried over analog or digital transmission systems and fiber optic facilities, distributed in single or multichannel arrangements. Fiber optic terrestrial, multichannel transmission systems rival the options.

ARCHITECTURES

- Figures show single or bi directional transmission (no directional arrows)
- Star, ring and bus topographies may also apply
- Figures 1, 2 & 5 are common to video transport on analog/fiber or digital/fiber transmission systems
- Figure 3 shows an analog/fiber multichannel multipoint system example
- Figure 4 shows an digital/fiber multichannel/multipoint example
- Digital Cross Connect Systems (DCS) currently switch at DS-0- DS-1, DS-2, DS-3 (Figure 4)

A simple end to end circuit design is shown in ***Figure 1***, where a television broadcast signal source is transported from a venue site through a telephone company central office Television Operating Center to the destination of a Television Broadcasting Studio. Multichannel distribution is possible between locations below.

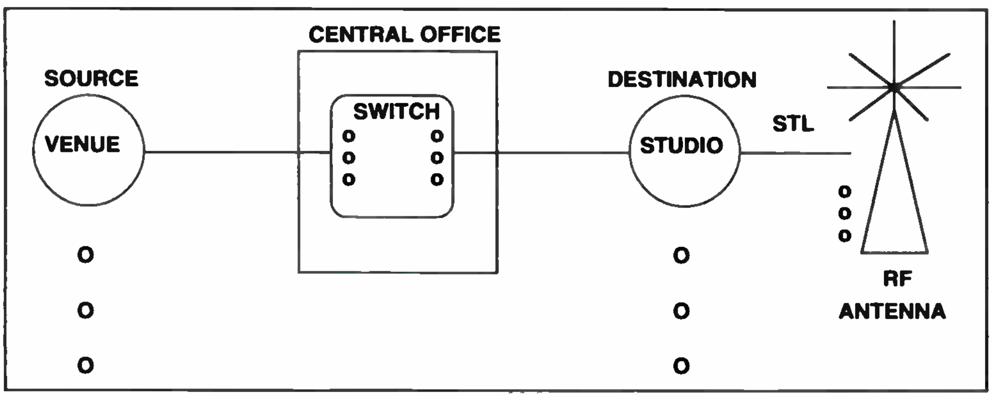

Figure 1

In ***Figure 2***, a more complex circuit design is shown where several source signals, i.e. local television stations arrive at a Headend location and are transported on a multichannel service through a telephone company Central Office Television Operating Center to one or several cable Distribution Hub locations.

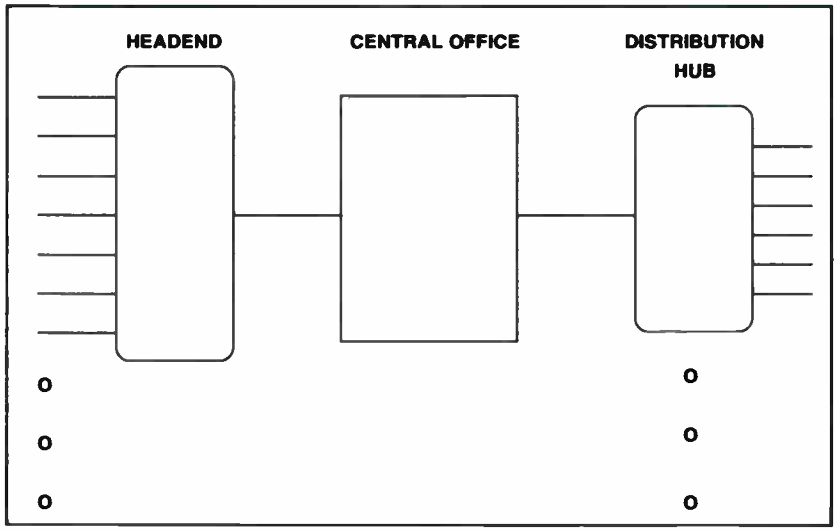

Figure 2

Figure 3 shows a television signal(s) or other video signal(s) originating from a Television Studio and transported through a telephone company Central Office, bridging the signal(s) to three example site destinations via *analog fiber systems*. The baseband video and audio source signals are pulse frequency modulated on to fiber facilities and sent to the telephone company Central Office where it is bridged to three separate fiber optic terminals feeding each destination site.

Video Modulation [includes audio sub carrier] (VM) within the FM/FDM system takes the baseband video and audio signals to Intermediate Frequency (IF) at 70 MHz and combines each channel (typically 1-16 channels) into a single broad band channel. This broad band channel is then converted to laser pulses by the associated fiber *optic terminal (FOT). At the destination sites the reverse transmission sequencing occurs to include video demodulation (VD). The baseband video and audio signals are then "Handed-off" to end users.*

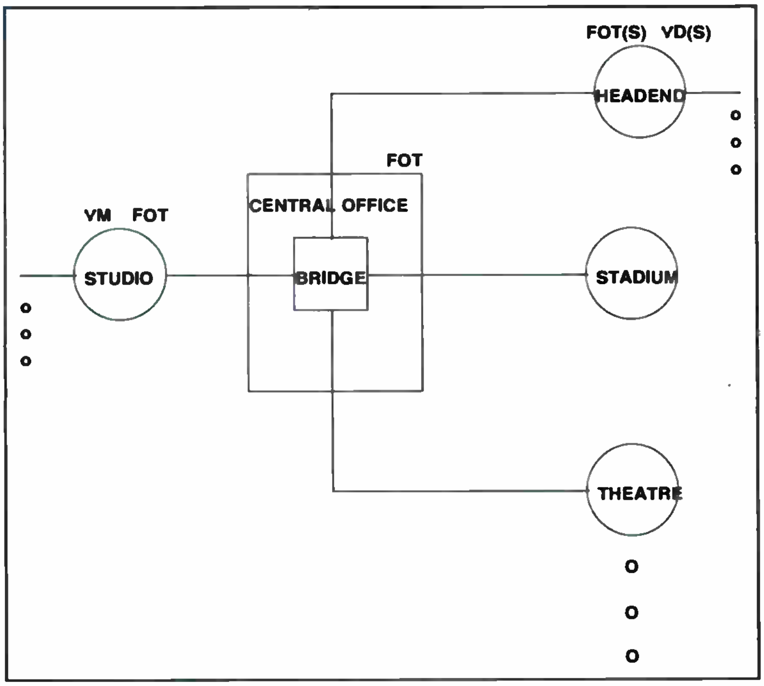

Figure 3

***Please note:** As technology advances, a higher quality of video at lower and lower bit rates is expected; e.g. MPEG 2 models provide higher compression algorithms for broadcast and high definition television. Also larger numbers of video channels may warrant more dynamic digital broad band systems, to include multichannel Fiber Optic Terminals, and in the near future Asynchronous Transfer Mode (ATM) platforms with Add/Drop [site to site] capabilities. Today, the evolution of digital broad band transport service points to ATM and SONET.*

In ***Figure 4***, television transported on a digital system is shown below. A single source signal is A/D and D/A converted by encoders and decoders (CODECs) originated at a studio and transported via DS-3 to a telephone company Central Office. In this example, each video signal is switched at the Central Office through a Digital Cross Connect System (DCS) and controlled by a Flexcom Linc Bandwidth Management and Reservation System. (A DS-3 routing or multicast switch could also be used in the place of DCS, for instantaneous switching).

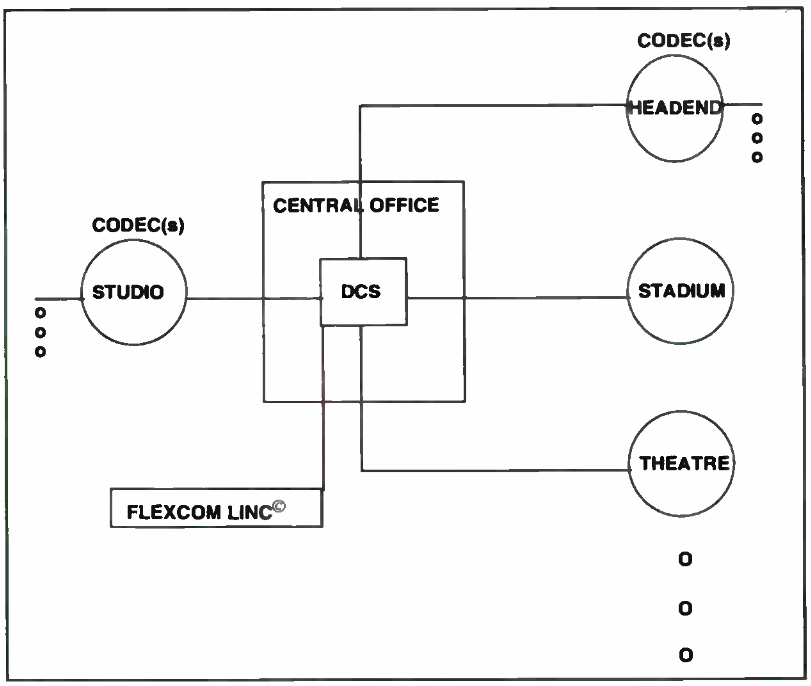

Figure 4

In *Figure 5*, several television studio signal source examples are being transported through a telephone company Central Office to a cable operator's Headend. The drawing is representative of analog or digital systems that can be provided in single or multichannel arrangements. The broad band "Pipe" (or "Supertrunk") between the Central Office and the Headend may be possibly cost shared by the broadcasters.

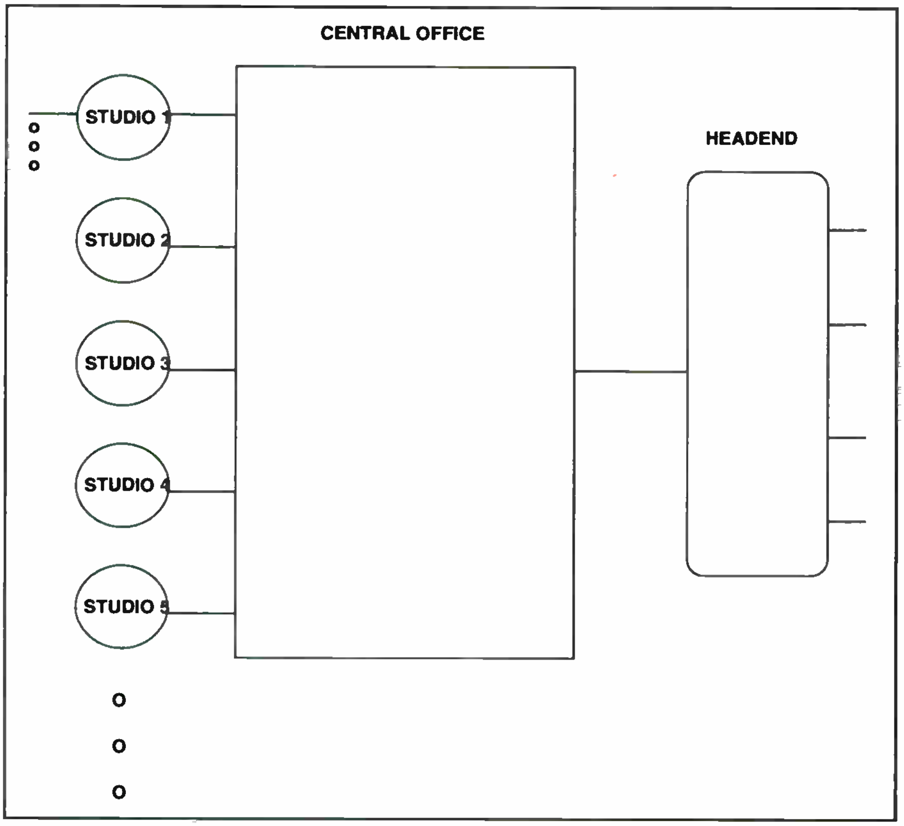

Figure 5

CONCLUSION

Although not specifically depicted, *uncompressed* digital systems for video transport is another multichannel broadcast quality service option. Some systems will provide up to 16 channels of video at 135.7Mbps, with associated audio channels multiplexed at 209 Mbps. This example is a total system provided at 2.4 Gigabit/Second, with channel routing capability.

Video applications transported over analog and/or digital systems provided by the Regional Bell Operating Companies (RBOCs) may include, video on demand (Video Dial Tone), pay per view, movies on demand, post production editing, boardroom conferences, desk top video conferencing, interactive video games, target advertisement, home shopping, medical imagery, court arraignment, government services, distance education, information retrieval: e.g. newspapers, libraries, stock exchanges, museums and hospitals. The importance of multichannel, multipoint video distribution should become greater and greater as interactive mediums and their potentials are realized. The television broadcasters, cable operators, software and hardware suppliers, and telecommunications industry will all have a stake in this new and exciting field of interaction and multimedia.

Single, multichannel (and multipoint) video transport service offerings, along with the predecessors of voice and data, have successfully merged within the telecommunications industry. Its market aligns with the opportunities of applications and services that are converging within the television, cable and computer industries of today. These industries are all expected to be brought together to some degree, as related products are already starting to find their way into homes and businesses. Multichannel, transport of video signals in the telecommunications exchange is a beginning to an era that provides the potential opportunity of gaining audio visual access to entertainment, business and information institutions and centers everywhere.

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TOWERS AND OTHER TRANSMISSION SUPPORT EQUIPMENT

Monday, March 21, 1994

Moderator:
Gerald Robinson, Hearst Broadcasting, Milwaukee, WI

UNDERSTANDING AND PREVENTING GUYED TOWER FAILURES DUE TO ANCHOR SHAFT CORROSION
Craig Snyder
Sioux Falls Tower Specialists, Inc./Anchor Guard
Sioux Falls, SD

CONTROLLING CORROSION ON BROADCAST TOWERS: A SUBJECT YOU CAN'T AFFORD TO BE RUSTY ON
J. Cabot Goudy
Structural Systems Technology, Inc.
McClean, VA

***REPLACEMENT OF A BROADCAST ANTENNA BY USE OF A SKYCRANE HELICOPTER: SOMETIMES IT'S THE ONLY WAY TO FLY**
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Structural Systems Technology, Inc.
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***INSPECTION, MAINTENANCE, AND TROUBLESHOOTING OF UHF/VHF/FM ANTENNAS, TRANSMISSION LINES AND WAVEGUIDES**
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Voorhees, NJ

PERFORMANCE OF A TRANSMISSION LINE HAVING A RIGID OUTER CONDUCTOR AND A CORRUGATED INNER CONDUCTOR
Ronald Tellas
Andrew Corporation
Orland Park, IL

*Papers not available at the time of publication

UNDERSTANDING AND PREVENTING GUYED TOWER FAILURES DUE TO ANCHOR SHAFT CORROSION

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Abstract

Towers have been used to support broadcast antennas since the early 1950's with very little attention given to corrosion of buried tower components. Many broadcast tower facilities are coming of age and the problem of anchor shaft corrosion is now becoming an industry issue. This paper analyzes the causes of anchor shaft corrosion and gives means of protecting existing and future towers against the catastrophic effects of corrosion.

HISTORY

On September 26, 1990, a 350' microwave tower fell to the ground. Tragically, two tower service personnel were on the tower at the time of its collapse. Both men were seriously injured and required multiple surgeries and much time to recovery from their injuries. The cause of the failure was discovered to be due to excessive corrosion on one of the three steel anchor shafts that supported the structure. Although other tower failures have occurred due to anchor shaft corrosion, the seriousness of this failure involving personal injury has prompted wide-spread industry attention.

The Electronic Industries Association committee responsible for writing the standard entitled Structural Standards for Steel Antenna Towers and Antenna Supporting Structures (EIA/TIA-222-E), has been engaged in studying this issue and changes to the standard in this regard appear imminent. The Canadian Standard Association's standard entitled Antenna, Towers, and Antenna-Supporting Structures (CAN/CSA-S37-M86) currently contains an appendix that outlines the problem and lists possible solutions. It is clear that as more and more data becomes available action to curtail or mitigate corrosion on anchors will be an important part of maintaining the integrity and long life of guyed tower facilities.

WHAT IS CORROSION?

Corrosion Fundamentals
Corrosion is an electrochemical process. It is essentially the tendency of a refined metal to return to its native state. There are certain conditions which must exist before a corrosion cell can function. Figure 1 illustrates the four essential elements of a corrosion cell. Necessary elements include the anode, the cathode, an electrical path between each, and an electrically conductive electrolyte.

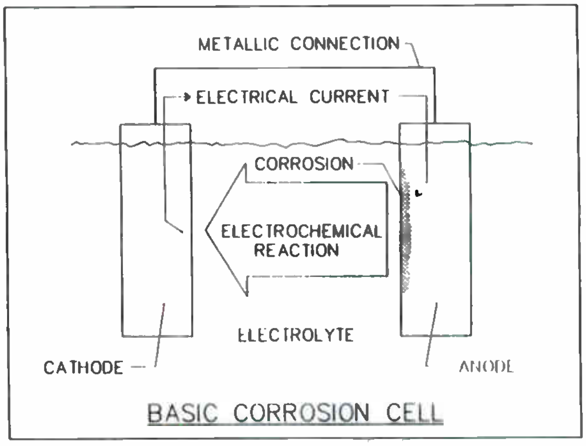

Figure 1 - Basic Corrosion Cell

The driving force behind the corrosion cell is a potential or voltage difference between the anode and cathode. Once each of the four conditions have been met, an active corrosion cell is set in place. When functioning properly, there will be a measurable DC voltage which can be read in the metallic path between the anode and the cathode. When the two are electrically bonded, the anode is positively charged and the cathode is negatively charged. Conventional current flows from positive to negative and thus current discharges from the anode and is picked up at the cathode through the electrolyte. The current then returns from the cathode to the anode through the electrical path. This flow has a detrimental effect on the anode known as **corrosion.**.

The above stated example is a corrosion cell in its simplest form. It is important to know that each of the four elements of the corrosion cell will effect how severe or mild the effects of corrosion are. We will begin by discussing the four elements of a corrosion cell and how they interact with one another.

Dissimilar Metals and the Galvanic Series

In a corrosion cell, the anode and the cathode will typically be composed of dissimilar metals. Each different metal finds its place in the GALVANIC SERIES. In figure 2, we see where several more commonly used metals are located in this series. The metals placement in the series is a function of the electrical relationship or POTENTIAL the metals have to one another.

Magnesium	-1.75
Magnesium Alloy	-1.60
Zinc	-1.10
Aluminum Alloy	-1.05
Commercially Pure Alum	-0.80
Mild steel (clean & shiny)	-0.50 to -0.80
Mild steel (rusted)	-0.20 to -0.50
Cast Iron	-0.50
Lead	-0.50
Mild steel in concrete	-0.20
Copper, brass, bronze	-0.20
High silicon cast iron	-0.20
Mill scale on steel	-0.20
Carbon, graphite, coke	+0.30

Figure 2 - The Galvanic Series

If we were to choose two metals from the scale, electrically bond them and immerse them in an electrolyte, we would find that the voltage produced would equal the differences in the two metals potentials. For example, if we use the mild steel and copper as our anode and cathode, the voltage measured between the two will be approximately -.6 volts.

MILD STEEL	-.8
COPPER	- -.2
	-.6

In corrosion terms, the metal higher on the scale is the anode and the metal lower on the scale is the cathode. In this example, the mild steel is anodic to the copper and therefore will corrode - provided that the other two conditions of the corrosion cell are met.

There is also a direct relationship between the sizes of the anode and the cathode as to the severity of the corrosion cell. If the area of the cathode is very large in relationship to that of the anode, the corrosion cell will be more severe, and thus the faster the anode will deteriorate. On the other hand, if the anode is very large in relationship to the cathode the effects of corrosion are much less and the anode deterioration is more gradual. Figure 3 illustrates this relationship as it might be seen on a typical tower anchor support.

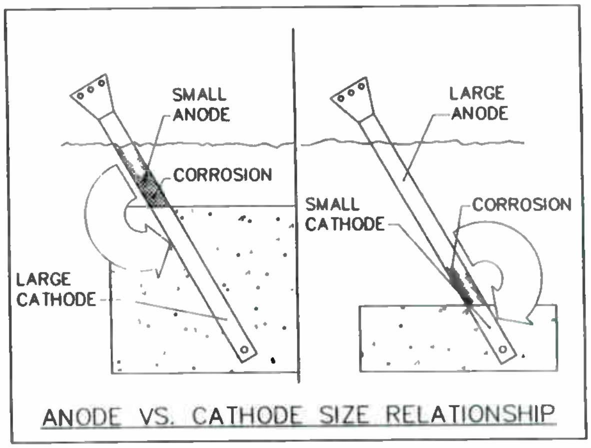

Figure 3 - Anode vs Cathode Size Relationship

Effects of Electrolytes on the Corrosion Cell

Many different environments could be considered electrolytic. Tower anchor supports are most commonly located in soil and concrete. This paper will confine its study of electrolytes to only these two.

Each type of soil has a measurable resistivity. Soil resistivity can be measured by using a soil resistivity meter. This measurement is defined in ohm-centimeters (ohm-cm). The method of measuring soil resistivity used most often is the four-pin method. Figure 4 illustrates the soil resistivity meter as it would be set up using the four-pin method.

A lower resistivity measurement typifies a more conducive environment to the flow of current. The absence of oxygen in the soil also contributes to the enhancement of the corrosion cell. For example, clay type soil in a wet climate may measure 1000 ohm-cm and therefore be a very good electrolyte. Sandy or rocky soil in a dry climate may measure 20,000 ohm-cm and therefore be a very poor electrolyte. The resistivity of the soil is a very important factor in evaluating the variables of the corrosion cell. When all variables of a corrosion cell remain constant, the electrolyte resistivity becomes the determining factor in the design and application of corrosion control measures.

It is also important to know that soils vary drastically from place to place across the globe. Soil type can even vary within inches! The variability of soil can cause multiple corrosion cells on the same structure. Figure 5 illustrates how various soil types can create a corrosion cell on a tower anchor shaft. In this illustration, we find the upper soil layer a looser, somewhat gravely soil and below that a more dense clay type soil. The portion of the shaft in contact with the clay type soil acts as an anode to the portion of the shaft in contact with the looser gravely soil, which is consequently the cathode. Again, we have a corrosion cell where the shaft deteriorates in the anodic areas.

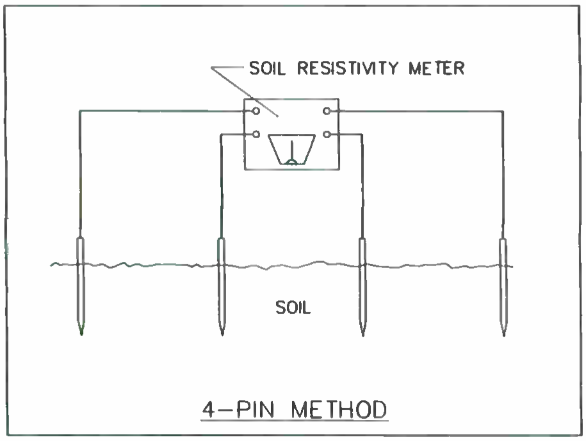

Figure 4 - Soil Resistivity by the Four Pin Method

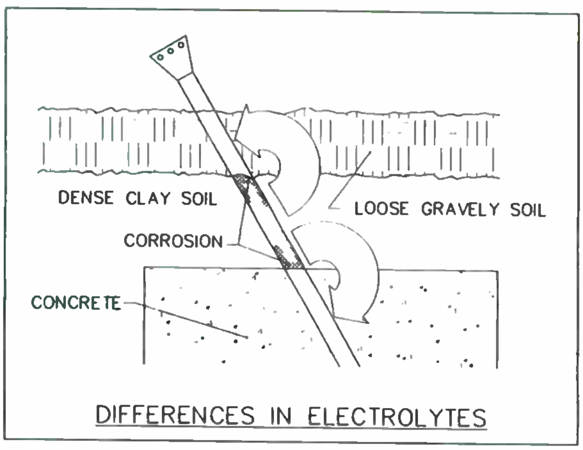

Figure 5 - Differences in Electrolytes

When considering the placement of a new tower site, the geotechnical soil investigation should include a determination of soil type including soil resistivity and chloride and sulfate ion presence. These items are critical to the design of corrosion control measures. If a geotechnical study is not required, soil samples can be taken and analyzed separately by a competent corrosion control firm.

HOW DOES THE CORROSION CELL AFFECT ANCHOR SUPPORTS?

For many years, the tower industry has used a basic anchor design similar to the one shown in Figure 6. This design presents a large number of benefits especially in its use of the soil as a means to contain the anchorage. However, it is now becoming increasingly evident that the problem of corrosion on the anchor has not been sufficiently addressed in this design. In this section, we will discuss anchor support designs as they relate to the corrosion cell.

Galvanic Corrosion

Example 1. Figure 6 shows the basic design of a typical anchor support. The anchor support has all the necessary elements of a corrosion cell. The shaft itself acts as both anode and cathode as well as the electrical path between the two. The concrete and soil act as dissimilar electrolytes. The soil has less oxygen just above the concrete anchorage and consequently, less resistivity. The electrical path is the shortest between the anode and cathode at the same point as the oxygen deficient soil, thus the deteriorating action of corrosion is most strongly in effect in this area. Experience has shown that this is the area most likely to deteriorate and cause the tower to fail.

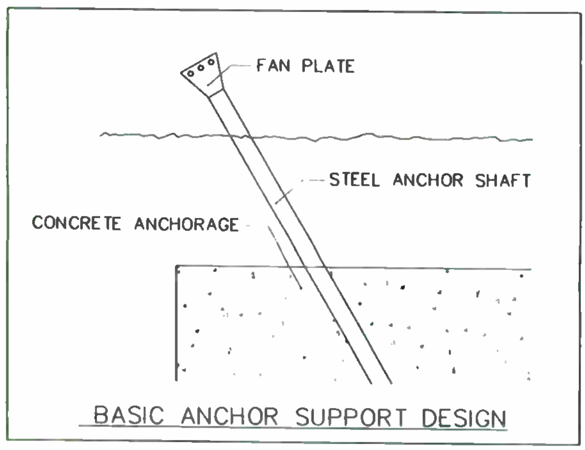

Figure 6 - Basic Anchor Support Design

Example 2. The second example shown in Figure 7 has the same basic anchor support design, however now a copper grounding rod is incorporated as a means of lightning protection. The electromotive force series notes that copper is lower on the scale than steel and when coupled with steel will produce the measurable current previously spoken of. The two dissimilar metals activate a corrosion cell in which the steel anchor shaft is the anode and the copper ground rod is the cathode. Depending on the resistivity of the soil, this combination can contribute to rapid deterioration of the anchor shaft.

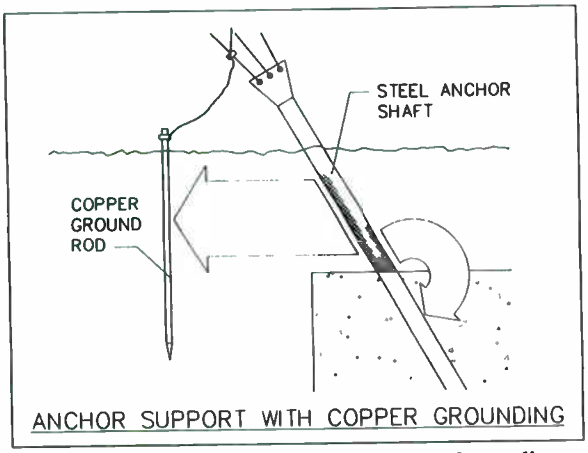

Figure 7 - Anchor Support With Copper Grounding

Experience has shown that by using galvanized grounding rods for lightning protection, this particular corrosion cell can be avoided. However changing ground rod type alone does not preclude the possibility of the existence of other corrosion cells, such as those mentioned in example 1.

Stray Current Corrosion

The third common example of anchor support corrosion is known as "stray current" corrosion. Figure 8 illustrates how stray current can adversely effect the buried components of a tower facility.

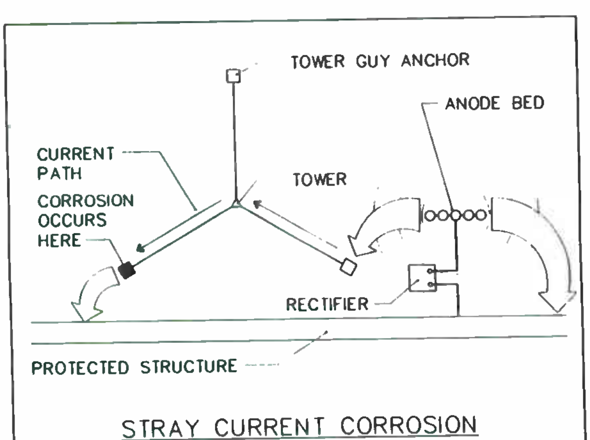

Figure 8 - Stray Current Corrosion

Many buried structures such as petroleum pipelines are protected against corrosion by means of impressed current cathodic protection systems. (See the following section for a more detailed explanation of Cathodic Protection.) Electrified railways and welding or plating operations will also put direct current into the ground. If a tower is near any one of these, it could be exposed to stray current corrosion. In Figure 8, the tower is used as a path of least resistance for current flow through the electrolyte. Stray current is picked up on one anchor support and travels by means of the guy cables to the alternate anchor where it subsequently discharges. The structure rapidly deteriorates at the place where the current discharges. This problem can be alleviated by electrically bonding the tower to the structure generating the stray current, essentially making it part of the protected structure.

HOW CAN CORROSION BE MITIGATED ON ANCHORS?

There are several options available to curtail, if not wholly eliminate, the problems associated with corrosion on anchor supports. In the following section, we will discuss these options. When designing a tower corrosion control system, all of the following examples should be considered. More than one of the options will be required in most instances.

Galvanization and Epoxy Coatings

The first example of corrosion control and the one currently used most often is that of coating the buried steel shaft. Usually anchor shafts are hot dipped galvanized with a zinc coating. This is advantageous as the zinc acts as an anode to the steel shaft. However, if galvanizing is used by itself as corrosion control the zinc can rapidly deteriorate leaving the exposed steel shaft open to damage. Hot dipped galvanizing in many cases gives a false sense of security because of the satisfactory appearance of the galvanizing above grade.

Epoxy coatings are also often used to protect shafts. This is accomplished by coating the entire shaft with an epoxy that acts as an insulator to protect the steel shaft from direct exposure to the electrolyte. This method is beneficial in protecting against corrosion, however it has been proven that even the best epoxy coatings cannot guarantee 100% isolation from current. In addition, the coating can be damaged during shipping or installation, leaving small anomalies or "holidays". If the shaft is then buried with these "holidays", the "big cathode, small anode" scenario spoken of earlier comes into play. The shaft is open to accelerated corrosion in small areas that rapidly become larger. This type of concentrated deterioration is worse than if the shaft were left to corrode on its entire surface more evenly.

Concrete Encasement

Another option to prevent anchor shaft corrosion can be seen in Figure 9. In this option, the entire anchor shaft is encased in concrete. This method of anchor design is sometimes used for its structural capacity but it also has its corrosion control advantages. The greatest advantage is that it all but eliminates one essential element of the corrosion cell. The concrete has such a high resistivity that even with copper grounding electrically connected, current flow is substantially impeded. In addition, if the entire shaft is encased there is no anode/cathode relationship on the shaft itself.

The disadvantage to this alternative is the possibility that the concrete could become cracked or broken. If this should happen and water or soil fills the cracks, a corrosion cell would be created. The anode would be the area inside the cracks exposed to the water or soil, and the cathode would be the portion of the shaft inside the uncracked concrete.

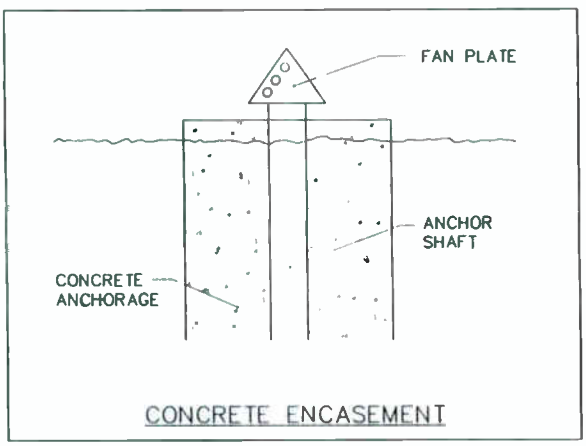

Figure 9 - Concrete Encasement

Cathodic Protection

Cathodic protection is a process of using the known variables of a corrosion cell to effectively mitigate the detrimental effects of corrosion. There are two types of cathodic protection commonly used. The first is known as galvanic anode and the second is impressed current. We will discuss each.

Galvanic Anode. Figure 10 illustrates the application of galvanic anode cathodic protection to a typical anchor support. The anchor support without cathodic protection installed is anodic to the copper grounding system and the portion of the shaft inside the concrete. By electrically bonding sacrificial anodes to the anchor support, the current flows away from the sacrificial anode and toward the anchor shaft and copper ground rod. It then returns through the electrical path. In this way, the elements of the corrosion cell have been used to make the anchor shaft the cathode, thereby eliminating corrosion where corrosion is not wanted.

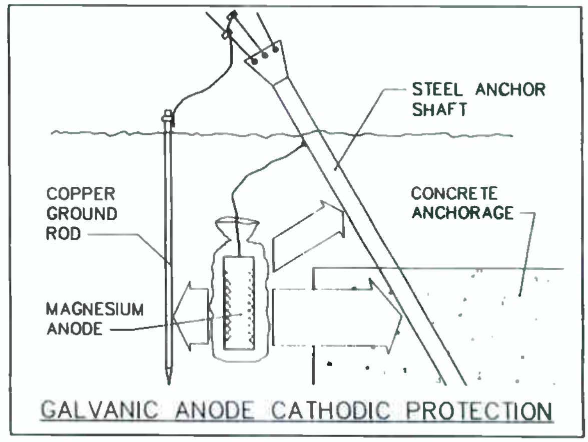

Figure 10 - Galvanic Anode Cathodic Protection

Sacrificial anodes vary widely in their sizes, shapes and make-up. Anodes are typically made of magnesium or zinc. The anode is usually placed in a cotton bag surrounded by a gypsum, bentonite and sodium sulfate mixture. This mixture is used for the purpose of assisting in the activation of the current flow and to ensure that moisture remains around the area of the anode. A wire is attached to the inner core of the anode and is designed to be bonded electrically to the member to be protected.

Following installation of the galvanic anode cathodic protection system, it is essential that it be monitored regularly to ensure its proper operation. A DC volt meter and copper/copper sulfate reference electrode (half-cell) is the most common method of checking the system after its installation. The tip of the half cell is placed in the soil with one lead of the volt meter connected to it and the other to the structure being tested. The measurement should show a voltage shift from the same test conducted on the structure before the system installation.

Impressed Current. Impressed current cathodic protection also uses buried anodes, but in a somewhat different fashion then the galvanic anode system. Rather then relying on the electromotive force of magnesium, DC voltage is impressed on the anodes by means of a voltage rectifier. The rectifier supplies ample current to the anodes to allow current to flow through the electrolyte and toward the protected structure. An electrical connection is made from the rectifier to the structure in which the return current flows. Figure 11 illustrates a typical application of this system at a tower site.

An advantage of impressed current cathodic protection is that the entire system can be centrally located near the base of the tower. The tower and it's guy cables are the electrical connections to the anchor supports - provided the guy cables do not have cable insulators. The anodes are placed strategically in the same central location. The rectifier is then mounted on the transmitter building or anywhere AC power is readily accessed.

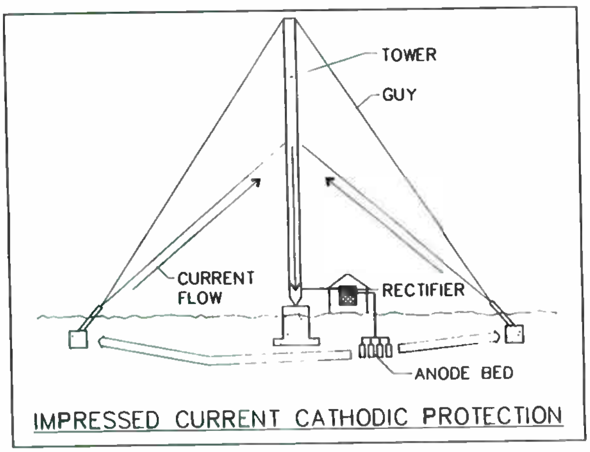

Figure 11 - Impressed Current Cathodic Protection

As with the galvanic anode system, the impressed current system also requires maintenance. It is recommended that the rectifier be inspected regularly and potential measurements be taken in the same fashion as explained in the Galvanic Anode example. This system allows for adjustment in current through changing the rectifier output.

Electrical Isolation

The final example of corrosion control is electrical isolation. Figure 12 shows an anchor support that has been electrically isolated from stray current or galvanic corrosion associated with dissimilar metals. This type of system is commonly used on AM radio towers to isolate the transmitting structure from the ground.

Electrical isolation employs guy wire strain insulators to eliminate the electrical path between the tower and the anchor support. It is an effective way of eliminating the galvanic cell problem associated with copper grounding systems, while at the same time taking advantage of the properties of copper as a means of superior lightning protection. In addition it protects against stray current corrosion. If applied correctly, the lead from the ground rod is attached to the guy cable on the "tower side" of the insulator so as to isolate the ground rod from the anchor shaft. This will ensure lightning protection and at the same time eliminate the copper-steel galvanic relationship. It also breaks the electrical connection required for stray current corrosion to take effect.

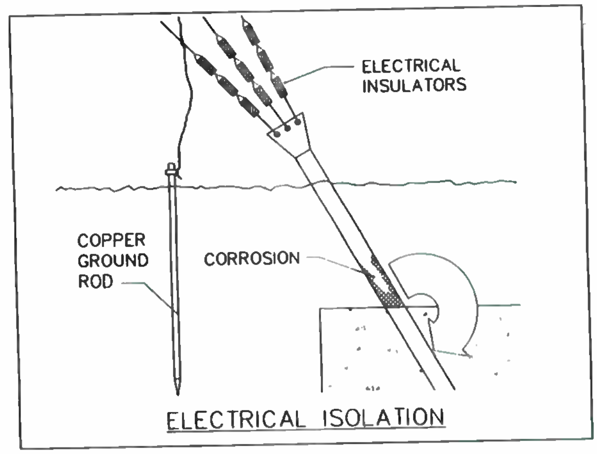

Figure 12 - Electrical Isolation

CORROSION ON EXISTING STRUCTURES

To this point we have discussed what corrosion is, how it works in the example of tower anchors, and we have given several methods for its control. The question now becomes what about existing facilities that may or may not have already sustained structural damage due to corrosion. This section will outline methods for investigating this potentially serious problem and give methods of remedial action that can be taken.

Anchor Inspection

It is becoming increasingly clear that many guyed towers have already sustained anchor corrosion. This has placed the structural capacity of the tower in question. Typical inspection procedures in the past have been to dig around the anchor to an approximate depth of twelve inches below grade. In many cases corrosion is found during such an inspection but is often determined to be acceptable. Based upon the principals of corrosion previously discussed, this method of detection is likely insufficient. The most extensive corrosion will usually occur closest the concrete anchorage which is often several feet below the surface of the soil. If any amount of corrosion exists at the upper levels of the anchor shaft, further investigation is usually warranted.

The most proven method of anchor investigation is to fully expose the steel shaft portion of the tower anchorage. This involves carefully removing the soil around the shaft to the depth of the concrete footing. Care should be taken so that the concrete footing is not overly exposed which could result in the failure of the anhcorage and the tower. If structural capacity of the anchor is in question, temporary anchorages may be advisable.

Once the soil has been removed from around the anchor shaft, the shaft should then be cleaned so as to allow measurement of the steel member. Typically sand blasting or some other acceptable method of cleaning the shaft to a bare metal condition will be required. It is important that the shaft be free of soil and corrosion scale so that accurate measurements can be taken. It would not be uncommon for the shaft to appear to be satisfactory based on appearance prior to cleaning. This is true in part because of the thick corrosion scale build-up that clings to the face of the steel giving the false impression that the shaft size or dimensions have not been altered substantially by corrosion.

After cleaning the shaft it can then be measured by means of micrometer or ultra-sonic thickness tester and reference given to the portion of the shaft above grade that has likely not sustained corrosion damage. Repair or replacement may be required, if corrosion has left question as to structural capacity of the shaft.

Applying Protective Measures

Epoxy Coating. If the shaft is determined to be structurally sound, then the bare metal anchor shaft provides an excellent surface for application of epoxy coating. Bituminous epoxy is the most commonly used and best alternative for coating of buried steel. Care should be taken to ensure complete saturation of the entire surface of the shaft. If anomalies or "holidays" are left in the coating, a corrosion cell could be even more severe then if the shaft were left alone.

Cathodic Protection. Once the shaft has been coated properly, additional measures such as cathodic protection should be applied. A properly coated member will allow substantial reduction in the amount of cathodic protection needed. A qualified Corrosion Engineer or Engineering firm may need to be retained to determine the type and amount of cathodic protection required. Research in the field of cathodic protection for tower anchors continues to proceed and viable products for the protection of anchors are forth coming. For more information contact the author of this paper.

SUMMARY

Corrosion of guyed tower anchors is fast becoming an issue of critical importance. Towers designed and built without consideration to corrosion control are in jeopardy. A towers age is a key factor when considering the potential for structural damage due to corrosion, but age alone cannot predict the extent of corrosion damage. Towers scheduled to be built should include corrosion control measures. Existing towers should be investigated for damage and then protected against corrosion. Several methods exist to mitigate corrosion of buried tower components.

Just as construction of each tower requires careful consideration of structural design, each tower site must be assessed to determine appropriate corrosion control measures to be taken. The long-term benefits of proper corrosion control will far exceed short-term cost savings of choosing none. As technology and awareness increase, appropriate corrosion control is becoming an important industry standard.

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CONTROLLING CORROSION ON BROADCAST TOWERS: A SUBJECT YOU CAN'T AFFORD TO BE RUSTY ON

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ABSTRACT: A summary of the classical preemptive treatments to control corrosion on broadcast towers is presented. Treatments examined include hot dip galvanizing, zinc spraying, painting and various combinations of the foregoing. Pros and cons of the various treatments are examined as well as the expected life span of each. Remedial solutions to corroded structures are presented.

The protection of structural steel from environmentally induced corrosion is a topic of considerable controversy among both laymen and professionals. The subject is complicated by a myriad of products and treatments available to the market as well as the specific application of those treatments to the end use. This paper will investigate various corrosion protection systems utilized in structural steel broadcast towers and attempt to enhance the readers understanding of how to select and maintain a corrosion protection system.

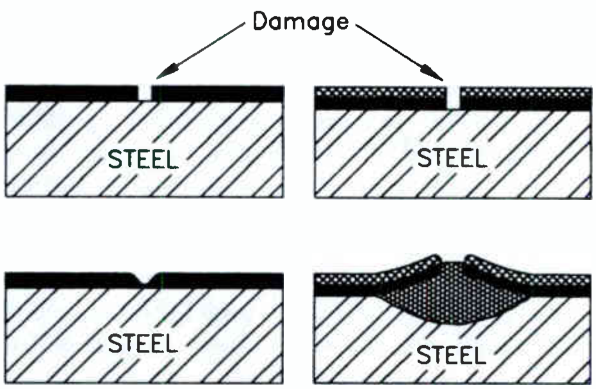
METHODS OF PROTECTION:

Most broadcast towers in service are protected from corrosion by one of the following systems; painting, hot dip galvanizing, zinc spraying or some combination of the foregoing. The painting of a broadcast tower, as its sole means of protection from corrosion is considered the least effective means of all systems. This is due to the fact that the mechanism of protection of the basic steel structure relies primarily upon the ability of the paint to form a protective barrier to the steel and seal it off from moisture and other corrosive agents. In the all too likely event of a breach of the protective barrier, such as chips and scratches resulting from shipping, handling, and erection rigging operations, the steel will begin to corrode. Certain paints contain zinc which will afford some additional cathodic protection to the base steel; however, few structures built after 1960 have utilized the paint only system and therefore, further discussion of this topic will be deferred to the section of this paper which addresses remedial solutions to in-service structures.

Virtually all contemporary broadcast structures employ hot dip galvanizing, zinc spraying, or a combination of one of these plus paint to protect the steel. Hot dip galvanizing is the most widely used method of protecting steel broadcast towers from corrosion. Hot dip galvanizing is the process of cleaning the steel free of all mill scale or other impurities and then dipping the steel into a molten zinc bath at a temperature of about 850 degrees fahrenheit. Upon removal of the steel from the zinc bath, the remaining zinc forms a thick, tough, abrasion resistant coating over the base metal. This coating completely covers the entire unit and is metallurgically bonded to it.

The galvanized coating affords corrosion protection to the base metal three ways. First, it acts as an effective barrier to prevent the penetration of water, salts, or other corrosive agents; second, the zinc itself sacrificially dissolves to provide cathodic protection to any exposed base metal; and third, over a period of time, the zinc layer will form a film of zinc carbonate which will seal over any voids or small damaged areas in the zinc coating.

It is the "cathodic" protection afforded by the zinc which primarily differentiates this coating system from the paint only system. When a painted steel is exposed to the environment, perhaps at a scratch in the surface, rust will form at the steel/paint interface. The rust occupies a volume several times that of the base steel and therefore the resulting expansion leads to failure of the paint/steel bond. In addition, the rust is porus and will attract moisture and other reactants which in turn will create additional rust and lead to further failure of the paint system. See Figure 1(b).



(a)
Zinc coating

A galvanic cell is formed. The zinc around the point of damage corrodes. Corrosion products precipitate on the steel surface and protect it. The steel is also protected because it is cathodic in realation to the zinc coating.

(b)
Paint coating

The steel rusts where the paint film has been damaged. Rust creeps under the paint film, which is lifted up from the steel surface. Corrosion continues until the damage has been repaired.

Figure 1. Consequences of damage to different types of rust-preventive coatings. Schematic illustrations.

(SOURCE : Zinc Development Association, "Rust Prevention by Hot Dip Galvanizing," London, 1980, p. 24.

Figure 1(a) illustrates a break in a hot dip galvanized coating. Here the zinc provides "cathodic" protection to the steel because the zinc is more anodic than the steel. Small scratches and cut edges do not rust due to the sacrificial nature of the zinc.

Another important feature of the hot dip galvanized coating of steel is that the zinc is metallurgically bonded to the base metal. This fact differentiates hot dip galvanizing from all other coating methods. The metallurgical bond insures that corrosion will not occur between the zinc and the base metal. Figure 2 illustrates the cross section of the hot dip galvanized coating. The zinc is deposited on the base steel in several layers ranging from 100% zinc at the surface, and then changing through various layers of iron/zinc alloy, to 100% steel at the base. The layers and their respective iron/zinc alloy compositions are shown in the schematic illustration.

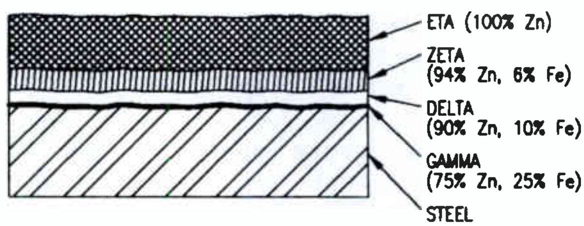


Figure 2. Schematic of Hot Dip Galvanized Coating Cross Section.

(SOURCE : American Hot Dip Galvanizers Association, "Hot Dip Galvanizing For Corrosion Protection of Steel Products," Washington, D.C., 1984, p. 3.

A third method of corrosion protection in use on broadcast towers is the application of zinc through a process called "zinc thermal spraying". This process is similar to hot dip galvanizing only in that a layer of zinc is applied to the steel. The zinc is applied by spraying a stream of molten zinc onto the base metal. The applied coating is mechanically bonded to the steel but no iron/zinc alloy layers are formed as with the metallurgical bond associated with hot dip galvanizing. The adhesion of the molten zinc to the base metal is considered quite good assuming the base metal is properly prepared prior to application. The resultant coating is slightly porus compared to the hot dip galvanized coating; however, the zinc affords the same "cathodic" protection and over time zinc corrosion products will form and act to seal the porus surface. The surface of the zinc sprayed steel is also slightly rough and makes an excellent base for a top coat of paint.

The final method of protection to be discussed is what is sometimes called a "duplex system", or the application of paint over hot dip galvanized or flame sprayed steel. As previously mentioned, the flame sprayed steel produces an excellent base for the application of paint. It is the application of paint over hot dip galvanized steel which is most often troublesome and frequently results in failure due to a loss of adhesion between the paint and the galvanized coating. Hot dip galvanized steel can however be successfully painted with proper surface preparations.

The zinc coating applied to hot dip galvanized steel undergoes certain evolutionary transformations which initiate immediately upon its removal from the molten zinc bath. Upon removal from the zinc bath, the outer ETA Layer (See Figure 2.) which is 100% zinc will begin to oxidize. The early stages of this oxidation will not produce films leading to an adhesion problem for some paints if applied within zero to 48 hours after removal from the galvanizing bath. Beyond 48 hours however, zinc oxides, zinc hydroxides and other various contaminants will be present on the surface and will form films preventing the adhesion of paint to the surface. This condition can be present between 48 hours and 2 years after galvanizing.[3] It is precisely in this time frame that most hot dip galvanized fabricated steel is scheduled for painting and it is for that reason that proper surface preparations must be made if painting is attempted at that time.

The final stage of evolution in the galvanized coating occurs between 8 months to 2 years after galvanizing.[3] In this stage the final film of zinc carbonates forms on the surface. These zinc carbonates are essentially inert and are tightly adhered to the surface. Painting of a hot dip galvanized steel after the final evolutionary stage has occurred and the stable zinc carbonates are present can be achieved with little or no surface preparation.

As previously stated, most hot dip galvanized steel towers, if they are going to be shop painted, will receive the paint during the time frame in which films of zinc oxides and hydroxides are present. Proper surface preparation is critical to a successful application. Surface preparation can be achieved by lightly sandblasting the material to roughen the surface and to remove the contaminated films. Sand blasting is the most costly system of surface preparation and requires a skilled operator being careful not to remove too much zinc. A next best alternative is the use of a high pressure washing followed by the use of a metal conditioner or "wash primer". Various primers are commercially available and the manufacturer should be consulted for specific recommendations as well as for top coat recommendations.

LIFE EXPECTANCY OF GALVANIZED COATINGS:

The ASTM standard which covers hot dip galvanizing of structural steels is designated A 123. This standard requires an average minimum thickness of zinc to be applied during the hot dip galvanizing process. The average minimum for the standard is between 2.0 and 2.3 Oz./sq.ft. of zinc or approximately 3.4 to 3.9 mils thickness(One ounce of zinc/sq. ft. of surface corresponds to a coating thickness of .0017 in.). Research has shown that the life expectancy of the hot dip galvanized coating is a function of the thickness of zinc applied and the type of environmental exposure which the in-service structure will experience. Figure 3 below illustrates the expected service life for various environmental conditions and zinc coating thicknesses.

Figure 3 is based upon research conducted on hot dip galvanized coatings; however, similar results would be expected from zinc applied through a thermal spraying process if the two applications were compared on an equal thickness basis. In practice, zinc spraying is difficult, if not impossible, to apply in cavities, depressions, corners and similar areas and therefore the zinc coating thickness can be variable over any given fabricated unit. It is for this reason, and due to the fact that the zinc sprayed surface is considered porus, that most applications of zinc spraying are embellished by applying a top coat of paint. The application of the paint creates a "duplex system" previously discussed in this paper.

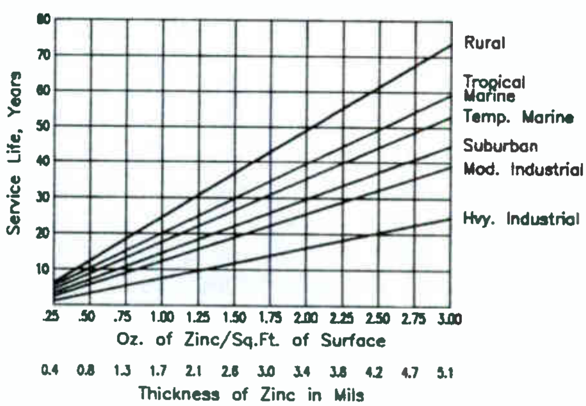


Figure 3. Life of Protection vs. Thickness of Zinc and Type of Atmosphere.

(SOURCE : American Hot Dip Galvanizers Association, "Hot Dip Galvanizing For Corrosion Protection of Steel Products," Washington, D.C., 1984, p. 5.

The most significant advantage to the duplex system, whether on a hot dip galvanized base or on a zinc sprayed base, is what is known as the "synergistic" effect of paint over a galvanized coating. The combined effect of a paint application over zinc is to add a 50% life span extension to the simple addition of the protection provided by the zinc and paint separately. That is to say, if a particular zinc coating has a life expectancy of 20 years and a particular paint system has that of 10 years, then the duplex system can be expected to last 45 years. Some research indicates that the 50% increase is a conservative value for the estimate of the synergistic effect.

One final comment on the life expectancy of hot dip galvanized coatings is in order. Many contemporary hot dip galvanized, non painted, broadcast towers have been reported to be "rusting" after only a few years in service. In almost all cases the so called "rust" is a yellow to orange-red discoloration of the galvanized steel and is not in fact rusting of the base metal. This phenomenon is called "alloy layer staining" and is characteristic of silicon killed steels which have been hot dip galvanized. The silicon in the steel has an influence on the formation of the iron/zinc alloy layer as shown in Figure 2. These particular steels will form a relatively thin top layer of pure zinc. The result of this phenomenon is that in some cases the outer layer of pure zinc will be expended in a short period of time thereby exposing the iron/zinc alloy layer. As the iron in the alloy layer is liberated and forms rust, the characteristic discoloration is evidenced. It should be noted that a significant amount of galvanic protection remains for the base metal after the first appearance of this discoloration.

REMEDIAL TREATMENTS FOR IN-SERVICE STRUCTURES:

Corrosion protection for an in-service broadcast structure relies upon the quality and integrity of the originally applied system. Once that system begins to fail, it must be corrected or enhanced by a field installed paint application. Field applications of paint fall in three basic categories as discussed below. Specific paint applications by brand name are withheld intentionally and manufacturers should be consulted for recommendations with respect to the generic paints and treatments as discussed.

The first corrosion protection system discussed in this paper was that of paint applied directly to steel. Although this system is rarely used on contemporary structures, there are many older structures still in service with paint alone providing corrosion protection. The most important factor to the maintenance of these towers is routine inspection and correction of paint failure and rusting. Rusted areas should be scraped clean to bare metal, painted with an organic zinc-rich primer and finally top coated. By performing routine touch up painting of this nature, severe remedial action can be avoided. In the extreme case however, a badly neglected paint system may require the entire tower to be sand blasted to bare metal followed by a zinc-rich primer and top coat.

The second category of field maintenance concern is that of the non-painted hot dip galvanized structure. As stated earlier, many hot dip galvanized structures can exhibit the "alloy layer staining" phenomenon after only a few years in service. This has caused many towers to be painted prematurely as the staining is often mistaken for base metal rusting. Generally there will remain many years of protection beyond the first appearance of the staining. When painting is however decided upon, either for aesthetic reasons or because sufficient depletion of the zinc has occurred, the weathered galvanized steel provides an excellent, stable, base for the paint application. The surface should be thoroughly rinsed with a high pressure water blast, primed and top coated. Some paint manufacturers will recommend the application of an inorganic zinc primer to severely weathered galvanized steel; however, many experts consider that almost any good quality paint, properly applied, will adhere well to the weathered surface and provide the necessary protection assuming that there is still some zinc remaining to provide cathodic protection to the base steel. Severely rusted areas, if present, must be cleaned to bare metal and primed with a zinc-rich paint.

Maintenance of the third category of corrosion protection system, that is paint over galvanizing, can range from minimal to extensive. As a minimum, touch up paint will be required after erection. If the tower is not strobe lighted, a color coating must be applied on a routine basis to meet FAA requirements as the paint color fades. In the extreme, the premature failure of a paint over galvanizing system will require substantial field maintenance work. As previously discussed, the most common failure of the paint over galvanizing system is caused by a lack of adhesion of the paint to the galvanizing. A failure of this nature will require extensive surface preparation to remove all loose paint using either water or sand blasting of the steel. Severely rusted areas will require heavy scraping to bare metal and priming with zinc-rich paint. In the worst case the entire tower may require a primer and top coat.

One final area of consideration is necessary in a discussion of corrosion protection maintenance for in-service structures. Any recommendations set forth by painting manufacturers or painting consultants must consider the realities and difficulties of working on an environmentally exposed tall steel structure, as much as 2000 ft. above ground level. These realities impose inherent complications which must be accounted for if one wishes to achieve practical solutions. Some of these considerations are listed below:

1) Sand blasting may not be an option. Many towers are located in urban environments which would require complete recovery of the abrasive particles. Power tool scraping or water blasting may

be required as alternates.
2) There is limited control of the environment; in particular, humidity. Many paints are not recommended to be applied in high humidity. This can become a very serious consideration to applications in locations such as Jacksonville, Florida and Houston, Texas.
3) The use of two part epoxy paints or "hot" paints have a very limited pot life. These paints perform well in laboratory tests and when properly applied under controlled shop conditions; however, their successful use in the field is complicated by their limited pot life and temperature sensitivity. It is exceedingly difficult to properly apply these paints to tall broadcast towers in the field.
4) Consideration should be given to the use of water based acrylic paints for top coating. The quality of these paints has substantially improved, they are durable, easy to handle and difficult to misapply.

CONCLUSION:

The above discussion was presented as a general guide to some of the problems and solutions associated with controlling corrosion on broadcast towers. If the problems are ignored, the consequences can be extremely costly. If the solutions are misdirected or misapplied the consequences can be even more costly. Careful consideration of the issues presented will avoid a great deal of future concern.

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PERFORMANCE OF A TRANSMISSION LINE HAVING A RIGID OUTER CONDUCTOR AND A CORRUGATED INNER CONDUCTOR

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ABSTRACT

A new type of rigid transmission line has been developed which overcomes the problems associated with sliding contact joints. This new transmission line is a hybrid coaxial line with corrugated inner conductor and rigid outer conductor. The performance of this line shows much promise in its ability to be used as a replacement to the traditional rigid line and to work in conjunction with existing transmission line components.

INTRODUCTION

Traditional rigid line relies on the outward insertion force of a spring finger contact to provide the mechanical and electrical connection of the inner conductor. Means must also be provided to compensate for thermal expansion. Over very long periods of time, loss of contact pressure may occur, and some thermal expansion compensation techniques rely on a sliding contact. When the required electrical contact is not met, or a sliding contact causes metal debris to form, arcing can occur and failure of the line is imminent. The benefit of using rigid line is its electrical performance and ease of replacing a damaged section with a new section.

Eliminating the spring finger contact, or bullet, entirely is possible by using a coaxial line with continuously corrugated inner and outer conductors such as Andrew's Heliax cable. The fact that the inner and outer conductors are continuously corrugated provides a natural means to compensate for thermal expansion. Eliminating the bullet in this way is effective. However, for equal line sizes, the system will be somewhat less efficient because of the slightly increased attenuation. There may also be some inconvenience getting the large reels to site, although Heliax cables are easier and quicker to install than rigid lines. Repair generally requires splicing in a significant length of cable in place of the damaged section.

The new hybrid line offers the best of both types of transmission lines mentioned. The bullet is eliminated by implementing a bolted contact joint. The bolted joint is eight times more reliable than the traditional bullet. Allowance for thermal expansion is provided using a patented[1] technology where the corrugated inner conductor is pre-strained and "locked" to the outer conductor at each flange by using a strain insulator. This technology keeps thermal expansion of the inner conductor contained within each segment. Since the inner conductor is continuously corrugated, repair of a damaged section is performed by removing the flange hardware and pulling the section apart to expose and disengage the inner conductor bolted contact. This capability, along with its electrical performance, provides the same benefits of traditional rigid line without the inherent problems of a bullet.

ELECTRICAL CHARACTERISTICS

Reflected Power

The maximum reflection for each section of the hybrid line is 1.02 VSWR (40 dB return loss). The reflected power can be modeled as a mismatch in impedance at each flange. This assumption is quite accurate since the strain insulator lies within each flange. A 300 foot section in the band 698 to 758 MHz was modeled using conventional transmission line equations. The resulting return loss curve is as shown in Figure 1. A 300 foot length of line was then assembled to verify this result. Figure 2 shows that the prototype is very similar to that which was predicted.

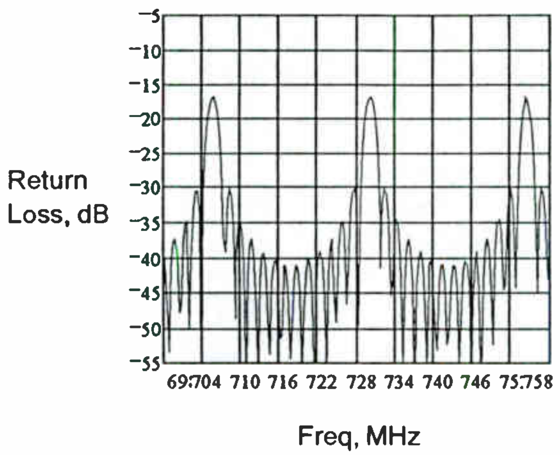

Figure 1: Model of a 300 ft Hybrid Line

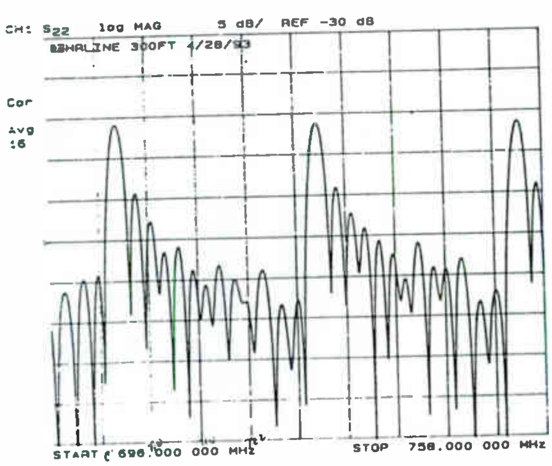

Figure 2: Prototype Test of a 300 ft Hybrid Line

Attenuation

For geometries of the hybrid line, the attenuation can be calculated as a function of frequency. Measurements confirm these calculations, and the attenuation can be expressed as equation 1.

$$Atten = 0.00463\sqrt{f} + 8.98 \times 10^{-5} f \; dB/100' \quad (1)$$

The contribution of the conductors is proportional to the square root of the frequency and the contribution of the dielectric is directly proportional to the frequency. Table 1 shows the values for attenuation given a VSWR of 1.0, ambient temperature and atmospheric pressure with dry air.

The 300 foot prototype length was also measured for attenuation, and the result is shown in Figure 3.

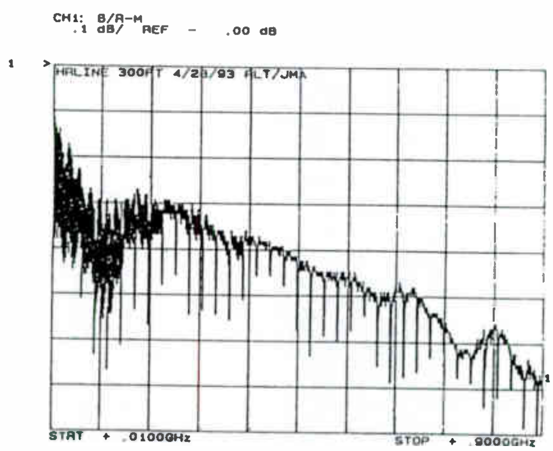

Figure 3: Attenuation of 300' length

Peak Power

Peak power is the maximum power allowable and is limited by voltage flash over. Voltage flash over is the voltage at which arcing will occur between the inner and outer conductors in an air dielectric. The magnitude of this voltage is directly proportional to the spacing between the inner and outer conductors. Peak power was determined by using equation 2.

$$P_{pk} = \frac{\left(\frac{V\alpha}{SF}\right)^2}{Z_o} \qquad (2)$$

Where V is the rms production test voltage (if a DC test voltage is applied, V=Vdc/1.414), α is a DC to RF conversion factor (empirically derived), SF is the safety factor and Z_o is the characteristic impedance of the transmission line in ohms.

For the hybrid line, and using equation 2, the peak power is 2260 kW.

Average Power

Average power is determined by the maximum allowed operating temperature of the inner conductor. Table 1 contains the average power rating for the hybrid line for an inner conductor temperature of 216°F, VSWR of 1.0, an ambient temperature of 104°F and atmospheric pressure with dry air.

Table 1: Attenuation and Average Power Values for the Hybrid Line

Television Channel No. (MHz)	Attenuation dB/100 ft	(100 m)	Average Power kW	Television Channel No. (MHz)	Attenuation dB/100 ft	(100 m)	Average Power kW
• (1.00)	0.0047	(0.015)	1613.73	35 (597.25)	0.167	(0.547)	66.03
2 (55.25)	0.039	(0.129)	217.10	36 (603.25)	0.168	(0.551)	65.70
3 (61.25)	0.042	(0.137)	206.19	37 (609.25)	0.169	(0.554)	65.38
4 (67.25)	0.044	(0.144)	196.78	38 (615.25)	0.170	(0.558)	65.06
5 (77.25)	0.048	(0.156)	183.60	39 (621.25)	0.171	(0.561)	64.74
6 (83.25)	0.050	(0.163)	176.86	40 (627.25)	0.172	(0.565)	64.43
• (100.00)	0.055	(0.181)	161.37	41 (633.25)	0.173	(0.569)	64.13
7 (175.25)	0.077	(0.253)	121.90	42 (639.25)	0.174	(0.572)	63.83
8 (181.25)	0.079	(0.258)	119.86	43 (645.25)	0.176	(0.576)	63.53
9 (187.25)	0.080	(0.263)	117.93	44 (651.25)	0.177	(0.579)	63.23
10 (193.25)	0.082	(0.268)	116.08	45 (657.25)	0.178	(0.583)	62.95
11 (199.25)	0.083	(0.273)	114.32	46 (663.25)	0.179	(0.586)	62.66
12 (205.25)	0.085	(0.278)	112.64	47 (669.25)	0.180	(0.590)	62.38
13 (211.25)	0.086	(0.283)	111.03	48 (675.25)	0.181	(0.594)	62.10
14 (471.25)	0.143	(0.468)	74.34	49 (681.25)	0.182	(0.597)	61.83
15 (477.25)	0.144	(0.472)	73.87	50 (687.25)	0.183	(0.601)	61.56
16 (483.25)	0.145	(0.476)	73.41	51 (693.25)	0.184	(0.604)	61.29
17 (489.25)	0.146	(0.480)	72.96	52 (699.25)	0.185	(0.608)	61.03
18 (495.25)	0.147	(0.484)	72.51	53 (705.25)	0.186	(0.611)	60.77
19 (501.25)	0.149	(0.488)	72.08	54 (711.25)	0.187	(0.615)	60.51
20 (507.25)	0.150	(0.491)	71.65	55 (717.25)	0.188	(0.618)	60.26
21 (513.25)	0.151	(0.495)	71.23	56 (723.25)	0.189	(0.621)	60.00
22 (519.25)	0.152	(0.499)	70.82	57 (729.25)	0.190	(0.625)	59.76
23 (525.25)	0.153	(0.503)	70.41	58 (735.25)	0.192	(0.628)	59.51
24 (531.25)	0.154	(0.506)	70.01	59 (741.25)	0.193	(0.632)	59.27
25 (537.25)	0.156	(0.510)	69.62	60 (747.25)	0.194	(0.635)	59.03
26 (543.25)	0.157	(0.514)	69.24	61 (753.25)	0.195	(0.639)	58.80
27 (549.25)	0.158	(0.518)	68.86	62 (759.25)	0.196	(0.642)	58.57
28 (555.25)	0.159	(0.521)	68.48	63 (765.25)	0.197	(0.646)	58.34
29 (561.25)	0.160	(0.525)	68.12	64 (771.25)	0.198	(0.649)	58.11
30 (567.25)	0.161	(0.529)	67.76	65 (777.25)	0.199	(0.652)	57.88
31 (573.25)	0.162	(0.532)	67.40	66 (783.25)	0.200	(0.656)	57.66
32 (579.25)	0.163	(0.536)	67.05	67 (789.25)	0.201	(0.659)	57.44
33 (585.25)	0.165	(0.540)	66.71	68 (795.25)	0.202	(0.662)	57.22
34 (591.25)	0.166	(0.543)	66.37	69 (801.25)	0.203	(0.666)	57.01

SYSTEMS USING THE HYBRID LINE

WFFT-TV, Fort Wayne, IN

The hybrid line was used in conjunction with existing compensated elbow complexes at the junction from horizontal to vertical and also at the antenna end of the transmission line. Without tuning, the system had a system return loss as shown in Figure 4 as seen at the patch panel.

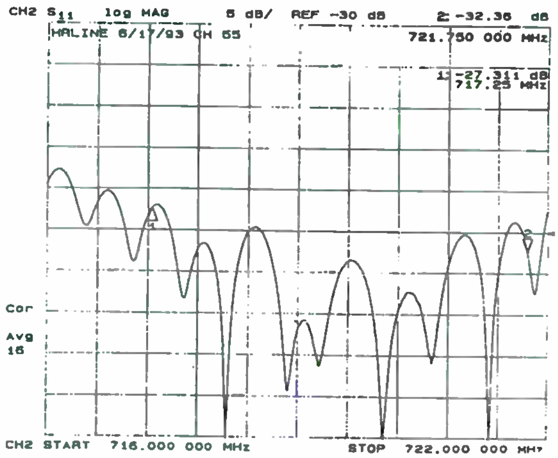

Figure 4: System Return Loss
WFFT-TV, Ch 55 Fort Wayne, IN

Since the system had used existing components, tuning was deemed necessary. A quick check at how the system would act with tuning on the near end revealed that the system VSWR would flatten out as shown in Figure 5. This was only a quick check. At the time of this paper, the tuning section had not yet been installed.

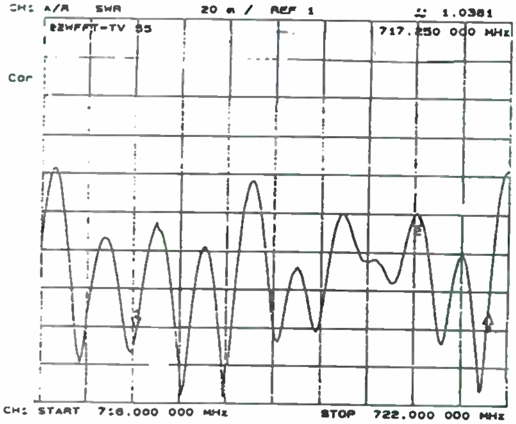

Figure 5: "Tuned" System Return Loss
WFFT-TV, Ch 55 Fort Wayne, IN

WFMZ-TV, Allentown, PA

For this system, the outer conductors were existing outers which were refurbished. Figure 6 shows the system return loss, with tuning, at the patch panel.

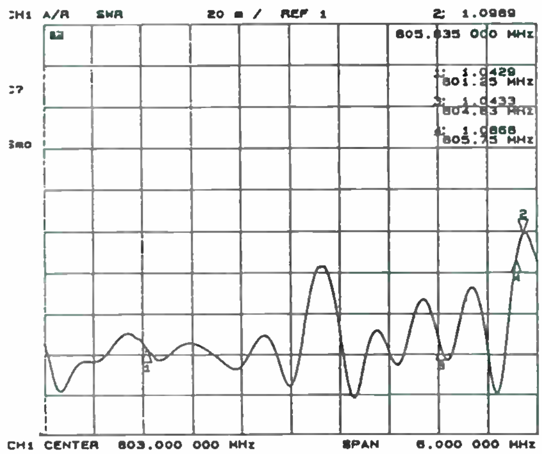

Figure 6: System VSWR with Tuning
WFMZ-TV, Ch 69 Allentown, PA

WGTU-TV, Traverse City, MI

This system involved no existing components or refurbished pieces. The system return loss with no tuning is as shown in Figure 7.

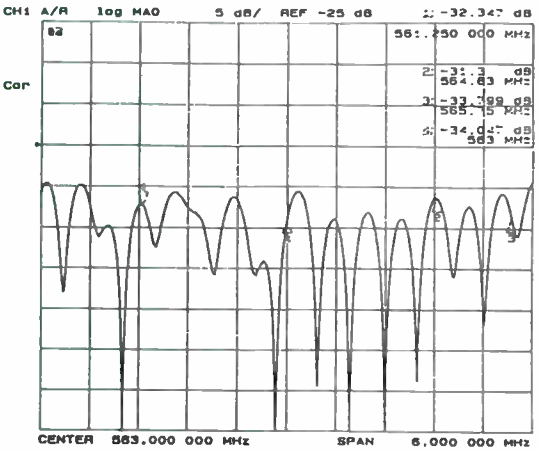

Figure 7: System Return Loss
WGTU-TV, Traverse City, MI

CONCLUSION

A new hybrid line has been developed for use in VHF, UHF and FM broadcast frequencies. The new hybrid line overcomes the problems associated with sliding contact joints. Its versatility shows that it can be used with hybrid components or use with existing components.

ACKNOWLEDGEMENTS

The successful conduct of a major development program such as this is the result of important contributions of many people. I would like, however, to especially mention those on the development team consisting of Mike Acosta, James Arredondo, Ed Devine, Paul Lawerence, John Leone, Len Ollearis, Jim Wu and all the others that helped to make this project successful. I would also like to thank the management of Andrew Corporation, Heliax Products, for supporting the publication of this paper.

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[1] U.S. Patent No. 4,831,346

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RFR UPDATE

Monday, March 21, 1994

Moderator:
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*Papers not available at the time of publication.

SBE

DESIGNING A SERIAL DIGITAL TV FACILITY

Tuesday, March 22, 1994

Moderator:
Marvin Born, WBNS TV-10, Columbus, OH

TEST METHODS FOR ANALYZING DIGITAL VIDEO
Konrad Finck
Advanced Audio Visual Systems
Sioux Falls, SD

***PUBLIC TELEVISION EXPANDING INTO THE DIGITAL WORLD**
Howard Miller
Public Broadcasting Service
Alexandria, VA

THE MERGING OF COMPUTER AND VIDEO: USING ETHERNET AND SCSI FOR DIGITAL VIDEO INPUT AND OUTPUT
Stephen Kilisky
Abekas Video Systems
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SERIAL DIGITAL, THE NETWORKING SOLUTION
Steve Haines
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BUILDING AND OPERATING A MULTI-FORMAT ALL-DIGITAL TELEVISION PLANT
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*Papers not available at the time of publication.

DIGITAL VIDEO SIGNAL ANALYSIS FOR REAL-WORLD PROBLEMS

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Digital video processing, storage and transmission have undeniable advantages, which have caused many facilities to adopt these technologies. The "lossless" nature of dubbing in the digital domain, coupled with the seemingly endless variety of effects available have made the technology indispensable in post-production, and increasingly, in broadcasting.

Unfortunately, although digital video has eliminated many problems that were common in the analog environment, digital video is not always quite as free of problems as one would like to believe. Incompatibilities between equipment of different manufacturers, equipment that does not meet published specifications, mechanical/wiring problems, and old-fashioned breakdowns have challenged the engineering staff of many facilities.

We will examine some real-world situations where digital video systems fail to give satisfactory performance, and illustrate how a careful analysis of the digital video signal allows the cause of these problems to be discovered and corrected.

We will also point out critical parameters that need to be carefully monitored in order to prevent problems from developing. Early warning signs will be discussed.

This subject will interest both those engineers who install equipment and perform proof of performance testing, and those who operate equipment on a regular basis.

When specifications for a standard are established, the various parameters of the interface, signal, or whatever is being specified, are examined and a determination is made as to the tolerance that can be allowed. These limits are established with the philosophy that variations less than those specified will not cause any degradation in the quality of operation.

Thus in the analog world for example, when RS 170 A specifies that the horizontal sync pulse should be 4.7 microseconds in duration, it also specifies a tolerance of ±0.1µS. The assumption is that a sync pulse that ranges in length from 4.6µS (4.7µS - 0.1µS) to 4.8µS (4.7µS + 0.1µS) will not cause any improper operation.

It is further assumed that all equipment that is intended to operate in this standard environment will produce signals that meet this specification, and that all equipment that receives this signal will operate correctly if the signal meets the standards set out in the specification. Thus we assume that a signal with a sync pulse of 4.8 μSec. will not cause improper operation in a properly operated apparatus receiving the signal. This seems like a most obvious conclusion, but the obvious is sometimes overlooked.

Specifications for digital video signals are likewise established with the purpose of setting limits for deviation that will not cause improper operation. Like analog signal specifications, digital signal specifications also establish the limits in which the sending equipment must operate and which the receiving equipment must accept.

In the analog environment, any violations of the specified limits may cause gradual degradation of the resulting image. The degree of degradation may depend on the receiving equipment tolerance. The acceptability of the degraded signal (image) will depend on the viewer, his expectations, training and personality. Minor violations of the specified limits occur regularly and are often viewed as unimportant. The amount of tolerated variation will often vary with the type of facility, corresponding to the type of end user for the product.

In the digital environment, small violations of the specified limits will usually show no effect at all in the resultant image. Both the trained and untrained viewer will see no image degradation. This is due to the nature of digital information transfer.

A digital receiving device has a latitude of acceptable variation that will not cause errors in the data transfer. Thus, for example, the input of an ECL device will interpret a level falling within the range of -0.81V to -0.96V as a "high" state, and a level falling in the range of -1.65V to -1.85V as a "low" state. Signal levels can thus safely vary within this range and cause no errors in data decoding. See Figure 1.

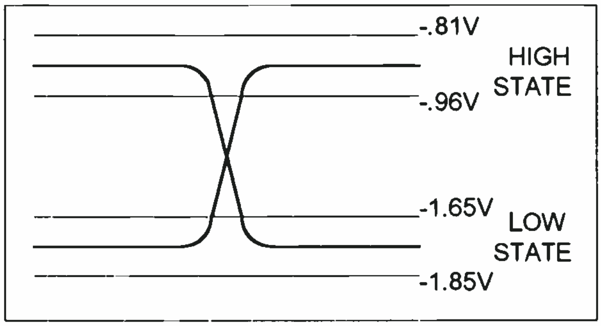

Figure 1

In the case of a digital video signal, allowing variations within the ranges mentioned above will cause no image flaws. Variations greater than those allowed by the specification may also cause no errors. This is due to the fact that the individual receivers may well have performances superior to those required by the specification. Deviations greater than those allowed by the specifications may not cause any increase in errors in the decoded data, and thus no decrease in image quality.

Continuing to increasingly violate the specified limits will eventually cause an error in data interpretation to occur. Such errors can have catastrophic consequences or may be as minor as a "sparkle" in the video image. The catastrophic nature of digital failure has often been referred to as the "digital cliff". The metaphor is very apt since the digital signal will be perfect even though a parameter is approaching and passing the normal limit. At a given moment however, the parameter will reach the point where the data can no longer be correctly interpreted and the signal will be errored. It will "fall off the cliff".

Unlike analog signals where the degradation is gradual and only apparent to the highly trained eye, errors in the digital signal are usually visible to the most untrained eye, and thus completely unacceptable.

The tolerance range of a digital signal being quite broad, and the nature of the "digital cliff" effect being what it is, users of digital video equipment often operate without much regard for the specified parameters. This is due partly to the difficulty in measuring the parameters and partly due to a false sense of security brought about by the nature of digital signal transfer.

The operator who desires a reliable installation is concerned with the safety margin in his system. In other words, he wants to know how close he is to the "digital cliff". Although it is unthinkable to operate a video studio without a waveform monitor, many digital installations are operated without any equipment to monitor the digital video signals. These operators are wandering about on the plateau of error free digital operation with their eyes closed.

Operators that use good practices are rejecting this risk and turning to methods that allow them to monitor the parameters so that they can predict dangerous conditions and remedy them before valuable time and material is lost.

We will examine parameters of a digital video signal to illustrate how this philosophy can be valuable in the real world.

All examples given will assume a video signal conforming to SMPTE RP 125M (Bit parallel component (4:2:2) digital video) and RP 259M (10 bit 4:2:2 component digital video). Composite (4fsc NTSC and PAL) video has very similar parameters and in many cases the examples also apply to these signals as well.

A bit parallel digital video signal is specified as having peak to peak amplitude of 0.8 to 2.0 Volts. Signals having a peak to peak amplitude less than 0.8 Volts may not be correctly interpreted. The average voltage (the midpoint between the "high" and "low" data states) is specified as -1.3Volts ±15%.

Operation outside these specified parameters is courting disaster, since the receiver may not correctly interpret such signals. Thus prudent operators will want to continuously monitor these levels or at least periodically verify them. Sound operating practice would insist on establishing proof-of-performance records that would document the levels existing at installation and their evolution over time. These parameters would also be verified and documented whenever repairs or changes to the system occurred.

Low digital signal levels, whether in a parallel or serial interface, can result in data errors. The errors will occur at statistically random times, resulting in "sparkles" in the video image at best or more seriously, in loss of sync.

The S310 continuously monitors both parallel and serial levels including peak to peak voltage, DC offset voltage (average voltage level) and common mode voltage (low frequency signals that modulate the digital bit stream waveform).

Another parameter that is critical to proper data decoding is the jitter present on the signal. Jitter is the difference in timing between where the data transition "should" be and the actual location of the transition. The actual transition may occur before or after its correct position. This is particularly critical for signals being transmitted in serial form, since the signal is sent without an accompanying clock signal. In order to be able to decode the logical levels of the signal a clock is recovered from the data stream, and this recovered clock is used to decode data. Since the data transfer is asynchronous, variations in time (jitter), if sufficiently large, can cause the data to be incorrectly interpreted since the synchronism is lost. The data then becomes "garbage" since the correct bit order is lost .

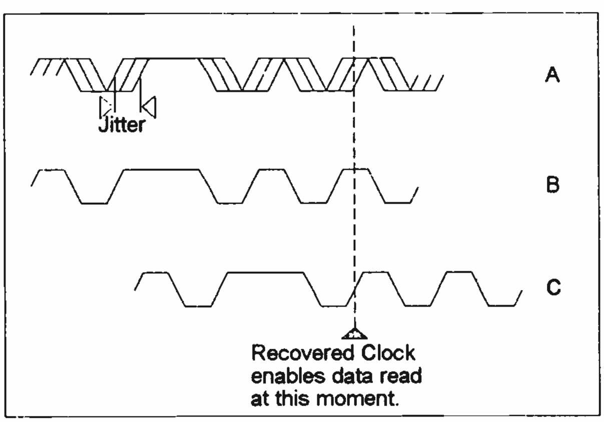

Figure 2

In Figure 2 we see a serial signal with jitter present. Waveform A shows the jitter present. At time B, the data is well centered on the recovered clock and will be correctly interpreted. At time C, the signal is instantaneously advanced, due to jitter. The data is now no longer properly centered at when the recovered clock validates the reading of the data, and an error will occur.

A small amount of jitter is not dangerous to proper data decoding, but if the amount of jitter becomes significant with regard to the period of the data cycle, errors can occur.

It is thus imperative to measure the amount of jitter present at the output of each transmitter, and to periodically check to see that the level has not changed. Again, the operator needs to be able to determine how close to the edge of the "cliff" he is operating.

Voltage levels and jitter are two parameters that relate to the electrical waveform that is used to transfer the data. They can be measured with generic equipment such as an oscilloscope. The difficulty in doing so relates to the frequencies involved.

A serial component digital video signal (SMPTE 259M) produces a signal at 270 megabits per second. In order to accurately represent this signal, a 'scope bandwidth of about 2 gigahertz is required. In addition, considerable knowledge of the signal structure is required to properly interpret the display.

The S310 Digital Video Analyser performs these tests automatically and displays the results of the measurements on an LCD display. No interpretation or special knowledge is required. This ease of use encourages constant or frequent monitoring of signal conditions and thus much lower risk for costly down time.

Other essential parameters of the digital video signal cannot be analyzed with generic equipment.

In addition to the electrical "carrier" of the data, the actual data stream can contain errors that will adversely affect quality.

One such parameter is the Timing Reference Signal, or TRS. The timing reference signal is a special combination of digital words that are used by the equipment for synchronization purposes.

Digital video equipment does not use a sync signal of the same type as an analog video signal. Instead the TRS words occurring in the data stream indicate to the receiving equipment that the time for horizontal or vertical blanking has occurred.

The TRS begins with a "preamble" which is a special combination of digital words consisting of FF,ØØ,ØØ$_{hex}$ in 8 bit format, or 3FF,ØØØ,ØØØ$_{hex}$ in 10bit format. The digital values FF and ØØ are called "reserved codes", since they are reserved for use as preambles to the TRS only.

Reserved codes that are allowed to occur in the active video period can fool certain equipment into believing that the sync time has arrived and cause false sync triggering. This is a very serious error which will destroy the integrity of the image.

The S310 Digital Video Analyser continually monitors for reserved codes during active video and indicates their presence. In addition, by connecting a monitor to the test output jack on the front panel of the S310-3 Service Keyboard option, the exact location of these reserved codes can be determined.

It is important first of all to determine if the errors are occurring randomly or at a specific position in each field of video. In this way the cause of the error can usually be determined.

Random errors indicate data that can be related to levels or jitter. Unconcealed tape errors could also be contributing to random errors. On the other hand, reserved codes that occur at the same place each frame usually indicate a failure in the equipment.

Following the three digital words of the preamble is the "XY" word (XYZ in 10 bits format) Figure 3 shows the significance of each bit in the XY word.

MSB									LSB
1	F	V	H	P3	P2	P1	P0	0	0

F= Field parity, 0= Field 1, 1= Field 2
V= Vertical, 0= Active video, 1= Vertical blanking
H= Horizontal, 0= SAV, 1= EAV
P3,P2,P1,P0= Hamming code

Figure 3

As mentioned earlier, errors in the TRS are quite serious since they can cause loss of synchronization in the receiving equipment. The S310 monitors the TRS continually and indicates any error that has occurred.

In addition, the S310 can output a pulse that allows the user to determine the position of the error in the video image. A "blip" appears in the image at the position where the error is occurring. Again, this allows the user to discern whether the error is occurring randomly, or at the same position in each field.

The S310 can also detect preamble errors and indicate their position as it does for reserved codes and TRS errors.

Normal digital video uses a range of digital quantization values to represent the variations in luminance and chrominance. These values do not extend all the way from Ø to 255 (1023 in 10 bit format). Instead a "buffer" zone is defined in the specifications. This "buffer" zone protects the digital video from "clipping" that would occur if the values reached Ø or 255. The zone also makes it less likely that valid video signals would produce the reserved code values of ØØ and FF_{hex}.

Valid video values range from 16 to 235 for luminance (64 to 940 in 10 bit format), and 16 to 240 for chrominance (64 to 960 in 10 bit format).

Values outside of this range are not recommended for use. Equipment should normally be set up to exclude these values from regular use.

The S310 monitors the video values continually and signals any excursions into these non-recommended areas. It also produces a video monitor signal that shows the image areas where the non-recommended values are occurring. This is very useful in determining which portion of the image has produced the values that are out of range.

The S310 also monitors the digital video signal for bit activity. Front panel LEDs indicate activity on each of the 10 bits. Lack of bit activity can indicate cable problems in parallel transmission modes, or hardware problems in serial transmission.

A specified parameter in parallel transmission is the phase relationship of the clock to the data lines. The data on the 8 or 10 data lines is read on the positive-going transition of the accompanying clock signal. In order to accurately decode the data, the data signals need to be properly centered on the clock transitions Figure 4 shows the proper relationship between the signals.

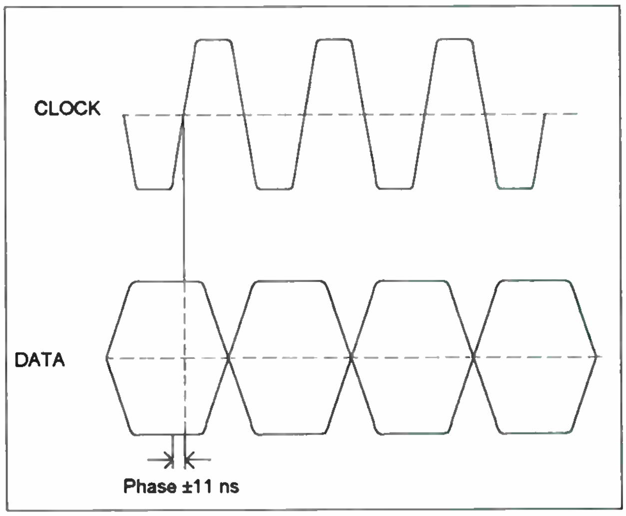

Figure 4

The specifications (SMPTE 125M) indicate that the phase difference between the center of the data transition and the rising edge of the clock signal must be less than ±11ns. Phase differences greater than this can cause errors in decoding the data.

The S310 allows the phase relationship between the clock and all of the data lines, or between the clock and any selected data lines to be constantly monitored so that any discrepancies can be detected and measures taken to correct the situation. This is an exclusive patented test performed by the S310.

If all of the above parameters are held within specifications there is an excellent chance that digital video signal transfers will occur without error. In the case where visible faults are occurring in the video image, an analysis of these parameters will almost always reveal a deviation from the spec that is causing the visible fault. Thus, detailed analysis of the video signal offers valuable insurance against problems in digital video, plus allows the operator to effectively troubleshoot any problems that arise.

THE MERGING OF COMPUTER AND VIDEO: USING ETHERNET AND SCSI FOR DIGITAL VIDEO INPUT AND OUTPUT

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As computer processing power increases and prices dramatically decrease, video facilities are looking to replace dedicated hardware with general purpose computers running software tailored for video applications. This revolution in the computer industry has also led to the proliferation of 3D computer graphics departments in most video facilities.

Ethernet and SCSI greatly enhance the versatility and functionality of video peripherals connected to computers. Currently Digital Disk Recorders are the most common video peripheral that use SCSI and Ethernet to integrate with computers. Ethernet and SCSI permit the disk recorder to take full advantage of the multi-user and multi-tasking UNIX operating system. In addition, Ethernet and SCSI are the simplest way to integrate video into a computer environment.

Today's engineer is faced with having to learn a new set of terminology and skills in order to integrate computer equipment into a video environment. This paper helps to explain some of the technical issues involved in integrating computers with video.

OVERVIEW OF ETHERNET

Ethernet is a Local Area Network standard originally developed at Xerox's Palo Alto Research Center in 1973. The standard is detailed in IEEE 802.3. Ethernet interconnects a group of computers. The transmission medium typically uses 50 ohm coaxial cable. Each computer is attached to the network using a device called a transceiver. The transceiver behaves like a traffic cop, routing all the data sent over the network to their proper destinations, and controlling all activity on the network.

Data is passed serially at 10 MHz in the form of packets, having anywhere from 46 to 1500 bytes of data. Each packet carries addressing information to show its source and destination.

Unlike the public switched telephone system or a video routing matrix the single Ethernet cable is shared by all the devices on the network so there are a set of rules to determine when a node can access the cable. The technique is referred to as Carrier Sense Multiple Access with Collision Detection (CSMA-CD).

Before transmitting, a node listens to confirm that nobody else is transmitting, then as it transmits it continues to monitor the cable in case another node started transmitting at the same time. If two devices transmit simultaneously it is referred to as a collision and both devices have to stop immediately and wait a random amount of time before attempting to transmit again.

Physical Integration

Installation on Ethernet is just a question of the plugging the video peripheral into a transceiver, which is a small box that provides an isolated interface to the Ethernet coaxial cable. In some instances (particularly in the case of "Cheapernet") the transceiver can be part of the Ethernet interface board in the host computer or video peripheral. Remote transceivers can be up to 50 meters away from the computer and typically come with a plug-in module to allow either a spike tap or BNC connectors to interface to the cable.

Cheapernet uses thin RG-58 50 ohm coax and BNC style connectors. In this case the coaxial cable is 'T'eed directly onto the Ethernet interface in the computer. Higher level applications use better quality thick cable and external transceivers which can attach to the cable with a spike-like tap. The primary advantage of a non-intrusive tap is that the transceiver can be removed without taking the network down.

The thick coax can be used for networks up to 100 nodes on 500m of cable. The transceivers should be placed at multiples of 2.5m on the cable. Cheapernet is limited to 30 nodes on 185m of cable. Only two repeaters are allowed on a local network because of the propagation delays through them.

Once connected to the network, the system manager specifies an Internet (IP) address for the video peripheral. The other computers on the network can then use this Internet address to access the services supported by the video peripheral. Typically the IP address is entered in the *hosts* file which resides in the */etc* directory of the computer.

As an Ethernet device, the video peripheral should appear to be another node on the network. In the case of a disk recorder, image files can be transferred to or from the disk recorder using Ethernet. There is no need for a separate VTR animation controller, frame buffer, sync generator, or special software.

SCSI

SCSI stands for *Small Computer Systems Interface*. The SCSI-1 standard is documented in ANSI X3.131-1986. The SCSI-2 standard is documented in ANSI X3.131-199X. SCSI has become one of the most popular methods for connecting peripherals in the computer industry.

Improvements in memory and CPU products are shifting the processing bottleneck to the I/O bus. SCSI is ideally suited for high performance single-user environments, where maximum performance is required of peripheral devices. As users require additional data transfer rates and more functionality, SCSI-2 will be a more viable solution, yet still maintain compatibility with SCSI-1.

Benefits of SCSI

- High data transfer rates
- Disconnect/Reconnect - to utilize the single bus SCSI interface, the peripheral devices receive a command, then reconnect with the bus after data has been found and retrieved. This eliminates bus tie-ups by slower peripherals such as tape drives.
- Cabling Configuration - The 50-pin ribbon cable allows users to daisy-chain peripherals. Up to 7 peripherals can reside on the bus. Typical maximum cable length is 6 meters. However, if differential drivers are used, then lengths up to 25 meters are possible.

General SCSI-1 Specs

- Uses 50 pin Centronics connector.
- 8-bit data path.
- 18 signal lines: 9 data (including parity), and 9 control lines.

Single-Ended vs. Differential SCSI

Single-ended hardware is the predominate configuration. It relies on a reference to ground, and the signal uses a single wire for each data and control line, varying the voltage for operation. The use of a ground produces some signal noise and ground shifts. This is why the single-ended specification limits the length of the bus to 6 meters.

Differential bus lines are balanced. This eliminates the ground and uses a pair of wires for the "high" and "low" signals for each data and control line. The resultant cleaner signal, allows for bus lengths up to 25 meters.

Asynchronous vs. Synchronous

Asynchronous data transfers means the receiving device acknowledges receipt of each byte before the next byte can be sent. The data rate for asynchronous SCSI transfer is 1.5Mb per second.

Synchronous data transfer allows blocks of data to be sent at a predetermined rate without receiving an acknowledgment for each byte. The peak synchronous transfer rate is 5Mb per second. Please note that a peak rate is not necessarily the same as a sustained rate.

Installation Notes

Even though a cable may meet SCSI specs for twisting and shielding, SCSI allows for a wide range of impedance. It is important to use the same quality cables along the entire bus to avoid impedance mismatches. The greater the mismatch the more noise that will be introduced onto the bus. it is also preferable that the cables be as short as possible.

Both ends of the SCSI bus should be terminated in the same manner. SCSI-2 specifies the use of an active terminator for fast single-ended busses. Active termination uses a voltage regulator in the circuit to terminate the signal.

SCSI-2

General SCSI-2 Specs

- Uses a 50-pin micro-D connector.
- 8, 16, or 32-bit data path.
- Compatible with SCSI-1.
- Command Queing & the Common Command Set (CCS)
- Fast SCSI
- Wide SCSI

Command Queing and the Common Command Set (CCS)

Command queing allows up to 256 commands from each initiator to each logical unit. The host can send several commands without waiting for acknowledgment. This allows for higher throughput.

Commands can be designated as ordered or unordered. Ordered commands are processed on a first-in, first-out basis; unordered commands are sorted by priority and executed in a sequence that achieves the fastest throughput.

The CCS is now standardized and fully documented. It also includes support of more types of peripherals than SCSI-1. The CCS includes four levels of commands: mandatory, extended, optional, and vendor-unique.

Fast SCSI

Fast synchronous SCSI doubles the sustained data rate to 10 megabytes per second using an 8 bit bus. Fast SCSI is an optional mode, both the host and target must be equipped for it.

Wide SCSI

Wide SCSI allows for 16 and 32-bit data paths. When combined with Fast SCSI sustained data rates of up to 40 megabytes per second are achievable. 32 bit wide SCSI requires two 16-bit connectors and cables. The "B" cable uses a 68 pin connector.

SCSI-3

Multiple fast SCSI-2 devices can clog a network. The pending SCSI-3 proposal attempts to address this issue. SCSI-3 will use a layered approach. Single-ended and differential techniques will be supplemented by two new layered approaches known as *P1394* and *SSA*. A minimum 20 Mb bandwidth will be possible. SCSI-3 will also remove the limitations on distance and increase the number of peripheral devices that can be linked together.

SCSI-3 will be referred to as the *Interlocked Protocol*. Another layer that will be added on a software level is the *Common Access Method 2 (CAM-2)*. CAM-2 will allow users to write device drivers that will run on different operating systems with few or no changes. Finally, SCSI-3 will document and standardize the specifications of termination methods.

APPLICATIONS

The key benefit of Ethernet on a video peripheral is to take advantage of the multi-user capabilities commonly available in a computer environment. Ideally multiple users should be able to access the video peripheral simultaneously. The primary video application of Ethernet and SCSI is digital image file transfer.

Ethernet Digital Image File Transfer

The simplest method to transfer images digitally between computers and video peripherals is to use standard computer Ethernet remote file transfer utilities. Most of these utilities are based on the UNIX operating system.

The UNIX operating system was developed in 1969 by Bell Laboratories. It is used by over one hundred computer manufacturers. The basic UNIX operating system has spawned many implementations. An implementation is nothing more than a customized UNIX version, usually for a specific brand of computer. Some of the more common implementations of UNIX are: System V, Berkeley (BSD 4.X), SGI Irix, and PC/TCP.

The UNIX *rcp* command is one way to transfer files between the video peripheral and other computers on the network. For non-UNIX computers, the official ARPA file transfer and remote login utilities called FTP (File Transfer Protocol) and Telnet are used. These are specifically intended to work between different computer architectures and operating systems.

Typically users transfer files to or from the video peripheral in the CCIR-601 file format or as RGB files. other common file formats such as TIFF, TGA, and PICT require a format conversion before they can be stored on the video peripheral. File transfers should be bi-directional, meaning that images can be transferred to or from the video peripheral.

SCSI Digital Image File Transfer

Historically, the primary use of SCSI is as a digital archive medium. Common archive devices are 1/4" streaming tape drives, 8mm Exabyte drives, and 4mm DAT drives. In addition, with increases in SCSI transfer speeds, SCSI has become an attractive medium for direct file transfer between a single host computer and video peripherals. SCSI file transfer is significantly faster than Ethernet. however, SCSI does not offer the networking support that Ethernet provides and has a fairly limited bus length when compared to Ethernet.

Ideally the computer and video peripheral should be able to share the same archive device. This requires that all the devices coexist on the same SCSI bus. It also means that they use a common SCSI file format. The standard UNIX *tar* file format is the most common SCSI archive format. The key benefit of video peripheral support for the *tar* file format is that users can then generate *tar* tapes on their computer that can be read directly by the video peripheral. In addition, *tar* tapes generated on the video peripheral can be read directly into the workstation.

Direct SCSI image file transfer between a single host computer and a video peripheral is known as SCSI target. In SCSI target mode, a computer becomes the master and the video peripheral is controlled by the computer. The peripheral is now a target device that the computer can transfer files to or from. A SCSI driver for each video peripheral must be written and installed on the host computer. The SCSI driver is simply an implementation of the SCSI common command set.

CONCLUSION

Ethernet and SCSI are standard computer industry interfaces available for a multitude of computer platforms. Including these interfaces on video peripherals provides two key benefits. First, it simplifies the task of integrating computer hardware with video hardware. Second, it increases user productivity by taking advantage of a multi-user networked environment.

GLOSSARY OF TERMS

ARP
Address Resolution Protocol - used to obtain Internet to Ethernet address mappings when they are needed rather than maintain a static list on each host.

ARPA
Advance Research Projects Agency, US Government agency responsible for developing TCP/IP family of protocols.

AUI
Attachment Unit Interface - the long way of saying transceiver cable. Normally limited to 50 meters and carries twisted pair differential signals for transmit, receive and collision detection. The AUI cable also carries 12v power for the Transceiver.

Bridge
Bridges are generally connections between Networks of the same type at the Data Link Layer.

Broadcast Packet
Ethernet packet carrying the address FF:FF:FF:FF:FF:FF which will be received by all the hosts on the network.

Cheapernet
An alternative form of Ethernet that uses thin RG58 50-Ohm cable with BNC connectors and 'T' pieces at the transceivers rather than that normally used for Ethernet. This cable suffers from greater loss and the cable run is typically limited to 185m. In all other respects it is electrically compatible with 'Normal' Ethernet.

Client
The consuming (user) end of a client-server relationship.

Connection
The link between two specific ports on two Internet Hosts analogous to a telephone call being set up between two phones on a network.

CRC
Cyclic Redundancy Check - a sort of serial check sum that assures the integrity of a serial data stream by using some sort of polynomial feedback.

CSMA-CD
Collision Sense Multiple Access with Collision Detection - Describes the mechanism that allows several devices to share the single Ethernet cable.

Domain Names
A name addressing scheme that uses a hierarchy of domain names to describe the address of a remote computer in a similar way to a mailing address eg simon@master.abekas.com

Ethernet V1 and V2
The original Ethernet standard was developed by Xerox and published jointly with DEC and Intel.

Ethernet Address
A 48 bit address conventionally written as 6 hex bytes separated by colons. Typically vendors ship equipment with the Ethernet address contained in a small bipolar ROM.

Fragments
Subdivisions of Internet packets - sometimes necessary if different networks have different maximum packet lengths.

FTP
File Transfer Protocol

Gateway
Gateways allow interconnection of different networks at the network layer.

IEEE 802.3
The IEEE standard for CSMA/CD Networks, forms part of the 802.x family of standards for Local Area Network Interfaces and Protocols.

Internet Address
A four byte number representing the address of a host on the "Internet".

Jabber
A condition detected by a transceiver to prevent locking up the network. If the output is active

for more than 1/10 second the transceiver should latch up and prevent further transmissions until the transmit signal is inactive for at least 1/4 second.

Jam
A collision signal generated by a repeater so that the cable segments on both sides of it are aware that a collision has occurred.

Heartbeat
See SQE

Host
A computer that is a node on a network.

Hostname
The name by which a particular machine is known at the user interface level - normally associated with a Network address.

NFS
Network File System (Developed by Sun Microsystems)

NIC
Network Information Center - The central repository for all information regarding the development of the ARPAnet

OSI Layered Model
The abstract model for the seven layer hierarchy of network protocols issued by ISO the International standards body.

Ping
Program that uses the ICMP Echo Request facility to verify the connections between two machines. Note that this only verifies correct operation up to and including the network layer or Internet Protocol.

Port Number
An addressing scheme within a host computer that allows more than one simultaneous connection to that computer.

Repeaters
Repeaters interconnect or buffer two sections of the same network with very little intelligent signal processing.

Server
A host or node on the network that provides a service, eg: a file server provides a file system for diskless nodes.

SQE
Signal Quality Error - some Ethernet transceivers generate a collision signal immediately after each transmission. This permits the Ethernet interface to verify that its collision detection circuitry is functioning correctly.

Socket
A socket is an abstraction in BSD Unix for the interprocess communications primitive known as the pipe. A socket can be opened in much the same way as a file would be opened.

Telnet
ARPA remote terminal/login program

10BASE5 and 10BASE2
IEEE names for Ethernet and Cheapernet respectively, abbreviated from 10MHz Baseband 500m and 200m maximum length.

Thin-LAN or Thin-net
see Cheapernet

Transceiver
The box which attaches the host computers' Ethernet Interface to the cable. The best ones use a spike arrangement to non-intrusively attach to the cable (eg without cutting it). The transmit and receive signals are coupled onto the cable through isolating transformers. The transceiver is also responsible for detection collisions.

X.25
Public Wide Area Packet Switched Network Standard

Yellow Pages
The Yellow Pages is a network service that provides network wide Password and system information from a single database. Written by Sun Microsystems but available on many systems.

SERIAL DIGITAL, THE NETWORKING SOLUTION?

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ABSTRACT

Serial digital video has revolutionised the television industry. It has given us the ability to connect a variety of systems together with a single cable and transfer video at the highest quality in real time. It does, however, have limitations which need to be considered when a network is required. The computer industry has used Ethernet networking for many years, and, when applied to the transfer of stills, Ethernet can show many benefits over serial digital video, provided the system design and management is optimised.

SDI APPROACH

The serial digital interface (SDI) provides all of the bandwidth necessary to accommodate eight or ten-bit CCIR 601 video, with audio if required, in real time, along standard 75 ohm co-ax cable. It is a standard which has enabled studios to reduce their cabling costs considerably, but it does have its limitations in certain applications when compared with other networking solutions.
A simple network based on SDI could be configured as shown in figure 1.

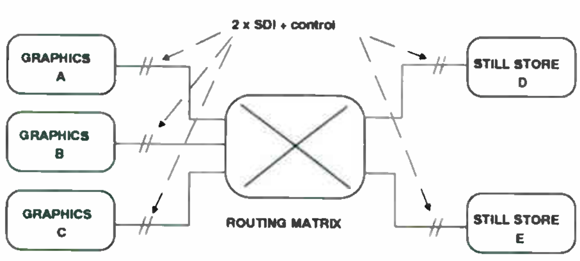

Fig.1 Typical serial digital network configuration

The system shown requires two video cables for each system so that the input of each system can be connected to the output of another. In addition a control cable is required so that there is no need for an operator at each end of the link to retrieve and capture images.

In order to facilitate transfers between a number of systems, routing technology is employed which can switch video and control between the required systems. Even under these circumstances, while system A is talking to system B, system C is locked out. If it becomes necessary to expand the network then the size of the routing matrix is the limiting factor. In order to keep costs to a minimum it is necessary to predict the future network requirements at the initial planning stage so that a matrix large enough to meet these requirements can be obtained. This means installing more than is initially required, which is not ideal. The alternative is to purchase only what is immediately needed and then replace the whole unit when it becomes necessary to expand, which effectively defers the cost but results in a larger outlay at the time of expansion.

If a network is aimed at moving stills around quickly, there are other factors, on top of the obvious point-to-point transfer time, which should be considered. First there is the time to record to disc to be taken into account. There is little point in providing a system which can transfer a frame in 1/30th of a second if it takes a few seconds at the transmitting end to retrieve the image from disc and a few seconds at the other end to record the image to disc. There are also library management functions such as titling and cataloguing of recorded images which need to be considered. There will be occasions where these considerations are not important and SDI will provide an adequate result, but for the majority of work, a more complete solution is required.

ETHERNET APPROACH

The computer industry has been utilising networking techniques for many years in order to provide a range of solutions, from localised office networks to international distributed networks. Ethernet has become the standard for this type of networking and the experience of the computer industry can be tapped to provide stills networks for television.

Ethernet networking allows all systems to be connected together at the same time and communication between them can be simultaneous. All the control information can be transmitted along the same single co-ax cable as the picture information. As Ethernet is a multi-drop system additional stations can simply be connected onto the line, which allows for very flexible expansion. If the software provides self-configuration, adding to, removing from or reconfiguring the network is straightforward. It's simply a case of plug-and-play.

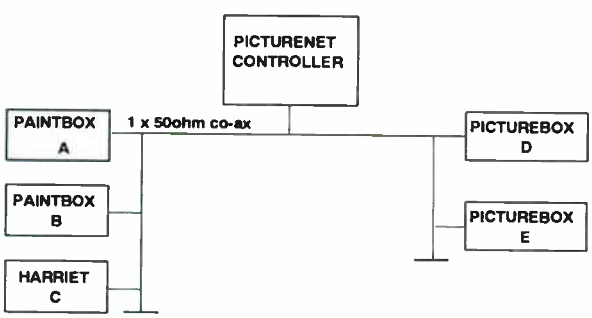

Fig. 2 Typical Quantel Picturenet configuration

Because all the systems on the network are able to communicate simultaneously, they each have access to every other system at all times. In this way, referring to figure 2, graphics stations A, B and C are able to service still stores D and E simultaneously. The fact that the control information is transmitted along with the picture means that the transmitted file can be saved onto the receiving system with its title, and any other relevant information, automatically. It can be a background task, which leaves the receiving system free to continue its operation unhindered.

In the past, one of the major limitations of Ethernet transfers has been the speed at which it operates. Standard Ethernet transfer protocols are generally slow as they have been designed for a wide range of applications. Data compression techniques have been developed to overcome this limitation, but when dealing with high quality graphics, full quality transfers without any data compression are needed. By streamlining the protocol and adapting it specifically for the task of transferring broadcast resolution stills and graphics, which has been done for Quantel's Picturenet system, it becomes possible to transfer a full-frame CCIR 601 image in a few seconds. Although this speed is perfectly acceptable for last-minute graphics and overall on-line archive, it is not fast enough for direct on-air operations. This has a bearing on the design of both the connected outstations and the system as a whole.

Individual outstations should have their own local storage for on-air operations or, in the case of graphics systems, for the storage of the basic graphics building blocks (backgrounds, textures etc.). Providing this local storage also gives the system inherent security because all the pictures required for local operations can be copied to local disc so that if the network fails in any way, the material is still available to the user. In order to ensure this security, each system must be a self-contained fully operational unit which can run independently from the rest of the network. This means each system must be autonomous with its own disc, processor and power supply. Distributing the power of the network around all the connected outstations means not only is there no weak link but a level of redundancy is provided to give the security essential for on-air operations.

In adopting Ethernet as a file transfer medium, the user can make use of the wealth of supporting equipment which the computer industry has developed over many years. A range of ancillary equipment is easily available which enables networks to be extended. For example, Ethernet repeaters provide very cost-effective means of interconnecting systems which are hundreds of metres apart, and with fibre-optic repeaters, interconnects of up to two kilometres can be achieved. Ethernet bridges further expand this capability, and by interfacing to standard communication lines, the network can grow across states and even to a global level. Any stills system which supports long distance links must also be capable of handling both 525 and 625 line images, as must any outstation on the network.

As the size of the network increases, so its traffic will increase proportionally which, in turn, may eventually lead to a reduction in the speed of file transfer. Ethernet bridges provide isolation between network areas, effectively dividing it, and can therefore be used to ensure that the speed of localised networks is not unnecessarily compromised by the increase in size. As always it is important that the system is well managed. For example the majority of the material necessary for transmission from a particular area should be available in that local area immediately prior to transmission. One of the major benefits of a distributed network is that any one system on the network is able to service any other system on the network. For example, it is conceivable that a Paintbox in Philadelphia could be responsible for providing all the stills for a Picturebox in Dallas via a network. Although this could be very useful for the transfer of the occasional graphic, it would not be as attractive for heavy use due to the cost of the link as well as the loading on the network. Consider the analogy of the road systems. If everyone living in or around town A worked in town B, and everyone living in or around town B worked in town A we would need highways at least twelve lanes wide to cope with the amount of traffic during the rush hour. Luckily this is not generally the case. Most people live in or around the town where they work. If you have an important appointment in another town which you must attend at a specific time, you would ensure that you avoid the rush hour when you travel. These same principles apply to still graphics networks.

So far the use of Ethernet for the purpose of distributing stills and graphics around a studio have been under consideration, but what about moving video? Obviously life is too short to consider sending video clips over the same network! Even using streamlined protocols the best times that could be achieved would be in the order of one and a half hours to send a minute of video. Currently the only way to transfer moving video around a network is to use serial digital video and utilise the routing technology described in figure 1. In many applications this has provided, and will continue to provide, an adequate solution. In the past the devices at either end of the links have tended to be tape-based, but real-time full bandwidth disc recorders have been available for almost a decade and as capacities have increased and costs have reduced, these systems have become much more viable. So a logical progression would be to develop a disc based video network where various outstations have simultaneous access to a central pool of video. This would allow clips of video to be input, edited and replayed without the need to transport them from station to station. In this case it would not be sensible to copy all the necessary output material to the local storage of the playout device, as is the case with stills, because it would increase the time to make the story ready to go to air. This raises the question concerning the worries of redundancy with centralised storage. For security the centralised storage must either be duplicated or incorporate redundancy such as has been built into Quantel's Dylan disc arrays. At the time of writing, the technology necessary for such a system has not been perfected, however, as faster data busses emerge, we may one day realise the dream of live video data networks.

In conclusion, serial digital video can provide an excellent means of system interconnection, having many useful applications in the studio. However, where there is a requirement for the interchange of stills and graphics, an integrated network, such as the Picturenet system, proves to be more appropriate. The principles developed for the integrated stills network, could equally be used as a basis for a future moving video network.

BUILDING AND OPERATING A MULTI-FORMAT ALL-DIGITAL TELEVISION PLANT

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Abstract

Four technology-driven developments constitute a rapidly growing and increasingly militant threat to the future financial health of analog NTSC television program producers and VHF/UHF broadcasters:

A Exciting, versatile and powerful computer-based work stations as alternative means to traditional camera/microphone/VTR equipment for shooting and editing analog NTSC television programming;

B Alternative wide-screen/high resolution and digitally compressed NTSC formats to provide quality versus cost tradeoffs to fractionate viewing audiences;

C Alternative means and competitors for instant real-time, and delayed time-expanded, home delivery of programming on request;

D Alternative means and competitors providing physical access to programming through direct mail and innovative retail merchandising channels.

To survive and grow in this hostile competitive environment, all participants in the original 40-year-old analog NTSC color television industry should already at least have started planning to participate in both alternative activities.

This paper describes the technical foundation of new, updated and/or expanded facilities which must be built before the marketing of any new production and/or distribution services can begin.

SMPTE standards which are the "blueprints" for constructing the foundation are identified and inter-related to the relevant standards being produced by international telecommunications and computer industry standards-setting bodies.

THE FUTURE: A MAELSTROM OF OPPORTUNITIES IN TV PRODUCTION AND DISTRIBUTION

Television program producers and broadcasters are now operating in an operational era full of new opportunities for generating revenues and profits.

The loftiest of the new program delivery services is the forthcoming "ATV" standard. It is being heralded for its ability to deliver wide-screen, high resolution motion pictures with theatrical surround sound into every living room. That prediction of course will become true only after the purse string holder in every TV home has bought an appropriate receiver, a large screen display, and a five channel sound system.

> ***In what near-future year will that population increase from "infinitesimal" to "miniscule?"***
>
> ***When will receivers be available which are compatible with all delivery service standards?***

In the meantime, the most profits-oriented of the delivery services being touted is the 10:1 or higher compressing of stereo NTSC channels into "VHS-like" (or lower) quality. This ostensibly provides viewers access to "500 channels," so they can receive whatever they want whenever they want it.

Asking the next obvious question:

> ***In this multi-dimensioned maelstrom of viewer options for "watching television," does the traditional VHF/UHF broadcaster of a fixed schedule of traditional commercials-supported entertainment, news and sports programs have any chance for survival?***

21ST CENTURY ROLES FOR TELEVISION PROGRAM PRODUCERS

As of January 1, 1994, television program producers were still free of government regulation as to what kind(s) of telegram program distribution formats they can integrate into their plants. Television program producers can therefore view as new customer prospects all the organizations transforming the Abstract's technology-driven developments B, C and D into revenue generators.

If they have the financial resources or can create new business order backlogs to take to the bank, they can acquire the A category equipment needed to create new products for still more new customers in new markets.

And if traditional VHF/UHF broadcasters somehow remain good credit risks, TV program producers can really profit by producing every script for release in all media!

Television program producers face an exciting and profitable era, perhaps even another 40 years long.

21ST CENTURY ROLES FOR VHF/UHF BROADCASTERS

"Broadcasters" now defines a substantial number of delivery channel operations alternatives to radiating a single signal from a single VHF or UHF stick. For the last decade the term has included "cable."

Every channel of the scores available on most cable systems looks the same (bad through good, both technically and programming-wise) as all the others. Viewers impulsively pick and choose what they watch by channel hopping, not by consulting a programming log. Each time "the network" pre-empts a time-hallowed schedule, or moves favorite shows around the weeknights in the "rating game," it sets itself up for permanently losing another prime time rating point fraction.

In 1994 the competitive picture has worsened in three more competitive dimensions. First, cable and telephone industry giants are simultaneously teaming up and waging wars to offer "500 channels" to viewers. Second, DBS and DAB (Direct Broadcast Satellite and Digital Audio Broadcasting) are beginning to attract audience away from both television and radio broadcasters and cable. Third, technical variations on MMDS (Multi-channel Multi-point Distribution Services) in newly available spectrum are being developed to offer competition to all the above.

Who's teaming up with VHF/UHF broadcasters?

The proliferation of competition prompts a question that needs clamorous asking and immediate thoughtful answering:

> ***What's the financial justification for continuing to pay the transmitter power bill and the rent and maintenance of the tower and antenna?***

In most metropolitan areas, 80 percent or more of viewers' TV sets can't directly "see" the stick from which the signal is radiating. If the transmitter quits in prime time, the station therefore loses only 20 percent of its tuned-in audience. The 60 percent (or less) of the 80 percent of viewers watching the station on cable continues to receive the signal. More and more that backbone delivery channel is local telco fiber, radiating from the station to cable system head ends both within and well outside the radiated signal coverage umbrella.

Will viewers buy new automated motor-driven VHF/UHF antennas on rotor-equipped masts as part of the receiver/display/ surround sound equipment package just to receive a broadcaster's second channel directly "off the air?" Especially if that channel is "only" a simulcast of the NTSC channel in wide screen high res with surround sound? The answer is an unarguable "NO!"

Are there any new ***profitable*** revenue generation possibilities for a VHF/UHF broadcaster to pursue, if the transmitter is dark, but cable systems re-broadcasting its signal still refuse to pay for the privilege?

The answer is an unarguable "YES!"!

They are the same four opportunities listed in the Abstract which television program producers can exploit to grow and prosper. VHF/UHF broadcasters can grow and prosper by changing their charters to become competitors to their suppliers and suppliers to their competitors.

BLUEPRINTS NEEDED TO BUILD A MULTI-FORMAT TELEVISION PLANT

Planning the Work; Working the Plan

The operating charter change called for in the previous paragraph -- "competing with suppliers and supplying competitors" -- isn't an activity that can be "winged." It takes time and visionary brainstorming to understand and assimilate what must be done to undertake the activities.

Planning the Work requires re-writing the organization's operating charter.

First -- Who are a station's suppliers that will become its competition?

(1) The networks that have in the past exclusively and loyally supplied their affiliates the prime time and daytime programming, global news and sports that build audiences.

(They are all already competing with local stations by working with the station's past home delivery competitors in well-publicized ways. A station's move to "bite the hand that (has) fed it" is therefore purely reactionary.)

(2) The independent national and local television production houses and distributors who have supplied much of the remainder of the programs and spot commercials that fill up the station's log.

(Competing with local ADI production houses and stations in the production of local spots is going to be price-oriented tough. But it's necessary to implement the "supplying competitors" plank in the new operating charter.)

Second -- Who are the station's existing competitors who will become customers for its supply of local programming, news, sports, spots, weather, traffic reports, et al?

(1) Independent stations with fractionated, loyal, small audiences, who will "stay tuned" to watch specialized, ad-supported programs they can't produce themselves.

(2) Cable systems throughout and beyond the ADI with the same audiences and the same programming needs.

(3) "Wireless cable" system operators operating in the MMDS and newly opened 28 GHz bands.

Are there new "non-broadcast" customers for the station's above-listed supply of customized television programming and production services? Of course!

(1) All municipal agencies and private businesses whose capability to function is determined by weather and traffic conditions.

(2) Business and government organizations that rely on videotaped communications and videoconferencing for the efficient operation of their enterprises.

(3) Education and health care institutions that are in the television production and distribution business because they have to be.

(4) Any business or government organization or library or individual who would pay to have access to information and programming resident in the station's news department and program archives.

(5) Etcetera, etcetera, etcetera. *(These "etc's" can be turned into specifics by more brainstorming conducted by everybody in the station, organized into new business development task forces.)*

Once this *Planning the Work* effort is completed, it's time to start ***Working the Plan.*** The following "blueprints" are offered to guide the creation of an action plan and budget that anticipates and efectively deals with all competitive challenges, by timely and judicious investment of resources of money and employee-based energy..

Blueprint 1: Upgrading the existing plant

After WW II ended, a large number of 1930s radio broadcasters applied to the FCC for licenses to operate VHF band (Channel 2 through 13) 6-MHz monochrome television channels, They subsequently embraced that 1950s "cash cow" profits-generating opportunity as their primary television market operating charter.

"Back to the Future" planning, focused on the 1930s "Golden Days of Broadcasting" profits-generating successes of their network-affiliated local radio stations, inspired these television broadcasting pioneers to to create "picture radio" television plant clones of their radio plants, and their late 1920s affiliations with the NBC Red and Blue and CBS radio networks. Local news was read by "rip and read" announcers in radio studios.

Film crews with Sound/CP-16s loaded with reversal film began to shoot news during daylight, which the 6 PM "talent" promised as "Film at 11." On-site film processors were a necessity.

"Telecines" with 16 mm sound film projectors and dual 35 mm slide projectors brought news clips, syndicated films and locally-produced art to air.

In the late 1940s, "CATV" (Community Antenna Television" systems for re-radiating VHF signals to homes behind the hills also came into being.

In 1953, some head-spinning, technically indefensible, politically pressured rule making reversals by the FCC finally established monochrome-compatible NTSC as the next television service for delivery of television programming on existing VHF channels. UHF channels 14 through 83 were made available for additional new local color television broadcasting services.

Later 1950s newscasts improved in presentation sophistication. Point-to-point 2 GHz microwave systems and Ampex VR-1000s made it possible to show unedited breaking news at 6 PM, with live voice-overs from the reporter on the scene.

In the 1960s, most local studio program originations went to color. News stayed with B/W reversal film rushed through the station's film lab until the mid 1970s. News Directors then began to use Sony 3/4U VO-3800 portables and VO-2850 editing decks to produce high-impact news packages for airing around the clock. That eliminated the need for film processors on the premises. (But the facilities architecture changes required by that technology-driven development are road blocks to the needed next wave of change.) The ad revenue-generating capabilities of these newscasts elevated innovative stations to un-forecast levels of cash flow. Purchase of early "DSP" (Digital Signal Processing) equipment like character generators and Vital Industries "DSFX" (Digital Special Effects) units was justified because they attracted more viewers and permitted regular rate card increases.

This historical recounting of the 1940s through early 1970s evolution of television broadcasting brings us to the threshold of the 1980s era. If we remember the earlier three decades as a bad dream with a happy ending, great! However, it 's not over yet!

The facilities expansions from "picture radio" that were implemented in the 1940s through 1970s decades are the root of problems of plant layout and the routing and distribution system that need immediate "rooting out.".

The ideal technique for remodelling the existing plant is "checker-boarding" -- clearing out obsolete and obsoleting capabilities to permit relocations and simultaneous upgrades of capabilities to be kept. That requires a mixture of vision and courage to scrap equipment and functions that "grew like Topsy" in the halcyon cash cow days. An example is the irreversible move of postproduction editing from linear videotape to non-linear disk-based editing.

Blueprint 2: Planning for "HDTV"

Way back in the late 1970s (actually, only about 15 years ago), "HDTV" first began to be discussed as the next television program production and distribution service which would perpetuate the television broadcasting industry's monopoly on home delivery of entertainment. The timetable for commercial development of the service was placed at about now.

However, broadcasters felt little pressure during the 1980s to prepare their existing facilities for the addition of HDTV production, postproduction and distribution operations. Actually, most producers and broadcasters either didn't think about the issue, or figured they would build a new facility for HDTV when the time was right. "After all," they thought, "Our NTSC channel cash cow is delivering enough 'moola' to permit us to do whatever's necessary."

What a difference a decade makes!

Is there even a single reader of these words that will say them firmly early in 1994? No???

Acquiring even one of the four new program production and/or distribution capabilities itemized in the Abstract can be considered only if it can be integrated into the existing or slightly expanded plant.

How can that be accomplished? Signal routing and distribution capabilities not available in any standard routing equipment or transmission media as of January 1, 1994 must be acquired as the mandatory first construction step.

Blueprint 3: Functionally defining the "future-proof" routing & distribution system

(1) Next-generation routers must accommodate:

Every variation of all existing video imaging formats and all being introduced, plus

Every variation on the growing numbers of program audio channels that require simultaneous real-time handling, plus

Growing numbers of ancillary data signals;

(2) Next-generating routers must switch these three signal group types --

With or without video as the reference signal;

Either individually or as one multiplexed signal;

Either operating under manual or automation system remote control, or internally automatically acting to decoding and execute instructions encoded in "Headers and Descriptors" appearing at the head end of each transmission.

(3) Transmission media and protocols must have specifications and attributes that make the transmission circuits and router cross points transparent to either analog signal bandwidths or digital bit rates.

Blueprint 4: Acquiring SMPTE 259M products

The first step toward realization of a simple and affordable system solution to these routing and distribution needs was taken with the issuance of SMPTE Television Standard 259 M, "10-bit 4:2:2 component and $4f_{SC}$ composite digital signals serial digital interface." This document's normative references include standards for converting analog composite and component video signals to parallel digital representations.

Other SMPTE and AES (Audio Engineering Society) standards and practices define processes for digitizing audio signals, and multiplexing parallel streams of serial video, audio and ancillary data signals into one serial digital bit stream.

SMPTE 259M Section 10, "Levels of operation," defines four "Support levels" in terms of specific bit rates:

Level A -- 143 Mb/s, NTSC
Level B -- 177 Mb/s, PAL
Level C -- 270 Mb/s 525/625 component
Level D -- 360 Mb/s, 525/625 component

This range of rates establishes the minimum range of transmission rates to be accommodated in the video signal section of the "future-proof" routing and distribution system defined in Blueprint 3.

Blueprint 6: Defining the maximum "future-proof" bit rate range

Level D anticipates that there will be a demand for television programming produced at Level C's 525 or 625 line raster vertical density, but with the 16:9 aspect ratio specified for the advanced television (ATV) higher vertical line density.

The FCC initially ruled that such signals could not be broadcast on the "second channels" allocated to broadcasters for ATV operation, with Faroudja "line doublers" operating in receivers.to provide subjectively perceived "better" vertical resolution. That stand has now been masked in a "fuzzy cloud" of possibly permitted digital signal transmission rates.

A SMPTE technology committee Working Group has been chartered to define the serial digital interface wave form and bit rate for a single ATV format now being designed by a "Grand Alliance" of former design competitors. That signal's uncompressed bit rate, including eight program audio channels and yet unspecified numbers and types of ancillary data channels, will be in the range of 1.0 to 1.5 Gb/s.

However, the digital rate over the 6 MHz "second channel" has at least tentatively been established at 20 Mb/s. Routing and distribution systems must therefore also function at that rate to meet the design requirement of "future-proof."

Three other independently developed ranges of bit rates dramatically lower the lowest bit rate that must be accommodated in a "future-proof" routing and distribution system. "Full-motion" color videoconferencing systems operate at telecommunications industry standard "T1" and "sub-T1" bit rates (384 kb/s through 1.544 Mb/s.

The cable industry's "500 channel" distribution system capacity goal creates bit rates for individual television channels upwards of 1.5 Mb/s.

The telecommunications industry initially provided the television industry with "DS-3" transmission services that operate at 44.734 Mb/s.

The computer industry's Video Electronic Standards Association (VESA) has implicitly defined a range of computer graphic work station interface bit rates by issuance of its "VDMT 75HZ Monitor Timing Standard." This document (obtainable from VESA in San Jose CA - call 408-435-0333 or fax 408-435-8225) directly establishes timing standards for monitors with 640x480, 800x600, 1024x768 and 1280x1024 pixel raster resolutions, refreshed at 75 Hz.

All these bit rates taken together establish that a television plant's "future-proof" routing and distribution system must transmit signals at any bit rate between at least 1.0 Mb/s and 1.5 Gb/s. To be "far-future-proof," the range should probably be from 64 kb/s (the telecommunications industry's "DS-0" standard rate for digitized voice bandwidth transmissions) to 2.5 Gb/s (a rate bruited about as important by television and telecommunications and computer industry future-thinkers).

Blueprint 7: Accommodating incompatible signal waveforms and strange signal groups

SMPTE 259M very precisely specifies the interface waveform parameters required for transmission of Section 10's Level A through D signals. SMPTE's Working Group S17.13 will incorporate those parameters in its forthcoming ATV signal serial digital interface standard.

However, bit rates established by other industries' standard-setting bodies, have waveform shape and jitter specifications which often are totally incompatible with SMPTE 259M. These signals must be transformed into its Section 3 "Signal Levels and specifications" for transportation throughout a television plant.

Missing Link 1 is a product to perform that transformation. The most urgent waveform conversion need is a unit operating at the DS-3 signal's nominal 45 Mb/s rate. The telecommunications industry standard for "DS" hierarchy digital signals is a tri-level waveform with levels of +1, 0 and -1. This symmetrical waveform was adopted several decades ago to enable transmission of the ostensibly digital signal through transformer-coupled networks. The SMPTE 259M waveform is NRZI binary.

Missing Link 2 is a digital data storage system that operates at 45 Mb/s. An incoming 45 Mb/s signal may be a compressed ATV signal, or a compressed NTSC or PAL signal, or a group of substantially compressed NTSC or PAL TV channels, all with accompanying audio. A 45 Mb/s data recorder eliminates the need to transcode the compressed and/or multiplexed signals back to their basebands, and record them on separate tapes, each time the 45 Mb/s signal has to be temporarily stored.

Until these products become available, DS hierarchy digital signals must be transcoded back to analog baseband, and then re-coded into SMPTE 259M Level A through D serial digital waveforms, before in-plant routing can be effected. The cost and nuisance of this temporary solution to a growing routing need should pressure users into pressuring manufacturers to produce the Missing Links pronto!

Blueprint 8: Locating multi-format operating functions to maximize productivity and quality

Until the appearance of SMPTE 259M serial digital signal interface products, analog NTSC had to be the norm for routing video signals without restriction around a television plant. Separate levels of routing had to be provided for each audio, intercom, machine control, test, et al signal. Facilities were physically arranged to set up isolated "digital islands" when equipment had to be interconnected via parallel digital cables. Current-generation routers (with 100 MHz-plus broadband switching electronics) may be able to handle SMPTE 259M serial digital bit streams in the initial stages of converting a plant for single format all-digital operation today. But multi-format operation is necessary to compete in the maelstrom of tomorrow! A budget commitment for a single-level router with a minimum 10:1 bit rate range from 100 to 1000 Mb/s is therefore an inescapable and immediate investment requirement.

The next step is designing an all-digital distribution system to interconnect the digital islands to each other, via the new router, from their current locations, using existing and new coax cable runs. Suddenly it will be discovered that the task is impossible, especially if some of the digital islands, like an editing room or tape room, are to include ATV equipment. A widely known characteristic of coax cable used to transport high speed digital signals is called the "cliff effect." [1]

In his paper Dr. Hood summarized the results of a transmission test of a 143 Mb/s digitized NTSC signal over 8281 coax cable:

> *For cable lengths of less than 350 meters the bit error rate (BER) is literally unmeasurable, at 370 meters the BER is just in the acceptable range (BER of $1x10^{-12}$ - about 1 error per hour) and by 390 meters the BER is completely unacceptable (BER of $7x10^{-9}$ - about 1 error per second). The implication of these test results is that signal regeneration and repeatering is necessary every 350 meters along a coax cable transmission path in order to achieve an acceptable bit error rate.*

The costliest way to eliminate the problem is to relocate all the digital islands closer to the router, with no tandem legs totalling more than 350 meters unless regenerating circuits are installed in the router. And that's also costly.

As-needed substitution of optical fiber transmission links on longer cable runs, or on runs through hostile EMI, RFI or wet or chemically noxious environments, is a technically feasible alternative.

Blueprint 9: Specifying a hybrid coax/fiber routing and distribution system

Hybrid routing systems accommodating both coaxial cable and fiber cable inputs and outputs have demonstrated in prototype form in the last year. At the time this paper was written, the author was not privy to exact details of the capabilities and designs of several hybrid routers being readied for 1994 introduction, presumably at this NAB exhibit.

The ideal hybrid coax/fiber router will be of modular design, with interchangeable 1 by N coax and fiber input and output modules field installable by the user. Crosspoint switching and provisions for N inputs or outputs would be carried out at serial digital electrical baseband. Switching automation and remote control provisions should be identical to all coax/single format routers offered in the past..

There must be no technical or operational restrictions on tandem intermixture of coax and fiber transmission circuits. Tandem fiber circuits must have sufficient loss budget to permit tandem extensions of router-originated or -terminated links via fiber patch panels and/or the equivalent of coax circuit "barrels." Both the fiber and connectors and all installation accessories must be available from multiple sources and be freely intermatable.

Optical transmitting and receiving equipment from each supplier source must permit its operation without restriction with complementing equipment from all other vendors.

Blueprint 10: Design options for optical fiber transmission links

In March 1993 the SMPTE recognized the need for such a standard for bit-rate-independent serial digital fiber transmission links. The S17 Television Signal Technology Committee established a Working Group on Serial Digital Fiber Interfaces, appointing this paper's author Chairman. A proposed standard document was circulated for balloting and technical and editorial comment in January 1994, and reviewed and modified in February.

Fig. 1 details the draft standard proposal's specifications for the bit rate range signals to be accommodated in both the optical and electrical domains within the router, and the types of fibers which can be installed or integrated into a television plant (premises) distribution system. These specifications harmonize with those in telecommunications industry standards and practices for "premises wiring."Fig. 2 specifies transmission link design options and performance specifications for circuits terminated in telecommunications industry standard light sources and receivers. The specifications permit competitive choices among the equipment and components offerings of world-wide vendors with long track records of service to telecommunications industry customers.

The paper as presented will conclude with a summary of changes incorporated in the proposed specifications, as a result of its balloting and voting returns review under due process rule established by ANSI.

Fig. 1. Bit rate throughput and fiber type options **

Throughput(Mbps)	Signal description	Specification source	Fiber Type (Note 1) Window I nom 850 nm	Window II nom 1310 nm	Window III nom 1550 nm
1.0	Nominal lower cutoff	None	MM	MM/SM	SM
1.544	DS-1	ANSI T1.408 1990 edition	MM	MM/SM	SM
44.734	DS-3	ANSI T1.404 1994 edition	MM	MM/SM	SM
143.000	NTSC 525	SMPTE 259M Support level A	MM	MM/SM	SM
177.000	PAL 625	SMPTE 259M Support level B	MM	MM/SM	SM
243.000	525/625 component 8 bit	SMPTE 259M Support level C	MM	MM/SM	SM
270.000	525/625 component 10 bit 4:3	SMPTE 259M Support level C	MM	MM/SM	SM
360.000	525/625 component 10 bit 16:9	SMPTE 259M Support level D	MM	MM/SM	SM
400.0	Nominal upper cutoff	None	MM	MM/SM	SM
To 2,500, at designer option	Includes ATV & high resolution digital graphic formats	Pending	Not applicable	SM	SM

<u>Notes</u>

1 Transmission of signals on Multi-mode fiber at up to a 400 Mbps rate may be possible at distances less than 1 km. Above this distance, Single-mode fiber shall be required.

** The degree of compliance of equipment designed to this standard shall be indicated by appropriate nomenclature in the equipment specifications. The description shall include the range of throughput bit rates, the transmission window, and the fiber type, for example: "Throughput range - 1-200 Mbps, Light wavelength - 1310 nm, Fiber type - Multi-mode." This indicates a transmission link which will meet all end-to-end performance specifications while transmitting serial digital DS-1, DS-3, NTSC and PAL signals, per SMPTE 259M waveforms, on Multi-mode fiber, over maximum fiber circuit lengths listed elsewhere in the specification.

1 Two excellent tutorial papers have been written by Leitch Corporation's Robert Proulx, *"Digital Systems: A beginner's guide,"* SMPTE 1993 Advanced Television Conference, New York NY, February 1993, and Consultant Dr. James Hood, *"Routing serial digital television signals,"* SBE/NAB 1993 Engineering Conference, Las Vegas NV, 20 April 1993.

Fig.2 Transmission Link Design Options

Parameter	Transmission Link Options			
	Multi-mode Links		Single-mode Links	
Nominal operating wavelength nm	1310	850	1310	1550
Minimum center wavelength nm	1270	800	1270	1520
Maximum center wavelength nm	1330	880	1330	1580
Spectral width (Maximum) nm FWHM (Full Width, Half Maximum) RMS	170	70	4	4
Data rate (NRZ-I signal) nom Mbps (Waveform per SMPTE 259M)	1 to 270	1 to 100	20 to 1200	20 to 1200
Maximum fiber attenuation dB/km	1.5	3.75	1.0 (Note 7)	1.0 (Note 7)
Transmission capacity (Min.) MHz-km	500	160	NA	NA
Light source	LED	LED	Laser	Laser
Minimum optical loss budget dB (Note 1)	10	8	11.5	11.5
Tx optical power range (Note 3) dBm	-12/-15	-11/-14	-7.5/-9.0	-7.5/-9.0
Output power stability +/- dB	N/A	N/A	1.0	1.0
Extinction ratio (Min/Typ)	N/A	N/A	5:1/10:1	5:1/10:1
Optical rise time (Note 4) (Max.) ps	1100	3000	330	330
Optical fall time (Note 4) (Max.) ps	1100	3000	330	330
Receiver photo detector	InGaAs PIN	Si PIN	InGaAs PIN	InGaAs PIN
Rx optical input pwr max Note 3 dBm	<-12	<-11	<-7.5	<-7.5
Rx optical sensitivity(Note 1,3) dBm	>-25	>-22	>-20.5	>-20.5
Electrical output - Amplitude mv	800	800	800	800
- Rise/fall time (Note 5) ns	<0.75	<0.75	<0.20	<0.20
- Maximum bit error rate (BER)	$1x10^{-12}$	$1x10^{-12}$	$1x10^{-12}$	$1x10^{-12}$
- Random jitter (Note 6) ps	<500	<500	<60	<60

Explanatory notes

1 These specifications must be met with the link transmitting any applicable compliant bit rate, and operating over the full temperature range with a Bit Error Rate (BER) of less than $1x10^{-12}$
2 Reference SMPTE 259M Section 4.1, "Connector and Cable Types" for connector details
3 Average optical power for a square wave input at 50 MHz, coupled into applicable fiber
4 Measured at the Transmitter (Tx) optical output, between 20% and 80% points
5 Measured at the Receiver (Rx) electrical output, between 20% and 80% points
6 Root Mean Squared (RMS) value, measured with a 100 MHz square wave input
7 For ISP (Inside Premises) cable; OSP (Outside Premises) maximum attenuation is 0.5 dB/km

SBE

AM/FM IMPROVEMENT

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TECHNIQUES TO EXTEND THE SERVICE LIFE OF HIGH POWER VACUUM TUBES

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ABSTRACT

High power vacuum tubes are a major expense in the operation of many radio stations. In 1990, the Voice of America began a program to extend the service life of its high power vacuum tubes. Voice of America has documented annual tube savings of over $300,000 at one station alone. Network wide, the savings have been much greater.

Major factors in tube life are filament voltage management and Off/On cycling. Other factors are black heat and inrush current

Operational Techniques

Minimize Filament Operating Voltage

This procedure is used to set the filament operating voltage for the power tube to a value that will increase the service life without reducing the power output or the quality of the transmission in any way.

In all cases where filament voltage is mentioned, the parameter to be controlled is actually the filament temperature. The near impossibility of measuring filament temperature in an operating power tube makes the use of filament voltage as a surrogate for temperature a necessity. For the purists, filament power is a somewhat better surrogate than is voltage. However, filament power takes time to calculate. Still, it is important to keep the temperature of the filament in mind, particularly in the case of tubes such as the former Machlett tubes that operate with filaments 50 to 100 degrees Centigrade hotter than most power tubes.

The filament voltage minimization procedure begins by operating the tube at the recommended filament voltage for 200 hours. This preliminary conditioning gets the filament through the initial aging process. During this initial "burn in", the tube parameters tend to change rapidly as the thoriated and carbonized tungsten filament wires become fully activated. The filament emission will often double during this period.

Next, the filament voltage is reduced in one or two percent decrements until there is a noticeable increase in distortion, a decrease in the grid, screen, or plate currents, or a decrease in power output. Distortion is usually the most sensitive indicator. Grid current is also a very sensitive indicator because the grid conduction angle is very narrow with most of the current drawn during the peak of the plate current. Once the first indication is encountered, the filament voltage is then increased back to the previous setting plus another percent to allow for tube aging. The filament temperature needs to stabilize after each change in voltage.

The above procedure must be performed for an AM transmitter with the modulation at the highest level used during program transmission.

In transmitters where several power tubes are operated in parallel, the distortion contribution of a single tube will be diluted by the other tubes and this reading will be desensitized proportionately. If the power tubes have separate grid current metering, the grid current will likely be the most sensitive reading to monitor. Multiple tube class AB_1 modulators will therefore have to be monitored with screen and plate currents or power output. Push-pull modulators may, in fact, experience a decrease in distortion as a result of the reduction of the filament voltages. This is probably due to the modulator tubes becoming more nearly alike (balanced dynamically) and cancelling the even harmonics more effectively. Obviously, the tube with the highest peak emission will operate at a lower filament voltage than its mating tube, and the final result will be a better dynamic matching of the tubes. This is an additional bonus and will not cause any ill effects so long as the output power and currents have not been reduced. At the VOA Bethany (Ohio) Relay Station, a reduction in distortion of about five Db (from 2.5% to 1.4%) was experienced after reducing the filament voltages of 4CV100,000C modulator tubes in the Collins 821A-1 transmitters.

A close watch should be kept on the tubes for the next few days to check for changes in the above operating parameters. The procedure should be repeated about once a month or if any adverse changes do occur.

A dramatic increase in the emissive life of the filaments can be realized if the tube has enough excess emission to permit a reduction of five percent or more in the filament voltage. For each three percent reduction in filament voltage, the emissive life of the filament will about double. A five percent filament voltage reduction will therefore more than triple the emissive life.

OFF/ON Filament Cycling

When a cold filament is turned on, damage is caused by two effects: (1) The current inrush into a cold filament can be up to 10 times the operating current if the filament supply is of very low impedance; (2) Grain reorientation occurs at about 600 to 700 degrees centigrade; this is called the Miller-Larson Effect. The grain reorientation will result in a momentary plastic state that can cause the wire to grow in length and therefore become thinner.

To illustrate the first effect, one can use an ordinary 60 watt light bulb. The light bulb should operate at 0.5 amps at 120 volts. This translates to a hot resistance of 240 Ohms. An Ohmmeter reading of a room temperature 60 watt light bulb will be less than 20 Ohms or, a hot to cold ratio of greater than 12. Power tube filaments will have about a ten to one hot to cold ratio because they operate at a lower temperature than light bulbs.

The second effect is aggravated by variations in the cross sectional area of the filament wires along their length. This will cause "hot spots" in the thinner cross sections due to the greater current densities. The higher temperatures at the hot spots cause increased growth during the warmup through the 600 degree Centigrade temperature when they have a higher current density than the rest of the wire, and a much higher power dissipation per unit length due to also having a higher resistance. Each time the filament is turned on, the wire becomes thinner until the hot spot temperature gets into a runaway condition. During the runaway, the filament temperature reaches the melting point during the turn on surge, and the light bulb (or filament) produces the familiar flash bulb effect that signals its demise.

Light bulbs nearly always burn out during the turn on inrush surge. Each Off/On cycle removes several hours of life from the light bulb (or filament) and this trauma is approximately proportional to the cube of the inrush current. By limiting this inrush current or, by preferably eliminating it, the life of the power tube filament can be lengthened several times.

The Egyptian Ministry of Information operates a two megawatt transmitter in the medium wave band. This Doherty transmitter uses four tubes, two carrier tubes, and two peak tubes working through a combiner. Even though the transmitter was operating more than 22 hours per day, the filaments were turned off and back on every night. The very expensive tubes (about $100,000 each) were lasting only 5,500 hours, and a stock of 10 spares was expected to last 18 months for the four sockets. After leaving the filaments on continuously and rotating the spare tubes annually, the station has yet to have a tube failure after seven years.

In Greenville, North Carolina, VOA has four 500 kW transmitters with vapor cooled tubes. The transmitters are used for 12 to 15 hours per day. Initially, the filaments were turned off and on two or three times per day during gaps in the schedules. Three years ago, VOA began leaving the filaments on at all times except for major maintenance. As a result, the expenditures for tubes for these transmitters has dropped from about $420,000 per year to under $100,000. This was done at a cost of $15,000 per year in added electrical power cost. For every extra dollar spent on added filament power, we had a return of more than 20 dollars in reduced tube costs -- eat your heart out Wall Street!

Calculations for both of the above situations indicate that each Off/On cycle of the filament was reducing the life expectancies by over 75 hours. An unexpected result was that it appeared to be independent of the filament construction - straight wire, hairpin, and mesh filaments all benefitted nearly equally by leaving the filaments on continuously. The mesh filament was expected to suffer more from Off/On cycling because of flexing of the many welds. Also, the mesh filament structure will grow in overall diameter as the wires lengthen. It is constructed in a manner similar to a Chinese finger trap which changes overall diameter as the axial length is changed. This mesh filament construction causes an additional failure mode because the overall mesh diameter can grow enough to cause a grid-to-filament short before a wire breaks, and the diameter tends to grow more at the bottom of the filament due to gravity. In many large power tetrodes, the grid to filament spacing is less than two percent of the overall filament diameter.

Measuring Mu to Predict Filament-Grid Shorts

As the filament wires lengthen and our Chinese finger trap mesh filament grows in overall diameter, this change in the tube geometry will cause an increase in the Mu or amplification factor. This is aggravated by gravity which causes added growth at the bottom of the filament.

Radio station HCJB of Ecuador successfully predicted filament to grid shorts by periodically measuring the amplification factor. A gradual increase in Mu will be noticed as the tube ages. As the **rate of change** increases (after at least 50,000 hours, it is hoped) to five or ten times the earliest rate, the short condition is imminent and the tube should be changed to prevent lost air time.

Black Heat

Discussions with tube designers have suggested that the "Black Heat" or stand-by filament voltage should be between 25 and 40 percent of the rated filament voltage. This range will produce a temperature high enough to keep the filament above 600 degrees centigrade area where the Miller-Larson Effect occurs and sometimes low enough so that forced cooling is not required during stand-by.

Operating histories of several "Black Heat" installations indicate that the trauma of raising the filament voltage from 25 percent up to the operating voltage is still damaging the filament. For this reason, the stand-by voltage should be "Orange Heat" or about 80 percent of the full rated voltage. At this voltage, the emission life is expended at about one percent of the full filament voltage rate and very little trauma is experienced by raising the filament voltage from 80 percent to 90 or 95 percent of the rated voltage. The voltage range of 40 to 80 percent is to be avoided because, in this range, the filament can act as a getter and absorb gases that will raise the work function of the thoriated tungsten filament.

With respect to anode cooling, large water cooled power tubes may create enough water flow due to thermo syphoning to prevent overheating the tube during stand-by. Vapor cooled tubes will automatically reduce cooling requirements because less make up water is required.

Graphical Representation of Filament Life

The graphs shown in Figures 1 and 2 dramatically illustrate the effects of filament voltage reduction and the reduction of Off/On cycling.

The top curve of Figure 1 assumes that, if the filament were operated at the rated voltage and never turned off, the filament would last for 30,000 hours. It further assumes that, for each three percent reduction of the filament voltage, the emissive life of the filament will be doubled (and, as with Db, a 10 percent reduction will produce a ten fold increase in the emissive life).

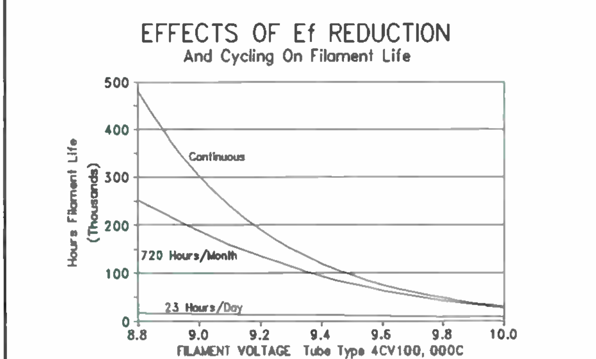

Figure 1

Note that the emissive life begins to increase dramatically with filament voltage reductions greater than five percent. Filament voltage reductions greater than 10 percent will generally eliminate the filament as a cause of tube failure.

The next two curves make the further assumptions that an Off/On cycle will remove 60 hours of life from the filament **at the rated filament voltage.** It amounts to a 0.2 percent reduction in the life of the tube, another way of saying that the filament is good for about 500 Off/On cycles **at the rated filament voltage** before burnout. The lowest curve shows the effect of turning the filament off every day (this corresponds to the original Egyptian operation). The life of the filament is determined mostly by the number of Off/On cycles because they account for about 72 percent of the filament damage (i.e., 60 hours lost because of filament Off/On cycling and 23 hours lost during the 23 hours operating time).

The 23 hours per day curve shows increased life from filament voltage reduction because the damage due to the inrush current is proportional to the cube of the inrush current. This current is reduced in almost direct proportion to the applied voltage. A 12 percent E_f reduction will reduce the damage to 0.88^3 or to about 40 hours (at the rated voltage) and increase the emissive life by a factor of 16. These two effects reduce the rated daily filament life consumption from 83 hours (60 plus 23) to about 41.5 (40 plus 1.5) hours and will about double the filament life. The reduction of over 20 percent at Greenville should more than triple the filament life - over double from lowering the filament voltage and the rest from the higher impedance caused by the autotransformer and the increased emissive life.

The difference between the lines representing a monthly Off/On and a daily Off/On is indicative of the importance of keeping power on the filaments during most non-operating periods.

There are probably two separate "banks" of filament hours that are separately drawn upon. One "bank" contains several hundred Off/On operations and the other contains a few tens of thousands of emission hours. The above analysis treats these banks as one and makes withdrawals from it according to the equivalent damage done by Off/On operations and filament operating time.

The truth is somewhere in between these methods. Off/On operations appear to also reduce the emissive life, and higher filament voltages appear to also reduce the total number of Off/On operations available to the filament.

To determine the degree of independence of these two failure modes is beyond the scope of this paper. It is important to realize that Off/On operations are very damaging to the filament, and high operating temperatures use up emission hours needlessly. Both conditions can be minimized to obtain extended tube life.

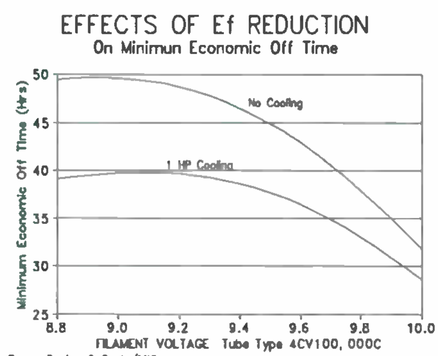

Figure 2

Figure 2 shows the off time required to regain the tube value lost during an Off/On cycle. At rated filament voltage, the tube will lose about 60 hours of life each time power is applied to the filaments. If the lost tube life is valued at $20, it will take less than 60 hours to regain this loss because of the savings in filament power. As the filament voltage is reduced, there are more hours lost at the extended hours of life we are then considering (but less at the rated voltage expected hours). If filament voltage is reduced by more than 10 percent, the reduced dollar, or percentage, damage to the filament becomes more important. The upper curve applies to cases where there is no cooling power needed and the lower curve to a one horsepower per tube cooling requirement. The latter curve peaks at a somewhat higher filament voltage.

Both curves indicate that is it more economical to leave the filaments on during standby periods of well over 24 hours. The savings in tube costs more than offset the cost of filament power.

In the above illustration, the curves were calculated based on a power cost of 9 cents/Kwh. Higher power costs and cooling needs will lower the curves and lower rates will raise the curves.

Circuits to Adjust Operating Voltage

Resistive Circuits

Placing a resistor in series with the power tube filament has the effects of both reducing the operating voltage and limiting the inrush surge current.

We provide a sample analysis

of a filament voltage reducing circuit based on the following assumptions:

1. The transmitter uses an Eimac 4CV100,000C power tube.

2. The surge current is limited by a filament supply with 25 milliohms internal impedance and open circuit voltage of 17.5 volts.

3. The application allows for a 12 percent reduction of E_f. (The 4CV100,000C is rated for a minimum peak cathode current of 165 amps, the Collins 821A-1 transmitter has a peak cathode requirement of about 80 amps at the carrier level and nearly twice this value at maximum modulation. The 12 percent E_f reduction will reduce the minimum peak current capability of the 4CV100,000C to 99 amps but the actual peak emission may be much higher.)

4. Damage to the filament during an Off/On cycle is proportional to the cube of the inrush current.

5. Each three percent reduction in filament current will double the emissive life of the filament.

At the rated setting of E_f = 10.0 volts, the surge current through the total of 28.33 milliohms will be about 618 amps. This is the open circuit voltage of 17.5 volts divided by 25 milliohms of power supply impedance plus the 3.33 milliohms cold resistance of the filament.

Adding 8 milliohms to the filament circuit will reduce the operating E_f to 8.8 volts and the operating I_f to 296 amps. The resulting surge current is 482 amps. Note that the hot resistance of the filament has dropped from 33.3 milliOhms to 29.7 milliOhms.

An installation of the above circuit will increase the emissive life of the tube by 16 times (almost half a million hours). The installation will further enable the tube to withstand more than twice the number of Off/On operations.

It may be more convenient to use a resistor of higher resistance value, but the same power rating in the primary of the filament transformer. The resistance value will be multiplied by the square of the filament transformer turns ratio.

If the operating voltage can be reduced by only seven percent, then a filament rated to operate at 10 volts and 300 amps would be in series with a 4.4 milliohm resistor. The resulting 298 amp filament current would then reduce the filament voltage by 0.70 volts and limit the maximum inrush current (assuming the same 25 milliohm supply impedance and cold filament resistance) to about 535 amps instead of 618 amps.

The cube relationship of trauma to current will enable the filament to last almost 50 percent longer because of Off/On operations, and the doubling of life every 3 percent voltage reduction will increase the emissive life about 400 percent or a factor of five. Greater voltage reductions will, of course, enhance both life aspects.

Inductive Circuits

Another method of reducing filament voltages is to use a bucking transformer in the primary of the filament transformer. The bucking transformer may be controlled by a variable transformer to reduce the filament voltage in the range of 0 to 10 percent. Another 10 percent bucking transformer could be switched in during stand-by operation.

In the case of several type 5682 power tubes used in VOA's Continental 420A 500 Kw transmitters in Greenville, the filament voltage was reduced by more than 20 percent through the use of tapped autotransformers for each of eight tubes. We expect the emissive life to be no longer a factor in the tube life of these transmitters. We also expect the filaments to withstand nearly three times as many Off/On cycles as they did at the rated filament voltage. Each transmitter contains eight type 5682 tubes and four type 5681 tubes. Because these former Machlett (now Eimac) tubes operate very hot filaments at their rated filament voltages, a reduction of greater than 20 percent did not cause filament gettering. Voltage reduction for these filaments was obtained by using custom-built autotransformers having taps at 100, 98,96,...82, and 80 percent. An additional one percent tap was provided at the bottom of the transformer to interpolate between these values. The cost of each transformer, case, meters, and switches was about $800. This was a relatively small cost to more than triple the life of a power tube costing over $13,000. Payback period is estimated to be about 600 hours of operation.

In Greenville, the Continental 420A transmitters have cooling requirements of 115 horsepower (or 85 Kw) so it is generally impractical to have stand-by conditions for long periods. There are plans to install smaller pumps and blowers to reduce the stand-by cooling power to less than two horse power per tube, but this has yet to be implemented.

In Thailand the voltage on 18 type 5682 tubes was reduced by about 20 percent. This transmitter had been consuming 10 to 15 tubes per year. VOA now replaces three to four tubes per year. If the filaments were left on continuously at less than 80 percent of the rated filament voltage for this transmitter the tube usage could be reduced to one or two tubes per year.

Potential Problems with Filament Voltage Reduction

Possible gettering by the filament has already been mentioned as a potential problem with filament voltage reduction. The possible gettering, if the filament voltage is in the range of 40 to 80 percent of the rated voltage, will cause the work function of the thoriated tungsten to increase and thereby reduce the emission of electrons. This problem can usually be cured by operating the filament at the rated filament voltage for a few hours to drive the contaminating molecules out of the filament.

Another potential problem is that if the filament voltage is set too low, the electron cloud that normally surrounds the filament can be depleted and expose the thoriated tungsten to ionic bombardment that may damage the filament. This damage is irreversible and can cause early failure of the filament.

The Power Tube as a Permanent Component

Using the above techniques to reduce the filament voltage to less than ninety percent of the rated operating voltage and eliminating off/on cycling is similar to curing cancer and heart disease. Tube life will not increase 10 to 20 times beyond the present life because tubes can still be mishandled, transmitters might be struck by lightning, seals may crack and manufacturer defects will still be found.

However, the service life of a vacuum power tube may be increased to a value that will enable transmitter designers to simply braze the tube in place which would eliminate the tube socket and its problems. No more finger stock problems, shorter RF paths with lower inductance, fewer water leaks - life is good!

In this scenario the vacuum power tube resembles the engine in an automobile, -- it is expected to last the life of the rest of the automobile. In the unlikely case that it fails prematurely (after 10 or 15 years) the extended replacement time is a small price to pay for the tens of thousands of hours of trouble-free operation.

Acknowledgements

Several people have contributed to the techniques presented in this paper. To mention only a few:

Bob Keller of Burle Industries (formerly RCA Power Tube Division). Bob specializes in large cross section trapezoidal filaments.

Marshall Loring of Eimac. General tube knowledge and filament theory.

Sterling McNees of Eimac. He was instrumental in developing mesh filaments for tubes in the megawatt range.

Adil Mina and Dave Russell of Continental Electronics.

Hilmer Swanson and Dick Frey of Harris Allied.

John Sullivan of Econco. Tube rebuilding aspects of filaments. John has a publication on power tubes which has some very good, though conservative, advice on filaments.

Dr. Wolfgang Schmeinke of ABB (now part of Thomson).

Gene Tosti of Vacuum Tube Industries. Tube rebuilding.

Spreadsheet for Calculating the Minimum Economic Off Time

The Voice of America has available a public domain spreadsheet program that will calculate and graph numerous parameters against reductions in filament voltage. Expected filament life, filament power and resistance, power cost, life reductions and lost tube value from off/on cycling, and cooling costs are some of these parameters.

The spreadsheet may be customized for any power tube filament and installation. Values to be entered by the user are rated filament voltage and current, tube cost, power cost per Kwh, cooling power in horsepower and type of off/on operation (ramp, black heat, orange heat, step-start, or switched).

Dozens of graphs may be generated from this spreadsheet including those used in this paper. The graphs will be customized for the particular power tube and operational conditions for which the spreadsheet is customized.

Fields containing the expected loss from the several methods of turning on the filaments, voltage vs. resistance tracking of tungsten, percent voltage reduction to double filament life, range of voltages to be analyzed and graphed, resolution of the voltage range (percent per step) are all accessible for customizing.

A separate portion of the spreadsheet will select one particular filament voltage and produce a more thorough analysis. This portion may be copied an unlimited number of times to analyze several filament voltages, types of off/on, and cooling requirement.

To fill the disk, files of interest are available such as a Smith Chart created using AutoCad that is on four layers and has several unique features making it quite versatile. For instance, the resistive circles and susceptance arcs could be copied, rotated 180 degrees, and plotted in a contrasting color to enable the transition from a parallel to series circuit by merely switching colors. Resistance circles may be copied and rotated to analyze stub tuners. All circles and arcs in the chart are accurate to six figures.

A spreadsheet for planning the filament reduction of any power tube in nearly any transmitter is also on the disk. VOA has used this spreadsheet to plan and predict life extensions for about sixty tube types used in over 140 transmitters in the VOA network.

Lotus 123 and Quattro Pro versions are available on either 3.5" or 5.25" disks by writing to:

Walter Johnson (B/EOTM)
Voice of America
330 Independence Ave. SW.
Washington, DC 20547

Phone: 202 619 2188
Fax: 202 619 2331

A self addressed envelope, mailer, or mailing label will be appreciated.

BROADBAND IMPLEMENTATION OF PHYSICALLY SHORT, HIGHLY TOP-LOADED ANTI-SKYWAVE ANTENNAS

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Abstract

The physically short vertical antenna usually is a high Q narrow bandwidth device. Toploading of short vertical antennas significantly improves the suppression of high angle skywave and at the same time improves the antenna bandwidth somewhat. Installation of a specially designed top loading adjustment network improves the resulting antenna bandwidth considerably so that a symmetrical broadband match is achievable with a simple T network antenna tuning unit. The results of this antenna optimization method as implemented at station KIAM Nenana, Alaska are detailed to illustrate the technique.

The Search For A Superior Short Antenna A Historical Perspective

Since the earliest days of radio the engineers and scientists have searched for a small efficient antenna. At the beginning they fully expected to find a small antenna that was as efficient as the full size 5/8 wave tower. The search went on through the thirties and into the fourties with every kind of structure and loading being tested. Although the tests found that it was possible to dramatically improve the efficiency of a short antenna through loading, it was never quite as efficient as a full size antenna. Because steel was relatively cheap and approval to construct tall towers was relatively easy to obtain, the search stopped and broadcasters were resigned to use full size towers. In the recent past top loading has mostly been used to marginally improve efficiency or bandwidth of a few of the most difficult installations.

New Problems

In the last ten years or so new problems have made use of full size towers difficult. New FAA policies have made it far more difficult to get aeronautical approval for tall towers. Local zoning authorities have determined that towers are an undesireable use of land in their domain. Homeowners want excellent radio reception but no one wants to be able to see a tower. Tower costs have also risen as new building code requirements have made tower designs more complex and difficult. The cost of steel has also increased dramatically.

New Methods

Some breakthroughs have made loaded towers more practical and more desireable. It has been found practical to make the top loading adjustable and to put the adjustment down at the ground level so that future adjustments do not require climbing the tower. This ground level adjustability allows significantly more creative optimization of the tower both with respect to efficiency and bandwidth.

The short antenna has always been considered a narrow bandwidth device but this need no longer be true. Top loading has always been known to improve the bandwidth of a short tower somewhat but now we can extend the bandwidth as far as we wish. In order to understand the magnitude of the breakthrough, first I should explain that the most important effect of adjustable top loading on bandwidth is that it makes the tower base impedance characteristics adjustable. However the best bandwidth base impedance occurs with a different absolute amount of top loading for each frequency. By simple trial and error adjustment of the top loading coil you can get a nice base impedance for your operating frequency but to achieve a broader bandwidth it will be necessary for the top loading adjustment to vary with frequency. A simple move of the loading coil tap can optimize the impedance for any single frequency but since mechanical adjustment cannot optimize more than one frequency at a time a more complex loading network is needed to bring about the desired tower base impedance over a wider span to achieve true broadbanding.

KIAM A Case Study In Top Loading And Broadbanding

KIAM radio on 630 kHz in Nenana, Alaska is a station limited by aeronautical considerations to a 290 ft tall tower. This is only 68 degrees electrical height. In order to get maximum coverage the tower was top loaded with an insulated adjustable top hat.

First picture a tower with an adjustable loading coil. Mount the loading coil remotely at the base of the tower to adjust the top hat loading. We adjusted the coil taps for best radiation efficiency on the operating frequency of 630 kHz. This resulted in a excellent low Q impedance of 30 -j7 Ohms. At 620 kHz the tower is electrically shorter resulting in a lower radiation resistance and a higher reactance. The top hat capacitance also causes less current to flow resulting in less top loading. All of these factors result in a base impedance of 27-j27 Ohms at 620 kHz. In order to correct for this the adjustable loading coil needed to increase the inductive reactance as the frequency is lowered. At 640 kHz the tower is electrically longer and has a naturally higher radiation resistance and a lower reactance. The top hat capacitance causes more current to flow resulting in greater effective top loading. All of these factors result in a base impedance of 34 +j10 Ohms at 640 kHz. In order to correct for this the adjustable loading coil needs to provide less inductive reactance with increasing frequency. A parallel tuned circuit changes reactance in the right direction with frequency to meet this requirement but a parallel tuned circuit with the right slope to match the tower characteristic would have rather high circulating currents. Since no capacitor was conveniently available with large enough current rating to handle the circulating current we decided to use an inductor and a 90 degree phase shift network to do the same thing. From available components we were able to construct a suitable loading network that improved the sideband impedance to 27.3-j19 Ohms at 620 kHz and 34 +j6 at 640 kHz. This relatively simple antenna loading network reduced the reactance swing in the tower by 40%. With a simple T network antenna tuning unit made from parts on hand and several hours of fine adjustment the coax input impedance was brought to 50 +j0 at 630 kHz, 42 +j2 at 620 kHz, and 42-j2 at 640 kHz. The old Collins 21E transmitter loved it and showed its approval with noticeably improved modulation capability and audibly lower distortion. When the new solid state transmitter was installed the reflected power meter sat on the bottom even with the densest music selections. The classical music hour Sunday afternoon sounds very clean and transparent with the same processor settings that sound best for the talk programming .

Results And Outlook For The Future

In summary, the short antenna was loaded to achieve a measured 1 dB field strength increase, the loading network was redesigned to improve the bandwidth of the tower, and an antenna tuning unit of conventional design was modified and adjusted to provide the transmitter a symmetrical VSWR of 1.35 at the 10 kHz sidebands. This is truly a very nice combination of results especially when you consider that it was possible with parts on hand at the site. Further improvement certainly can be achieved with purchase of additional components with the optimum values.

The future potential of such broadbanding efforts are not limited to the 10 kHz sideband area but can be applied even to optimizing a single tower for two or more operating frequencies in a diplexed situation. From now on antenna bandwidth for short top loaded towers will be limited only by our willingness to persue it.

A TUNEUP GUIDE FOR AM BROADCAST ANTENNA SYSTEMS

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ABSTRACT

Many engineers are not acquainted with or have reservations about making repairs to their AM antenna systems. A properly operating antenna is essential to complying with FCC Rules and Regulations. A station's ability to serve its marketplace is mainly governed by the amount of RF signal delivered to the listeners. This paper will relate to the reader via experiences with actual arrays some tips on how to identify and rectify typical problems encountered with nondirectional and directional systems.

PARAMETERS

The most important information about an array's performance is delivered by the sampling system. If a deviation from normal operating parameters occurs, it is important to have confidence in the sampling system. On too many occasions, the phasing system has been adjusted to compensate for a malfunctioning sample system.

Sample systems can take many designs. The three most common sample systems can be categorized as follows:

Loop Sample
Base Sample
Feedpoint Sample

Loop sampling delivers a sample from the current "loop" on the tower, the point of maximum current. Base sampling delivers a sample from the base of the tower via a rigid loop. FCC 73.68 mandates a minimum height above ground for sample loops of three meters. Feedpoint sampling delivers a current sample from the ATU output terminals using a shielded toroidal transformer.

Most FCC "approved" sample systems utilize solid outer sheath coaxial cable. Some systems consist of equal length sample lines for all towers. When all lines are of equal length, there will be no phase difference in the sample system. Where lines are of unequal length, the indicated phase angles must be mathematically adjusted to compensate for the unequal delays in the sample system transmission lines.

SAMPLE SYSTEM MAINTENANCE

If you are installing a new sample system or wish to determine if an existing system is operating properly, follow the checklist below:

1. For sample loops, carefully inspect the coaxial connector. Be absolutely certain all connections are free of corrosion and are tight. If any corrosion is present, replace the connector. Clean and tighten all connections on the loop.

2. For insulated sample loops, measure the resistance across the insulator, preferably with a megger. Insure there is no leakage resistance. Clean the insulators.

3. Open the far end of each sample line. Use a megger to check for unwanted leakage. If any leakage is present, check the RF connectors on each end of the line. Be certain you have a reattachment kit to weatherproof the outdoor fittings.

4. Measure the sample line resistance with the far end shorted. Be certain there are no high resistance conditions.

5. Measure the electrical length of each sample line and feedline using an RF oscillator and oscilloscope. This method is preferred over a TDR or other extravagant means unless the station owns the instrument. Keep a record of the measured data for future reference.

Be careful when using a TDR on a line connected to a sample loop. A TDR reading of longer than expected length can be an indication of leakage between the inner and outer conductors. It is always best to have a competent tower rigger disconnect the far end of the sample line to perform the above tests.

I have documented one case where a sample line on a tower known to be exactly 200 feet long measured 260 feet on a calibrated TDR. Opening the loop end of the line, 30,000 ohms of leakage was present on the cable. The problem was found to be water contamination along a 20 foot length of the line. The entire length required replacement.

Another source of sample or RF feedline problems are the connectors. Moisture and contamination can enter the line if the connector is not properly weatherproofed. I have witnessed on several occasions arcing and burning due to improper installation of end connectors, particularly poor outer conductor connection. In one instance the installer did not peel back the outer sheath. This caused excessive arcing and the connector eventually burned up inside.

Bonding RF and sample lines to the tower is extremely important. Loop sample lines should be bonded to the tower leg immediately below the loop and at least every ⅛λ thereafter and lastly at the tower base. The same procedure applies to FM and RPU transmission lines. Isocouplers must be bonded to the tower base leg and ground strap immediately after the input and output connections. Failure to properly bond transmission lines can adversely affect the tower's self and drive point impedance.

As an example of how bonding can affect base impedance, a tower with an electrical height of 190° has a sample loop located 200 feet above its base with the sample line bonded at the base and at the loop "N" connector. The tower measures 43 -j199 in this configuration. Placing two additional bonds on the line alters the base impedance to 63 -j195.

Sample loops placed at tower potential require an isolation coil to cross the base insulator. Most isolation coils are wound from ½ inch coaxial cable. Isolation coils are usually parallel resonated using a capacitor. The simplest method for resonating an isolation coil is to first measure the tower's self impedance with the isolation coil disconnected. With the isolation coil connected, tap the coil turns with the capacitor (fixed or variable) until the self impedance is identical to the impedance measured with the isolation coil disconnected. Considerable voltage may be present across the capacitor.

ATU MAINTENANCE

If your sample system is operating properly, any problem indicated on the antenna monitor is being caused by the phasing/coupling system. Common problems with phasors and ATU's are bad components and poor connections.

Poor or nonexistent electrical connections can be present on the tower itself. Many cases are documented where tower sections have become electrically insulated causing a sectionalizing effect, that is, only part of the tower is radiating a direct signal. The best solution for this problem is bonding the tower sections with an arc welder. A competent tower rigger can best advise you on the procedure.

The next problem down the line can be a poor connection from the tower itself to the ATU output terminal. In a recent illustration of this problem, a client in the Caribbean called to discuss his off-the-air problem. Two capacitors and an RFC in the final amplifier of his transmitter blew up. The standby transmitter, a PDM type would not load into the antenna. The ATU was checked and found to be in good order. The feedpipe to the tower, however, measured an open circuit. The feedpipe was not bonded to the tower base, only held in place with a bolt which was found to be tight. Corrosion had set in under the bolt causing an insulation between the copper pipe and galvanized steel. Always be careful when dealing with dissimilar metals, especially in an RF environment.

Tightness of connections should be checked as a matter of routine at least twice a year. This should be done with the AC and RF power **off**. Hardware in the RF ammeter circuit is susceptible to vibration from operation of the shorting bar, if a thermocouple type is used. Calibration of thermocouple RF ammeters should be verified at least once per year. These can be checked against a toroidal Rf ammeter or in a 60 hertz test jig using two ammeters in series.

Never overlook the outer sheath connection from the RF feedline to the ATU. Corrosion can set in causing a high resistance contact and intermittent VSWR. For best protection, use a good bond from the feedline outer sheath to the ATU and from the ATU to the main ground ring at the tower base.

Broken ground straps from the ATU to the tower base ring can cause problems ranging from poor RF matching to severe lightning damage. The ground strap serves to ground static as well as RF current. Make absolutely certain all ground connections are well bonded, not just held in place with screws.

While you are on the way up to the tower, bring along a can of oil for the tower gate lock, fence hinges, ATU lock and door hinges. It's best to do this before winter so that you don't have to try to unjam locks in the frigid cold.

Oil should also be used to lubricate RF contactors. With all AC and RF power **off**, lubricate all joints in each RF contactor. Exercise the relay by hand, giving the oil a chance to work its way through the joints. Be prepared to catch any excess oil with a rag. Lazy RF contactors can prevent a station from signing on the air when the day pattern won't engage on a cold morning. If your tower(s) still use mechanical flashers, this would be a good time to lubricate the joints in the flasher mechanism as well.

Check coil contact clips for tightness. Variable coils should have their rollers checked for signs of heating or arcing. Never use sandpaper to clean a coil. Many Rf coils have a silver coated copper construction. Exposing the copper or underlaying material invites corrosion to set in. If you need to polish up a roller, use Scotchbrite© cloth.

ATU TUNING PROCEDURE

The antenna tuning unit serves two purposes. One is to provide a proper match between the transmission line and the load impedance, the second is to provide the desired degree of phase shift between the line and the load.

In a directional antenna system, the degree of phase shift in an ATU will have a decisive effect on the radiation pattern as well as the pattern and impedance bandwidth of the array. In a nondirectional antenna, the degree of ATU phase shift will affect the impedance bandwidth of the transmission system.

For the sake of clarity, impedance bandwidth refers to the flatness of the load versus frequency at the final amplifier. Pattern bandwidth in a directional antenna refers to the amount of variation in the shape of the radiation pattern versus frequency.

Perhaps the most common mistake made during impedance broadbanding is the lack of consideration given to the transmission line and transmitter output network phase shift. The **only** important location for a flat or symmetrical impedance load is at the final amplifier, **not** at the common point or transmitter RF output terminals. When measuring frequency response, distortion and modulation depth consider that the only valid RF sample for the instrumentation is at the final amplifier or in the far field. Most transmitters provide a voltage sample at the antenna terminals which is inadequate for accurate measurements under mismatched conditions. The transmitter output network is a complex design i.e. not an exact transmission line model, a flat load at the RF output terminals may not yield best results.

Appendix A contains the design formulas for a Tee network. When designing networks for use in the AM service, you must consider the following:

1. The total power in each network equals carrier power plus modulation. The modulated power equals carrier power plus 50%. For example, a 100% modulated 1000 watt carrier contains a total power of 1500 watts.

2. For component current ratings, multiply calculated currents by 1.6 to account for +125% square wave modulation.

3. Remember that mica capacitors are current rated at 1 MHz. For frequencies other than 1 MHz, the capacitor current rating = I_{1MHz} * SQR($_{Fo}$) Where I_{1MHz} is the capacitor current rating at one MHz and Fo is the frequency in MHz.

4. Allow at least a times two safety factor for calculated voltages.

5. If high VSWR is expected, you must allow for additional voltage and current on leg components. High Q circuits can build up considerable voltage potentials.

6. Always allow for some phase shift deviation from design values. Remember, shunt leg current rises with increasing absolute values of phase shift.

7. When measuring impedances using an RF bridge, do not forget to correct the reactance for each frequency. For an OIB, multiply the measured reactance by the frequency in MHz, for a GR bridge divide the measured reactance by the frequency in MHz.

8. Experts have debated the use of open versus shorted unused turns on coils for some time. In the shorted unused turns case, shorting only one or two turns of a coil can cause overheating of the connections and strap. This is due to the inductance of the strap. Take care to avoid this situation, if possible.

9. When adjusting tapped coils, it is important to realize that there is **not** a linear relationship between coil turns versus inductance. As a matter of fact, the relationship will be quite different between the shorted and open unused turns case especially at the coil ends.

AN ACTUAL ATU DESIGN

Let's examine a -95° Tee network connected to a 66° length of transmission line. The frequency is 1310 KHz, the power is 1000 watts. We choose a shunt capacitor of .001uf and an output capacitor of .0015uf. The tower impedance measures 58+j77.

From Appendix A we find the input leg requires a 7.1 uh coil, the output leg requires a 7.67 uh coil and the shunt leg coil will be 8.19 uh. For an unmodulated 1000 watt carrier, the shunt branch current will be 6.36 amps. For +125% square wave modulation, the shunt current rises to 10.2 amps. If we adjust the network to a phase shift of -125°, we find that the +125% square wave modulated shunt current equals 12.2 amps. This is nearly double the value calculated for the original, unmodulated design value.

It is customary to use a capacitor in series with the tower to provide DC blocking from the tower to the ATU. A static drain coil should shunt the tower before this capacitor. A three layer lighting choke provides an excellent DC ground. For high impedance towers, an Austin transformer is used for passing lighting AC voltage across the tower base.

ATU PHASING

The choice of ATU phase shift is critical for optimum bandwidth. In the nondirectional case, it is sufficient to adjust for a flat load or "horseshoe left" at most transmitter output terminals. Horseshoe left refers to a Smith chart representation of the load impedance versus frequency. Most plate modulated transmitters and a number of popular solid state transmitters have a 225° output network. Presenting a load with higher resistance at the upper sideband, lower resistance at the lower sideband and negative reactances at both sidebands will produce the correct symmetry at the PA final stage for these transmitters. Contact the transmitter manufacturer if you are uncertain about the output network design. A symmetrical load presented to the antenna terminals of a transmitter with a 225° output network will produce a very unfavorable load at the PA stage.

There is an easy test that can be performed for a nondirectional station to determine if your ATU phasing is optimum. Modulate the transmitter 50% with a 10 KHz sinewave. Measure the field intensity using an FIM at the carrier and ±10 KHz. The sidebands should appear symmetrically about the carrier at an amplitude equalling 25% of the carrier field intensity.

Substantial deviation from the desired sideband amplitudes and symmetry is an indication that work on the ATU is necessary. The exact percentage of modulation may be difficult to sample since most transmitters provide only a voltage sample at the RF output.

For the directional antenna, a much more complex situation is at hand. The factor of pattern bandwidth enters the picture. A simple test for pattern bandwidth would be to run the transmitter at low power, just sufficient to obtain a 100% loop current reading on the reference tower with the calibration pot set to maximum sensitivity. Substitute an RF oscillator in place of the transmitter crystal and measure the antenna parameters at carrier and ±10 KHz. Recalibrate the reference loop current at each frequency. If the antenna monitor parameters vary greatly with frequency, the phasing and coupling system need to be redesigned.

Each tower in a directional antenna system has a branch phasing value comprised of the ATU, transmission line, phase shifter (if used) and power divider phase shift. There are an infinite number of branch phasing combinations that will produce the correct radiation pattern. A wise choice of system phasing, however, will produce the best pattern and impedance bandwidth.

The methodology to determine optimum branch phasing in a DA system is beyond the scope of this paper. Computer modelling is used to simulate phasor and coupling system design. The technique of "moding" can be employed in any symmetrical array (equal height only arrays for night operation) to determine which of 2^{n-1} parameter combinations produces the most favorable results. n equals the number of towers in the array. Moment method computer analysis provides accurate impedance and drive point parameter data, this coupled with measured self and mutual impedances can afford excellent bandwidth characteristics often without buying a single RF component. The choice of power divider and coupling network design is crucial to obtaining the best performance from your directional antenna system.

You may encounter variations of network designs in an AM antenna system. For nondirectional ATU's, the technique of employing a series compensation input arm (X1) comprised of a coil and capacitor can be used.

If the upper sideband reactance is capacitive (-j) we can take advantage of the fact that a series resonant circuit appears inductive above resonance. A unique combination of coil and capacitor can be used as the input leg of a Tee network to cancel the sideband reactances so that the carrier and sidebands contain pure resistive values.

On some occasions, you will find a coil-capacitor combination used on the input arm because the design value calls for 1 or 2 uh of inductance. The designer may choose a capacitor of such size to allow the coil value to be 3 to 6 uh for greater adjustment ease without compromising bandwidth.

A simple trap technique to keep a neighboring station from entering your feeder system is to employ a series resonant L-C combination for each ATU shunt leg. Appendix A contains the equations necessary to design such a circuit. We are choosing a unique combination of L and C to produce the required shunt reactance at the operating frequency and be series resonant at the neighboring frequency. The neighboring frequency must be greater than Fo for a lagging Tee network and lower than Fo for a leading Tee network.

CONCLUSION

While it is not possible to address every combination of network or feeder scheme in this paper, it is sufficient to advise the reader to use whatever combination provides the best results. The intense mathematics associated with system analysis and optimization are relegated to computers.

Prior to any network adjustments, all mechanical connections should be thoroughly checked, cleaned and tightened. Always keep an accurate record of starting taps on coils, dial counter readings and measured impedances, currents, etc.

In many instances, some simple maintenance can solve bothersome problems and improve the efficiency and stability of the transmission plant.

APPENDIX A

Equations for Tee network design:

$$X1 = (Ri \div \tan\psi) - X3$$

$$X2 = (Ra \div \tan\psi) - X3$$

$$X3 = \frac{\sqrt{(Ri \times Ra)}}{\sin\psi}$$

$$I1 = \sqrt{(P \div Ri)}$$

$$I2 = \sqrt{(P \div Ra)}$$

$$I3 = \sqrt{(I1^2 + I2^2 - 2 \times I1 \times I2 \times \cos\psi)}$$

$$E1 = I1 \times X1$$

$$E2 = I2 \times X2$$

$$E3 = I3 \times X3$$

Where:

ψ = Phase Shift of Tee Network
Ri = Input Resistance (50Ω)
Ra = Antenna resistance
X1 = Input Leg Reactance
X2 = Output Leg Reactance
X3 = Shunt Leg Reactance
I1 = Input Leg Current
I2 = Output Leg Current
I3 = Shunt Leg Current
E1 = Voltage Across X1
E2 = Voltage Across X2
E3 = Voltage Across X3

Remember to include the antenna reactance as part of X2.

Equation for Input Arm (X1) Series Compensation Network. This circuit will cancel equal sideband reactances provided upper sideband reactance is capacitive (-j):

$$L = \frac{Fu \times Xu}{2 \times \pi \times Fu - Fo}$$

$$C = \frac{1}{4 \times \pi^2 \times Fo^2 \times L}$$

Where:

L = X1 coil in microhenries
C = X1 capacitor in microfarads
Fu = Upper Sideband frequency MHz
Fo = Carrier frequency MHz
Xu = Absolute value of upper sideband reactance
π = 3.1415926589793

Series resonant trap for use in shunt leg (X3) to provide low impedance to reject frequency while maintaining correct shunt leg reactance.

$$C = \frac{Fo}{2 \times \pi \times X3 \times Ft^2} - \frac{1}{2 \times \pi \times X3 \times Fo}$$

$$L = \frac{1}{4 \times \pi^2 \times Ft^2 \times C}$$

Where:

C = Shunt Capacitor in uf
L = Shunt Coil in microhenries
Ft = Trap frequency in MHz
Fo = Carrier frequency in MHz
X3 = Absolute value of X3

By rearrangement, we can calculate the zero (reject) frequency of any series circuit:

$$Ft = \frac{1}{2 \times \pi \times \sqrt{(L \times C)}}$$

MODERN PRACTICAL TECHNIQUES FOR INSTALLATION AND REPAIR OF AM ANTENNA GROUND SYSTEMS

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ABSTRACT

While the basic requirements for AM antenna ground systems have changed little over the life of the broadcast service, certain modern tools, materials and techniques are now available to ease the process of installing or repairing such a system. This paper undertakes to provide practical guidance for the broadcaster planning to construct a new or renovate an existing ground system. While much work has been done over recent years in developing innovative above ground systems, the traditional buried copper approach is by far the most commonly employed, and is the subject of this paper.

INTRODUCTION

There can be no doubt that a sub-surface copper ground system is an integral part of a well-functioning AM antenna or directional array. A damaged or poorly installed system can present an unstable load for the transmitter and may contribute to several other well-known operational challenges such as poor radiated efficiency, drifting antenna monitor indications, and difficulty in maintaining licensed monitor point field strengths.

Once completed and buried, there is seldom a practical opportunity to easily inspect or repair a station's ground system. Since digging up and replacing a poorly performing ground system is likely to be time-consuming and expensive, there is sufficient impetus for the engineering manager planning a new or substantially renovated ground system to take considerable pains to make sure the job is done right. Fortunately, the skills required for the task are easily mastered and the materials necessary are generally available; these together with careful advance planning are all that is required for a successful project.

A SERVICEABLE GROUND SYSTEM

First, a description of a typical well-executed ground system is in order. A standard practice for AM antenna ground systems (absent any unusual terrain or lot-size considerations) is to install a mat of expanded copper mesh screen beneath the tower. See Figure A. This mat is typically deployed as a square (nominally 48 feet per side, but sometimes 24 feet per side) with the tower at the square's center. Beyond the close-in copper mat area, some 120 lengths of 10 AWG copper wire radiate outward from the perimeter of the square 90 to 140 electrical degrees in length, radially spaced at 3° intervals. At areas in the field where large obstacles such as buildings exist, a 4-inch copper strap is buried surrounding the obstacle, the radial wire is attached to the strap and picked up again on the other side of the obstacle, again attached to the strap.

At the perimeter of the close-in area, pressure-treated lumber is set into the ground to serve as a frame for the square copper mat. A continuous length of 4-inch wide copper strap is nailed to the top of the lumber frame; the copper mesh, all radial wires, copper straps leading to the tower base insulator, and any other straps that leave the close-in area all terminate on the strap square.

In directional antenna arrays, each tower has its own ground system as described above, but in places where the radial wires from adjacent towers would otherwise intersect, those wires are shortened and bonded to a 4-inch copper strap running midway between and on a line perpendicular to the line between the towers. See Figure B.

PLANNING

Scope of the work. Depending on individual circumstances, a project may entail building a ground system from scratch or replacing an entire existing ground system. Also, in cases where only the fragile inner copper mesh has deteriorated to a point requiring replacement, it is practical to replace only the inner system components and re-use the existing radial wires. In evaluating the scope of a particular project, it should be recalled that ground system specifications are detailed in each station's license or construction permit. Assure whatever system installation or repair being planned ultimately meets those specifications.

Other considerations in the initial planning stage should include price and availability of materials, integration of outside landscaping and fencing contractors into the project's time line, determination of utility (telephone company, electric, natural gas, etc.) easements and underground pipelines in the intended dig areas, worker safety in the vicinity of broadcast towers, and any off-air or reduced facility operation that may be required.

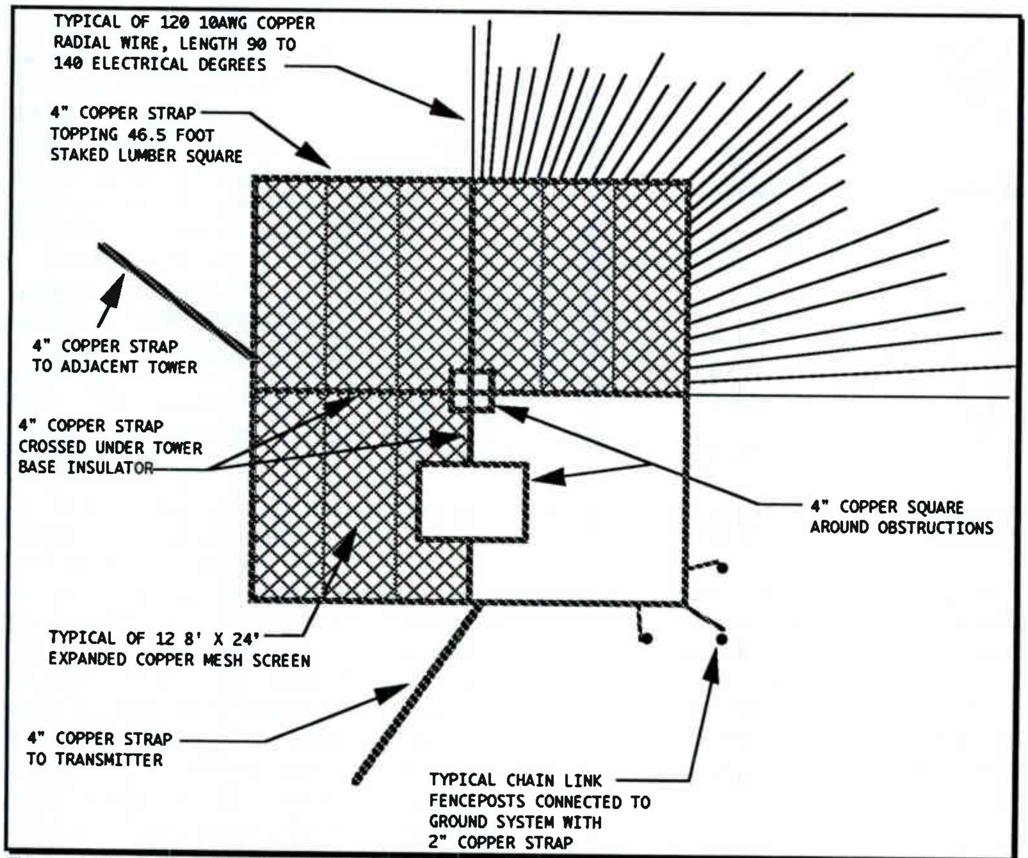

Figure A. **Close-in view of tower ground system showing major elements**

Materials and Quantities. While installations vary, it is possible to make fairly accurate estimates of the amounts required for many materials used in the project. An accurate plot plan of the site is of invaluable assistance in calculating tower-to-tower and tower-to-building distances for copper strap runs and for determining lengths of radial wires. Required amounts of expendable materials such as solder, flux, and welding gases are more difficult to judge, because the amount used can vary greatly with the skill of the operator. While the amounts suggested in this report could be considered general purchase guidelines, it may be possible to arrange with suppliers to buy extra quantities of materials with the understanding that unopened packages may be returned for credit at the conclusion of the project. If the time frame of the project is at all critical, it is wise to either have extra materials on hand or be assured of the ability to acquire them quickly.

Material List.

- Copper strap. 4-inch copper strap comes in .034" thickness suitable for ground system work. An amount of strap sufficient to cover the interior lumber frame, to form the crossed straps under the base insulator that bisect the lumber square, to outline the perimeter of any obstructions in the ground screen field (such as tower base pier, ATU building, etc.), to connect in a line between towers, to connect tower(s) to the transmitter building, and to run between adjacent towers in a directional array should be determined. 2-inch copper strap is suitable for connecting between the 48' frame and any metal fence poles (all such poles should be grounded; they may be either soldered or clamped).

- Copper screen. Expanded mesh copper screen is available in rolls that are 8 feet wide by 24 feet long. Three such screens are sufficient for a 24' x 24' square; twelve are necessary for a 48' x 48' square.

- Copper wire. Number 10 AWG copper wire is typically used for radial wires. Determine how long your radials are to be (specified in the station license), then multiply by 120 to determine length for a single tower. You may allow less for copper wire if the radials from adjacent towers are to be shortened and soldered to an intersecting strap. Assuming a pipe stand of sufficient size is available, there will be less waste of copper wire and less time spent changing out empty spools if the copper is ordered in large quantity reels.

- 45% Silver solder. Typical solder for this application is 45% silver, 15% copper, 16% zinc, 24% cadmium. This solder is used to join adjacent pieces of copper strap and to attach radial wires to copper strap, and has a melting range of 1125° – 1145° F. Buy it in wire coil form; it is available as either 1/16" or 3/64" diameter wire, and sold by the troy ounce. Depending on the skill of the welder, 25-30 troy ounces per tower interior ground system will be required. Add to the amount if radials from adjacent towers must be soldered to an intersecting strap.

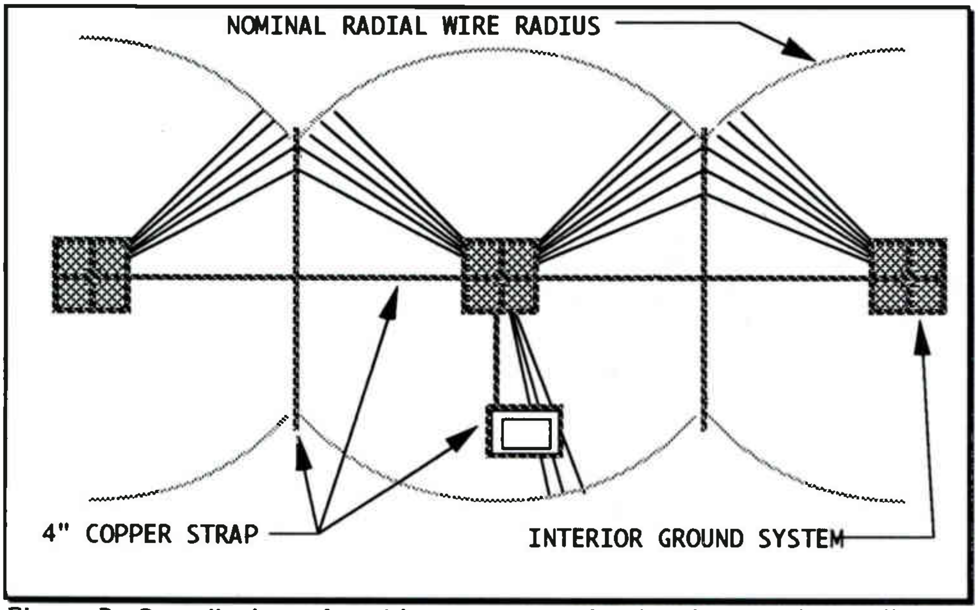

Figure B. **Overall view of multi-tower array showing intersecting radials**

- White silver brazing paste flux. This material is used in silver-soldering to carry away oxides formed in the brazing process, strengthening the finished joint. It can be procured in half-pound and one-pound jars; and is usually heavily diluted with water before application. Three pounds of undiluted paste flux will handle the interior ground system; more may be required if radials from adjacent towers must be soldered to an intersecting strap.

- Silver-Phosphorous-Copper brazing alloy rods. Typical solder for this application is 15% silver, 5% phosphorous, 80% copper. The phosphorous in the alloy acts as a fluxing agent on copper, so separate flux is not required. This material is used to weld the seams in adjacent rolls of expanded copper mesh, and to attach the mesh to copper strap. It should be ordered in 1/8" square by 20" long rods, which are commonly sold in one-pound packages. Twelve to fifteen pounds will be required for each tower interior ground system.

- Galvanized-steel barbed roofing nails, 1-1/4" long, 11 gauge. Nails are used to attach the copper strap to the 48 foot lumber square. These nails come approximately 219 per pound, two pounds will be sufficient for up to three towers' interior ground systems.

- Wire scratch brushes. 8-1/2" long stainless steel. These brushes are useful for painting flux onto the copper to be soldered. Each welding team should have two per tower; additional brushes may be useful in the case of directional antenna radials welded to intersecting straps.

- Welding gases. Oxygen cylinders of 20 cubic-foot capacity are easily held in a portable welding carrier. Approximately ten to twelve such cylinders will be necessary for each tower's interior ground system. The fuel gas of choice is MAPP® gas, procured in 16-ounce cylinders. MAPP® gas in unopened cartons is UPS-shippable, although a hazardous materials surcharge applies. Expect to use ten to twelve cylinders per tower interior ground system.

- Lumber and Stakes: 4 x 4-inch pressure treated lumber is used to form the square frame around the tower. For a nominal 48 foot square, estimate 200 feet of lumber. This lumber should be staked to the ground at five to six foot intervals using two-foot long rebar stakes. This will help prevent the lumber frame from shifting in the future, when such movement could damage the delicate copper mesh screen. If the tower field is subject to frequent flooding by moving water, consider installing an additional outer 50 foot square of 2 x 12-inch pressure treated lumber as a crib to help prevent the gravel from being carried away.

- Landscape Fabric. Sufficient anti-weed matting to cover the approximately 2500 square feet within and immediately outside each tower's interior ground system should be planned. Landscape fabric is a porous, black slightly spongy material commonly supplied on rolls six feet wide; it is stapled to the ground in overlapping sheets. A 10 percent overlap suggests 2750 square feet of material. Landscape fabric is also known as geotextile fabric.

- Gravel. "2-inch stone" is suitable for interior ground system gravel. The rule-of-thumb for gravel volume is: for each 1000 square feet to be covered to a depth of two inches, seven tons of stone are required. Calculating this out for a typical tower interior, where we fill a 46-1/2 foot square to four inches, then top a 50 foot square with two more inches, about 48 tons of gravel are required. Since one ton of gravel is approximately equal to one cubic yard, and a standard dump truck holds about fifteen cubic yards, a little over three truckloads of gravel are needed for each interior system.

Tools and Equipment. Two or three portable oxygen/MAPP® gas welding kits are recommended. Each kit should also be equipped with eye protection, an igniter, welder's gloves, two large and two small welder-style gripping pliers, a hammer, a pair of heavy pliers, aviation snips, medium grit sandpaper, and several pieces of non-woven abrasive. More than one team of welders can share a gas cylinder wrench. Each worker who will be touching the copper should have a pair of flexible plastic-coated stainless-core anti-cut gloves to protect his or her hands. Finally, since nearly all the work is done at ground level, several sets of knee pads will make the job noticeably less painful.

SEQUENCE OF CONSTRUCTION

1. First and foremost, and for every day the project is ongoing, assure that all work to be done takes place in a safe environment. Make sure that no one connected with the project can possibly come in contact with a live tower, and that no person is exposed to RFR in excess of allowed levels.

2. Existing fences within the inner ground screen area will probably need to be removed for the project. If the fences are removed, assure that adequate temporary fences protect the general public from contact with the tower or from RFR overexposure during periods (for example, overnight) when the tower is operating but no station personnel are present.

3. If new radial wires are to be installed, easily visible stakes should be installed at each of the 120 compass points distant from the tower where each radial will stop. A surveyor and assistant can be hired for one day to mark the points, or a transit can be rented and the job done by station personnel. The stakes will serve as visual targets to allow a tractor operator to plow in the wires in straight, equally spaced radial lines. Lines midway between adjacent towers in a directional array (where radial wires from each tower will be shortened and attached to a 4-inch copper strap) should also be determined and staked at this time, as should any strap paths from towers to the transmitter building.

4. If required, trenches for the between-towers and tower-transmitter straps should be dug at this time. The trenches should be six to eight inches deep and five inches wide to accommodate the 4-inch straps. Note the actual copper strap will be installed later, after the radial wires have been pulled in.

5. If new radial wires are to be installed, they should be run at this time. Set up a spool of wire of sufficient length on a pipe-stand near the base of the tower such that the spool is free to rotate in place. Tie the wire from the spool through a small hole drilled in the tractor plow, and have the tractor drive toward the stake with the plow set at a depth of six to eight inches. An assistant should observe the pay-out of the wire spool to assure kinking does not occur, and to provide some back tension on the reel in case the tractor stops unexpectedly. When the tractor reaches the stake, the operator should untie the wire from the plow and remove the stake to make the next stake in turn visually apparent. At the tower side, the wire should be cut from the spool with sufficient length to allow it to be attached to the inner ground system frame. The length of the radial should be walked with a shovel to assure the wire is indeed inside the trench, and to cover the wire with the plowed dirt.

6. An alternative arrangement has also been used to lay copper radial wires. In this arrangement, the reel of copper wire is mounted on a pipe attached to the tractor. The wire feeds through a plastic sleeve attached to the plow that extends in a gentle arc down into the furrow that is made by the plow. The wire is tied off near the base of the tower, and the tractor drives toward the marker stake as before. The plow cuts a diagonal slice in the ground of the required depth, and the copper wire is unreeled into the slice. Upon reaching the stake, the tractor operator cuts the wire leaving the reel, allows the last few feet of the cut wire to be pulled through the plastic sleeve and buried, removes the marker stake, then drives backward along the trench using the large rear wheel to tamp down the ground in the furrow along its entire length. The operator then repeats the process for the next radial. This variation of the process has the advantage of requiring only one person (the tractor operator) to install the radials and speeding somewhat the time required for the job, but does require a specially modified plow. If this aspect of the job is being sub-contracted, talk to several landscapers (or even tower rigging firms that may have done this type of work before) to determine if any can fashion a wire-burying arrangement of this type.

7. When all intersecting radials have been laid, the between-towers and tower-transmitter 4-inch copper straps can be run into their trenches. Those radial wires to be terminated on a strap should be run under the strap, then be bent over the strap, cut and crimped with a pliers. Wires that cross a strap should be bent and crimped to form a hairpin loop that can be easily soldered to the strap. In both cases, leave a little slack in the radial wires so they will not be later be stressed and broken by earth movements associated with freezing or presence of heavy equipment being driven. See Figure C.

8. While skilled welding operators are working on attaching radials to outlying straps, landscapers can be preparing the inner ground screen area. This area should first be hand-dug to expose the old ground system.

Existing (as well as any new) radial wires that are to be retained in the new system should be identified, cut well inside the ground screen area, have their ends brought back outside the area, and marked with taped flags so they can be found and attached later to the copper-topped frame. Other remnants of the old interior ground system should be removed.

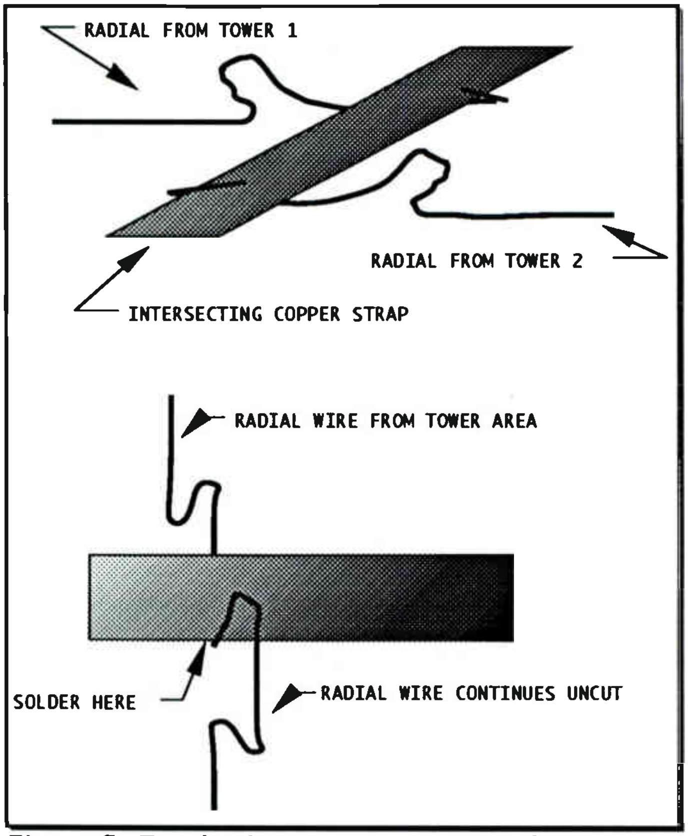

Figure C: **Terminating (above) and crossing radial wires on a strap. Note strain relief loops.**

9. The inner ground screen area should be excavated to a uniform depth of four inches below surrounding grade. At that point, landscape fabric should be installed to cover the entire area and prevent future weed growth.

10. Once the fabric has been laid, the ends of the installed radial wires identified in step 8 above can be draped back into the inner ground screen area. Over the wires, begin installing lengths of 4 x 4-inch pressure-treated lumber to form the frame for the inner ground screen area, with the tower as the center of the square. See Figure D. To allow for the overlap of adjacent rolls of screen, and to leave sufficient surface on the top of the frame to solder the mesh to, the outside dimensions of the lumber square should be 46' 6" on a side. The lumber should be staked to the ground with two-foot sections of rebar material, with the rebar driven flush with the top of the lumber.

11. Fold the radial wires back over the lumber square, so that they drape outside and the area within the square is clear. Landscapers should now install gravel to a uniform depth of two inches inside the lumber square.

12. Once more, fold the radial wires back in to the square atop the gravel. 4-inch copper strap can then be unrolled onto the top of the lumber frame, and tacked at eighteen-inch intervals with galvanized roofing nails. Start unrolling at a point 1/3 of the way from one corner, and unroll in the direction of the far corner. One person can unroll the copper strap and keep it aligned on the lumber top while another follows behind with nails. Do not cut the strap at the corners, but rather double-fold it. For example, to make a right turn, first bend the copper strap on a 45° angle towards the left, then fold the copper back on itself in the new direction. The copper will then continue to unroll from the bottom of the roll, where it is much easier to manage.

13. When the copper-topped square is completed, one team can continue laying out the other copper straps required in the inner screen area (those around obstructions like the tower pier or the ATU, and those that connect the crossed straps under the tower), while another team can solder together the splice of the copper strap atop the lumber and begin attaching radial wires to that strap. Radial wires run from outside the frame under the lumber, are inserted between the strap and the lumber, folded over the top of the strap, cut, crimped and soldered. Note: if old radials are being re-attached, make sure to clean the copper with sandpaper or non-woven abrasive down to bright metal for a solid connection to the strap. Attach all radials and make a note of the number of radials attached to each side of the square for future reference.

14. Strap-to-strap splices come in three main varieties: T, + and I, named for the alignment of the straps in the joint. For a T splice, run the piece of strap that terminates on a perpendicular run under the other strap, fold it over the top, and cut it so that one or two inches protrude over the crossing strap. Hammer the joint flat, and make two solder beads: one at the end of the terminating strap, the other at the point where the terminating strap goes under the running strap. For a + splice, run one strap under another and make two solder beads, one each on the top and bottom of the intersecting strap. Lastly, for a I splice where one strap is spliced on to extend the length of another, make mating folds leaving one inch of folded material in each strap, align the folds so they join, hammer the joint flat, and make two solder beads, one each at the intersections' visible joints on the top and on the bottom of the strap. See Figure E. In all cases, use sets of locking welder's pliers to keep the straps flat and tightly aligned while you are applying solder to the joint.

15. Existing copper straps that are to be tied into the new ground system should be cleaned down to bright metal. This is very easily done with a high-speed (12,000 rpm) portable electric sander/grinder outfitted with a wire cup brush. This moderately priced power tool is well worth acquiring for this application. (When using high-speed tools, be sure to use brushes and other accessories specifically marked as intended for high-RPM applications.) 2-inch copper straps leading to any metal fence posts in the immediate ground system area can also be installed to the copper-topped square at this time. At the fence post end, the strap can either be clamped or soldered to the post. Scrape off any galvanizing before soldering.

16. When all the straps are cut and soldered, unpack and lay out the first roll of copper screen. Use anti-cut gloves at all times when handling the copper screen; the screen material is quite capable of causing deep and painful cuts to unprotected flesh.

17. Begin installing the screen at one corner of the frame. The mesh is braze-welded at 6-inch intervals to the copper-topped frame. A team of two persons works best for soldering the mesh to the strap. A helper uses a convenient length non-flammable pushing stick to push the mesh onto the strap so the welder can see where a mesh diamond intersection would touch the strap. The mesh is then lifted out of the way, and the strap at that point is heated. Silver-phosphorous brazing alloy is applied to the heated strap at that point until it flows into a small puddle. When the welder is satisfied with the size and character of the solder puddle, he signals for the helper to push the mesh down into the puddle. Heat is applied for only a very short time to cause the puddle to flow over the mesh, then removed. Pressure is maintained on the mesh by the pushing stick until the solder pool cools and solidifies. Both team members then move six inches down the line and repeat the process. See Figure F.

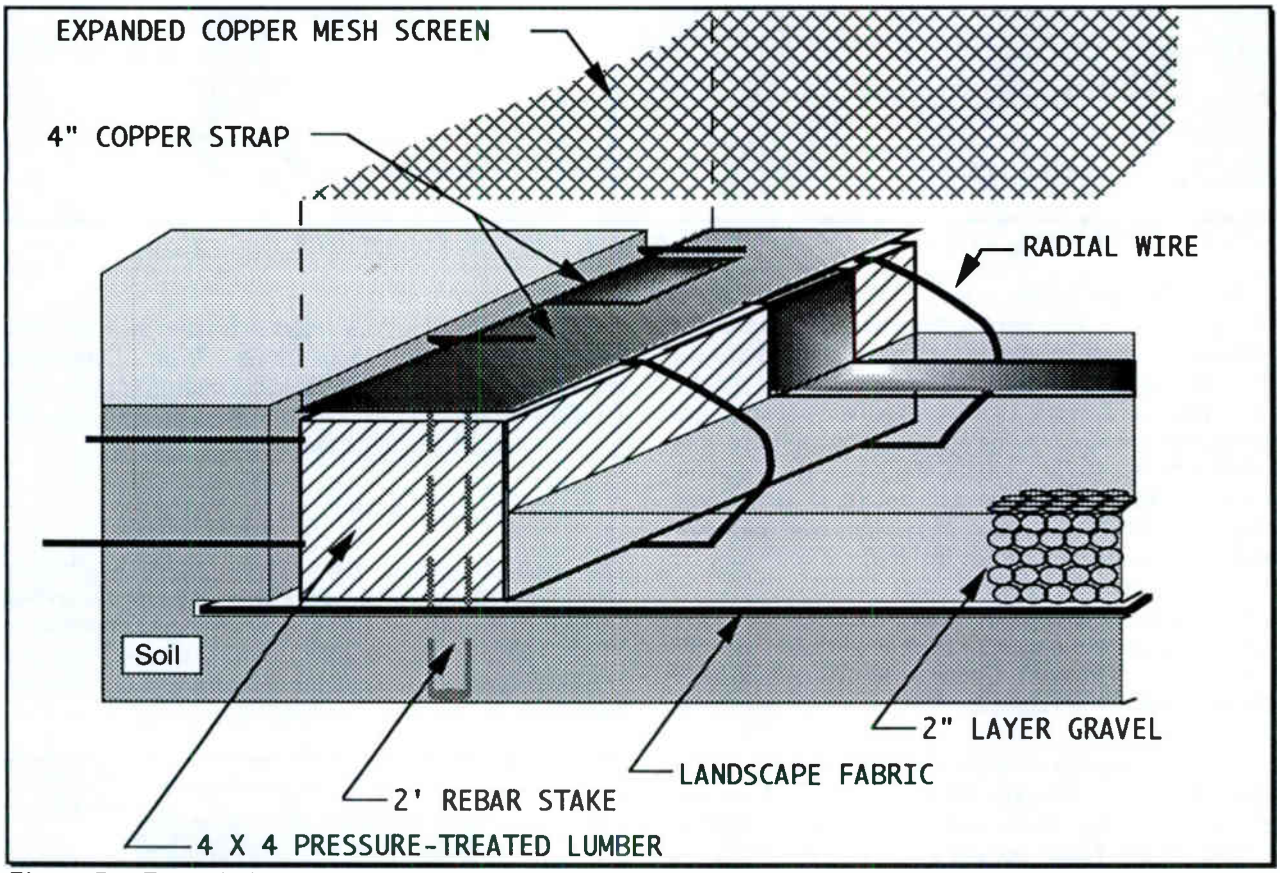

Figure D. **Exploded cross-sectional drawing of a section of interior ground system perimeter frame. Note that approximately four more inches of gravel (not shown) will cover the finished copper screen.**

18. After the first roll of screen is laid in place and temporarily weighted at its corners to keep from rolling back up, and while the first team carries out step 17 above, a second team can unpack the second screen. This screen should be positioned to abut the first screen along its long dimension of 24 feet, with about 1/2 of a diamond mesh overlap. Again weighing down the corners of the unrolled screen, the two team members each begin splicing the adjacent screens together by twisting the

diamond points together. Two adjacent points are revolved 360° twice around a common imaginary axis. The result for each diamond splice is two strong points of contact. Each and every adjacent diamond should be twisted; do not skip any splices. See Figure G. Two persons can twist at the same time most effectively if one starts at one end, the other starts in the middle, and both move in the same direction.

24. Landscapers should return to fill up the ground screen framed area with two more inches of gravel, and should then add an additional two inches of gravel, slightly crowned, over the entire surface including the copper-topped lumber. Landscapers must carefully use hand tools only (shovels, inverted rakes, small wheelbarrows, etc.) to spread the new layers of gravel; do not allow tractors or heavy carts to be driven on the fragile copper mesh.

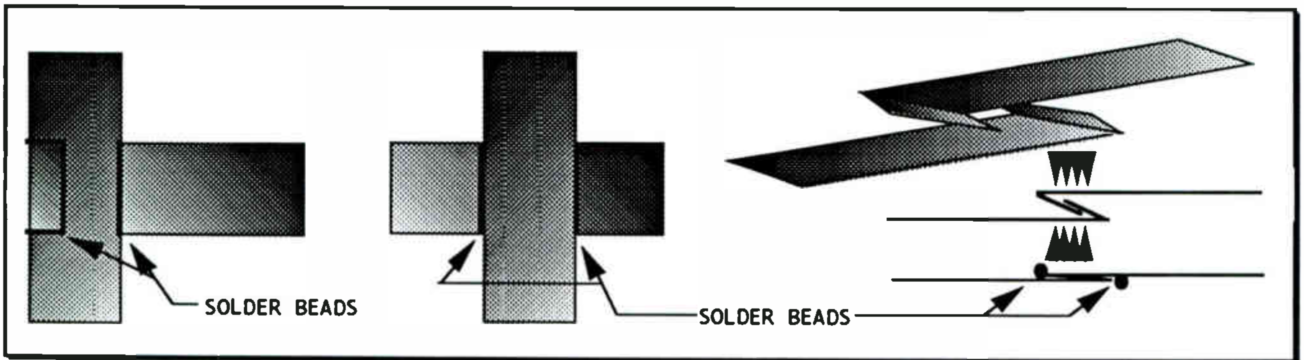

Figure E. **Terminating, crossing, and overlapping strap-to-strap connections**

19. Soldering the twists is next. Making each solder joint requires only one person, but two can work at the same time, as above, with one starting at one end and the other starting at the midpoint of the 24 foot length. Solder each and every twist with silver-phosphorous brazing alloy, heating the delicate copper carefully until the solder begins to flow around the joint. Do not allow the copper mesh to become red hot (it is very easily damaged at this temperature, and solder does not flow readily onto the material if it is this hot).

20. Usually one team can work on soldering the mesh to the copper strap while another team unpacks and unrolls the next piece of screen. Continue in this fashion until the first six screens are completed.

21. If obstructions exist within the framed area, cut the copper mesh with aviation snips to fit around the obstruction, and make sure to solder the newly cut edges of the mesh to the 4-inch copper strap that surrounds the obstruction. Likewise, at all places the mesh is unrolled over copper straps running in the framed area, the mesh must be soldered to the strap at 6-inch intervals.

22. The remaining six mesh screens are installed the same way as the first, but must additionally be twisted and soldered along their 8-foot dimension with the adjoining already installed screens.

23. When all parts of the ground system are complete, carefully inspect the work to make sure no solder points have been forgotten. Take photographs of important features of the ground system for future reference. Permanent fencing around the ground system may be installed at this point.

25. Check operating parameters to be sure the transmitter system is well-matched to its new load. Directional antenna stations may well find that phasor adjustment is required to re-establish licensed parameters if the old ground system was badly deteriorated.

SOLDERING PROCEDURES

Definition of terms. A lay person might often use the terms welding, brazing or silver-soldering interchangeably to refer to the process of joining the copper components of an AM antenna ground system. To a welding professional, however, these terms describe distinctly different processes.

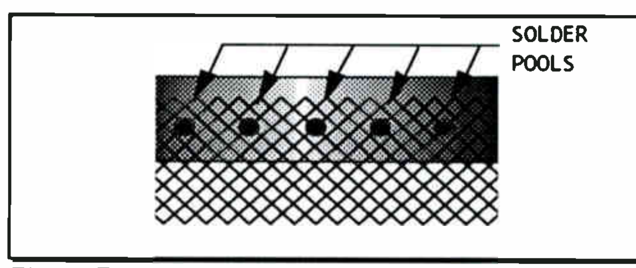

Figure F. **Mesh to strap connections at mat perimeter**

Welding refers to the joining process in which the base metals to be joined are heated to the melting point and fused together. Filler material (more metal) is often melted and added to the resulting joint to improve the strength of the weld. Welding per se is not used in ground-system work

Brazing refers to a general process where the pieces to be joined are fitted with extremely close tolerances (.002 to .006 inches) and heated to a point where a filler material with a melting temperature below that of the materials to be joined begins to flow and is drawn into the joint by capillary action. *Silver-soldering* is a specific form of brazing that uses a silver-alloy material as the filler. Welding and brazing also differ from one another in their manners of joint selection and preparation. Silver-soldering is the process used in ground system work to join copper strap.

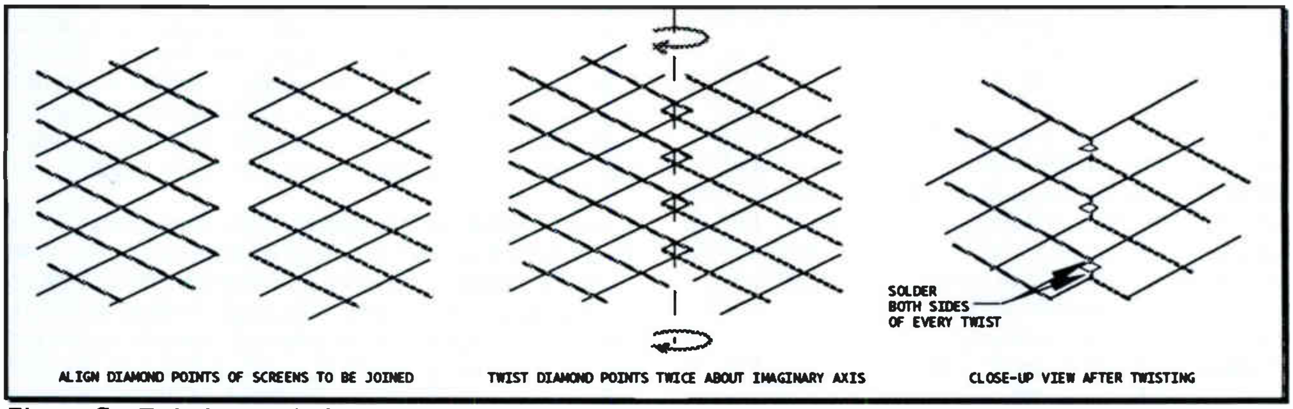

Figure G. **Twisting technique to join screen material**

Braze-welding is the term for a fourth process that more accurately describes the methods used in making joints involving expanded copper mesh. As in brazing, an alloy that melts at a lower temperature than the copper material to be joined is employed, but unlike brazing, the process does not depend on capillary action to pull the alloy into the joint. Rather, the alloy filler material is fused to the surrounding copper at the point where it melts. Braze-welding does not require the extremely tight joint tolerances of brazing.

Materials needed. All of the silver-soldering and braze-welding needed for an AM antenna ground system can be done with an oxygen-fuel gas brazing set. While acetylene can be used as the fuel gas, a much safer-to-handle fuel that works very well in this application is stabilized methylacetylene and propadiene, known commonly by its trademark, MAPP® gas. A number of manufacturers sell small "totable" welding kits based on oxygen-MAPP® gas. These kits include the necessary hose set, regulator, welding handle and tips for this type of work, all housed in a convenient plastic carrier with a handle, making it easy to carry from place to place. Once you have the welding set, you will need to rent or purchase refillable oxygen cylinders and disposable fuel gas cylinders.

Torch Assembly and Check-out. Follow the welding set manufacturer's directions carefully to assemble its component parts, to attach gas cylinders, to light the torch, and to shut down the torch. Inspect all mating points for cleanliness and good repair. Use only accessories, attachment tools, and procedures recommended by the manufacturer. Use only hoses, regulators, torches, valves and tips specifically designed for your application. Never use accessories intended for use with different fuel gases. Do not forget that welding gases are highly combustible and inherently dangerous. Before opening a cylinder's main valve, set the regulator for that cylinder all the way (usually counter-clockwise) for minimum pressure to avoid damaging the regulator diaphragm. When opening a cylinder's main valve, always position yourself with the main valve between you and the regulator; do not stand in front of or behind the regulator when opening the main valve. Do not wear loose or flammable clothing. Wear welding gloves and eye protection whenever using the torch. Never use matches to light a welding torch; gases collecting near the tip can ignite rapidly and cause burns. Use only a safety igniter. A number-2 brazing tip is suggested for ground system welding; consult the torch manufacturer's instructions for correct oxygen and MAPP® gas regulator working pressures for use with this tip.

Brazing copper strap. Good brazing depends on a thoroughly clean and well-fitted joint. The copper strap to be joined should be polished down to bright metal with sandpaper or a non-woven abrasive, then rinsed clean of abrasive material. The overlapping joint should be fitted as closely as possible. White silver-solder flux, which comes in half-pound or one-pound jars of thick paste, should be diluted with water until it is the consistency of milk, then painted with a wire scratch brush onto the metal surfaces to be joined. The joint should then be assembled and clamped firmly in place. Heat is applied evenly over the joint area until the work becomes hot enough to melt the 45% silver solder. If the joint is clean, fluxed, and closely fitted, the molten solder will be drawn into the joint by capillary action. Always heat the work, never the solder. Apply the heat uniformly to preheat the joint and the surrounding metal, and avoid holding the torch too long in one area, a practice that causes "hot spots" to develop in the copper that deform its shape and ruin the tight fit of the joint. When the metal becomes hot enough to melt solder, begin at one side edge of the joint and apply solder. Excess molten solder will flow towards the heat, so let the torch precede the solder along the seam, always moving the torch in an oscillatory manner to prevent overheating. If the solder appears to be "balling up" instead of flowing into the joint, resist the urge to apply more heat. Instead, allow the joint to cool, apply more flux to the joint, then re-heat and try again.

Braze-welding with silver-phosphorous rods. The twisted mesh-to-mesh joints are heated carefully with the torch (because of their low mass they reach solder-melting temperatures quite quickly), then the heat is removed and the silver-phosphorous rod is touched quickly to the joint. The solder melts and flows easily between the wires of the joint, then solidifies, fusing to the wires to make a strong bond. *Tip*: As the material in the brazing rod is used up over several joints, the rod will eventually reach a point where it is too short to hold safely. At that point, use a small amount of heat from the torch to fuse the scrap onto the end of a new rod. This avoids waste and can prevent accidental burns to a welder who might otherwise be tempted to make "just one more" weld with a too-short rod.

Type of flame. Four distinct types of flame can be produced by an oxygen/fuel gas welding set, simply by varying the amounts of the two fuels delivered to the welding tip with the valves on the welding handle. The torch is initially lit with MAPP® gas only being supplied; its flame is distinguished by a yellow-orange appearance and significant smoke. As the oxygen valve is slowly opened, the flame changes in character so that a small intensely white cone with a feathery edge appears at the tip, with a larger white flame visible somewhat farther from the tip, and a light orange flame noticeable at the periphery of the flame. This is known as a *reducing* or *carbonizing* flame. As more oxygen is added, the sound of the flame changes. A small, well-formed white cone is visible at the tip of the torch, and the outer envelope of the visible flame is nearly colorless and flicked with bluish-orange at its outer edges. This is known as a *neutral* flame, and is the one best suited for brazing and braze-welding. As still more oxygen is added to the mixture, the flame gets much shorter and bluer, and is known as an *oxidizing* flame. Each of these flames is progressively hotter; the oxidizing flame is hot enough to easily cut the copper material and should be avoided. The area just beyond the visible flame, called the outer non-luminous envelope, is the part of the torch that should be applied to heat the work. Never touch the intense cone of white near the tip nor its surrounding colored "beard" to the work.

The *ratio* of the gases determines what type (reducing, neutral, etc.) of flame is present; the *total volume* of both gases determines the working temperature. If the flame seems to be not hot enough to heat the work, increase slightly the amounts of both oxygen and MAPP® gas to maintain a neutral flame. Experience and practice will enable the user to quickly determine the correct heat for a given joint.

HOW LONG WILL IT TAKE?

Because the copper and other materials required for a ground system installation may have long lead times, pre-planning for the project should be started a month or more before the actual work is to begin. Determining the scope of the work, making a materials list, collecting competitive quotations from vendors, and arranging for any necessary building permits or easement marking for underground utility lines are all activities that can most easily be done away from the pressures of tight deadlines.

Inasmuch as the requirements of individual stations differ so greatly, it is difficult to generalize about the time needed to install radial wires. A one-tower station with no obstructions in its ground field might be able to install, on average, one radial wire every ten minutes (or six per hour). At that rate, the entire field of 120 radials would require twenty hours. The time could, of course, be cut in half if two tractors operated simultaneously in opposite directions. In the case of directional stations, the time required to cut and solder radials to the intersecting strap(s) must also be considered.

The interior ground system is much easier to quantify as its design will be the same for nearly all stations. The first landscaper portion of the interior square, clearing, excavation, identifying and flagging existing radial wires, installation of landscape fabric, the pressure-treated lumber square, and the first two inches of gravel (as detailed in steps 8 through 11 above) will take a landscape team of three working with a small tractor about two days. If existing radial wires are to be reused and soldered onto the new copper-topped lumber square, cleaning the ends of the old radials down to bright metal in preparation for soldering will consume about ten to twelve man-hours.

Installation and brazing of the actual copper components of the interior grounding system, including attachment of the radial wires, will ordinarily require 70 to 80 man-hours. A team of six, equipped with two complete welding sets, can do the complete installation of one tower's interior system in one very long (13 to 14 hour) day. If several towers' ground systems are to be rebuilt in a directional array, subsequent towers may take somewhat less time as the team members accustom themselves to the procedures.

Finally, landscapers will require another day with a team of three to top the completed copperwork with gravel.

CONCLUDING WORDS

The ground system is a critical component of a well-functioning AM transmitter site. Its installation cost in both time and money is so significant, its physical size is so large, and the opportunities for inspection and repair are so infrequent that any such installation project deserves to be undertaken with the greatest of care. Do not delegate inspection of the finished product to outside contractors, for it is you, not they, that will have to live with problems that develop months or even years later as a result of poor installation practice. Be completely satisfied with the condition of every radial wire and every silver-soldered joint before the first shovel is lifted to cover them up; you may never get a chance to improve matters later.

When the last gravel truck rumbles away and the ground system installation or replacement is over, very little if any evidence of the amount of work performed or materials consumed will be visible to the naked eye. A well planned and carefully installed ground system will, however, make its presence felt through years of stable, dependable operation to an appreciative station engineer.

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PREPARING FOR RADIO MULTICASTING AND THE R-LAN ENVIRONMENT

Skip Pizzi
Broadcast Engineering Magazine
Overland Park, Kansas

Economic and regulatory developments in radio have recently given rise to a significant increase in duopoly ventures. Among the many ramifications of this movement is a trend toward the origination of multiple audio programming services from a single facility. If properly implemented, such *multicasting* can offer substantial economies of scale in the production and presentation of these multiple program channels.

Serendipitous developments in broadcast technology have provided a useful method of managing these multiple streams from a co-located facility. Generally classed as the latest incarnation of *radio automation systems*, these devices are typically PC-based controllers, some of which can drive two or more simultaneous audio programming schedules from a single terminal.

The synergy of these two developments may resonate strongly in the next generation of radio facilities and their operation.

Consolidation

Duopoly actions now dominate the trading of radio stations, as broadcasters shuffle their assets among markets for best advantage. Some group owners now prefer to leave a market if they can't obtain multiple stations within it. (In markets with 15 or more commercial stations, a broadcaster can own up to four stations -- two AM and two FM -- as long as the combined market share does not exceed 25%. For markets of 14 or fewer commercial stations, up to three stations -- no more than two in either band -- can be commonly owned, as long as an owner's total number of stations does not exceed 50% of the stations in the market.)

Digital automation devices lend themselves well to the duopoly application. This current trend toward producing the same number of radio broadcast services in a given market from a smaller number of facilities can be called a *regressive consolidation*.

A related future trend that may be of value to some broadcasters uses the same technological approach to serve a different purpose. In such cases, an existing broadcast facility adds *new* service streams to its existing programming, which are delivered to listeners via alternate or newly available means. This allows an existing broadcaster to provide a programming channel (or channels) to a cable or telco audio service, or to separately program a new DAB channel, etc. Such a process could be termed *progressive consolidation*.

In both of the above cases, a digital automation system is the key to providing multiple services from where once only a single service originated, allowing an increase in programming output without requiring a proportional increase in staff and space.

Radio stations, today and tomorrow

Consider the assets of a typical radio station today: It is the hub of many incoming audio and data sources; it is the repository and archive of significant recordings of audio and data; it contains the means of original audio production; it employs a talented staff; and it has an established administrative infrastructure appropriate to its type of business. From a traditional viewpoint, it seems normal that all of these assets are brought together in a given facility to produce a single product. A more contemporary appraisal, however, might consider this input vs. output arrangement quite inefficient. The assets on hand at a radio station could more wisely be applied to the production of multiple simultaneous products under this modern view. Appropriate automation and facility design make this paradigm shift viable.

The radio stations of tomorrow therefore may produce numerous programming streams, destined for both wired and wireless deliveries. Consider also that audio will not be the only medium that these services consist of. *Auxiliary data* will also be contained in the mix of services offered by the station. Such auxiliary data could appear in any or all of the following three basic categories:

· *Program-associated data (PAD):* This data typically appears as alphanumeric text or control data (graphics might also be supported), which provides information related to the audio program currently airing. Text data is displayed on a screen contained in the radio receiver, with applications including station and program identification, title/artist of current musical selection, and "tell me more" data (such as phone numbers, bibliographic/discographic references or purchasing information). Control data provides "smart radio" features such as identification of alternate frequencies carrying the same program in neighboring areas, automatic recording or other peripheral device control, and emergency or traffic alerting.

· *Transparent data:* This refers to strictly ancillary services, most of which can be operated by broadcasters as separate profit centers. This data is not program-audio related, and it is not intended to be received on general-audience radios. This implies that separate devices are built and sold for reception of this data. These devices are usually addressable, and can be designed for a wide variety of specific, narrow applications. Service is typically sold on a subscription basis, and is normally generated by a third party who pays a fee to the broadcaster for carriage of the data signals over the broadcast service area. Applications include paging, data services, software downloads, database updates and remote device control. A station can also use some of this capacity for its own remote device control or for backfeed of data/audio to remote sites.

· *Auxiliary audio services:* Narrowcast, low-to-mid-fidelity secondary audio channels were the first auxiliary applications used in FM radio via the Subsidiary Communications Authority (SCA). Although such applications in FM radio today are limited generally to public service use (such as reading services for print-handicapped and agricultural reports), a wider range of applications may be possible in future DAB or wired delivery systems.

Therefore, the ultimate incarnation of the radio origination site will include methods for creation and assembly of both audio and non-audio data streams, delivered from the site via one or more analog and/or digital delivery channels. (In the digital case, the delivery channel is simply a data "pipeline," capable of carrying *any* kind of digitized data in a flexible and fungible manner.)

Note also that some auxiliary data is originated at the radio station with its programming (e.g., PAD or embedded station machine control commands), while other data is simply "managed" as it passes through the system from external clients' sources, and is inserted in the output data stream(s).

The Radio LAN (R-LAN)

The ideal environment for creation of this flexible output puts the entire radio station -- both studios and offices -- on a high-speed (multi-megabit) audio-plus-data LAN (local area network). Conceptually diagrammed in Figure 1, various production areas (newsroom, production studios, edit stations, etc.) share access to this LAN and its common mass storage facilities. (In actual practice, R-LAN manifestations may use *distributed* rather than the *central* core storage implied by Figure 1, thus avoiding access-time backups at a single storage device, and adding reliability and redundancy.)

In this environment, each room within the facility (e.g., newsroom, production studio, master control) has different terminal hardware and software designed to best serve the purposes of that room's typical functions, but all share a common operating system and storage.

A single computer system and operator could handle two (or possibly more) streams in a single master control room, accessing local and network programs from the mass-storage system, or switching live to any source, via preprogrammed schedules that have been downloaded from the office computer's traffic program. (This traffic-downloading function is already in place at some automated radio stations.) Automated recording of network programs for time-shifting purposes can also be handled in this way. Long-form programs could continue to live outside the computer (on DAT, for example), but such peripheral storage devices can also be placed under full computer control.

A station may want to continue "traditional" production/assembly methods for some programs, such as personality morning shows or talk radio, so at least one conventional radio studio will remain within the facility in many cases. On the other hand, other program streams might be completely automated, and comprised exclusively of externally produced materials, therefore requiring *no* physical space for production or assembly at the radio station (other than some rack or counter space for storage devices and controllers). These program streams are generated purely in the "cyberspace" of the computer system.

Alternatively, some dayparts of a particular stream could operate in the middle ground of the *live-assist* mode, in which an operator manually starts sequences of computer-controlled events from the automation control surface, intermingling them with live elements.

Careful scheduling will be required at this future facility, to most efficiently apply (and not over-commit) a station's physical assets to its multiple output streams.

Migration paths

The pathway from today's station to this future environment must be accomplished in a sensible, incremental fashion. In this respect, some unsettled issues remain. From what is known today, however, a few important conclusions can be drawn, which should influence broadcasters' actions during the intervening period as follows:

· *Make all equipment purchases with eventual centralized computer control in mind.* No significant new hardware items should be considered if they are not fully and flexibly operable (commands-in and status-out) via some reasonably common control protocol.

· *Begin the transition to the R-LAN in the master control room, not the production studio.* Using a building block approach, build the foundation of the R-LAN on a digital automation (or "audio management") system, not a digital audio production workstation (DAW). While the DAW may seem more appealing and "high-tech," most current designs do not include the ability to fully integrate the DAW into a station-wide radio environment. Even those that do include networking capability only envision interconnection to another identical production device, not a different platform/operating system in another (i.e., master control or newsroom) part of the station. In contrast, most well-designed digital audio management/automation systems *do* foresee such an arrangement, and some such systems are already under development or in production. The audio management/automation platform expects to be a hub for future spokes, whereas many DAWs comport themselves as self-sufficient islands.

· *Keep abreast of market and regulatory movements.* Multicasting is far from a purely technological pursuit. Just as business and regulatory trends have started the trend toward multicasting, continuing events in that environment will fuel or stunt its progress.

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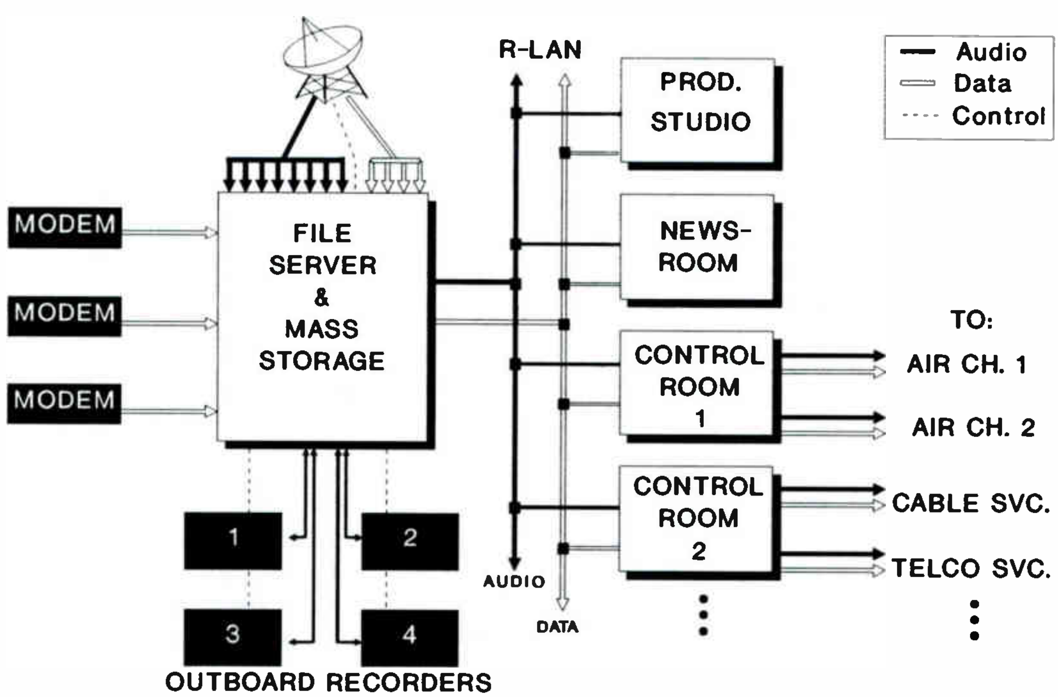

Figure 1. Conceptual diagram of the Radio LAN (R-LAN).

SBE

THE NEW EBS SYSTEM

Tuesday, March 22, 1994

Moderator:
Dane Ericksen, Hammett & Edison, Inc., San Francisco, CA

***THIS IS NO LONGER A TEST**
Richard Smith
FCC
Washington, DC

***TESTING NEW TECHNOLOGIES FOR EMERGENCY ALERTING SYSTEMS**
Paul Montoya
Broadcast Services of Colorado
Lakewood, CO

***CABLE TV AND THE NEW EBS**
Ken Wright
Jones Intercable
Englewood, CO

USER FRIENDLY EBS
Richard Rudman
KFWB/KTWV-Group W Radio, Inc.
Los Angeles, CA

ALL CHANNEL ALERT: CANADIAN EMERGENCY BROADCAST SYSTEM
C. Patricia Lok
Meteomedia
Montreal, Quebec

***A PARTICIPANT'S REVIEW OF THE FCC CONDUCTED TESTS LEADING TOWARD A NEW EMERGENCY BROADCAST SYSTEM**
Darryl Parker
TFT, Inc.
Santa Clara, CA

***A PARTICIPANT'S REVIEW OF THE FCC CONDUCTED TESTS LEADING TOWARD A NEW EMERGENCY BROADCAST SYSTEM—PART 2**
Jerry LeBow
Sage Alerting Systems, Inc.
Stamford, CT

*Papers not available at the time of publication

USER FRIENDLY EBS

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Abstract

The FCC is in the process of evaluating proposals to overhaul EBS, The Emergency Broadcast System. Most of the attention concerning the FCC's current EBS Rulemaking has been focused on alerting technologies. Operational issues, and the electronic media's relationship with local government and public safety need our attention as well if the critical process of alerting and informing the public during emergencies is to evolve to a truly user friendly and reliable system.

HISTORY

CONELRAD

The Emergency Broadcast System (EBS), started life as CONELRAD.[1] It was conceived amid Cold War fears of nuclear attack. It was born in 1951 to allow government to reach the public rapidly through designated radio stations in event of nuclear attack. EBS is now 40 something, and going through its own version of a mid life crisis. Sharing another trait of human offspring, EBS has always had its faults. Perhaps the writer who first scripted those ominous words, "WE INTERRUPT THIS PROGRAM..." , unintentionally set this tone of antisocial misbehavior. EBS joined the broadcasting family under the figurative image of the FCC shotgun. Broadcasters have had many problems over its history living with government imposed EBS technical standards from the start. CONELRAD frequency hopping took its toll on transmitters and the broadcast engineers responsible for their maintenance.[2]

After the Cuban missile crisis in the early 60's, CONELRAD became EBS. Fears that a potential enemy might jam the 640 and 1240 kHz. alerting frequencies began a trend to let more stations stay on the air on their own assigned frequencies to warn the public of impending doom. Government had acknowledgement by that time that bomb delivery technology did not depend on broadcast stations. Post CONELRAD broadcast transmitters still took the brunt of the EBS alerting function, the carrier break. A series of carrier breaks were part of the first EBS attention signal.

EBS Changes in the 70's

EBS has had to reexamine and reinvent itself. This self examination began in the 70's. In 1976, EBS was recast to alert the public for local and state emergencies. Operational Area meetings with broadcast engineers, hosted by the FCC, the National Weather Service, FEMA, and state emergency coordinators, set the tone. They presented the Parkersburg, West Virginia plan as a prototype. Parkersburg was a simple plan designed to deal with floods. Broadcast engineers in many EBS Operational Areas worked with their local emergency officials to create their own plans based on regional threats and disasters. This initial federal and state local level missionary work began to bring a higher and more productive sense of purpose to EBS. Local emergencies became the focus. Alerting the public about tornados, floods, and other weather-related emergencies became the primary mission.

Unfortunately, EBS did not lose its intrusive sense of interruption to programming, especially for the FCC-mandated weekly tests. And, not everyone in broadcasting was caught up in the federal government's missionary zeal. Broadcasters in regions where local emergencies did not take place periodically found it hard to develop a stake in the EBS mission.

In 1976, the FCC presented a series of technical changes to improve the reliability of EBS to perform its basic alerting function. Elimination of the most ill-mannered part of the attention signal, the carrier break, was an important technical change adopted at that time.[3] Rough on transmitter contactors, hard for board operators to perform with consistency, and dependent on special receivers that often malfunctioned, it was replaced with a more user and transmitter friendly two tone "Attention Signal".

The ubiquitous FCC EBS Checklist, the "Orange Book", outlined new Rules that spelled out EBS use for local widespread emergencies was written. It was developed by the FCC to help broadcasters comply with what the FCC thought was the right way to alert the public. The FCC was attempting to have EBS become, if not a welcome partner of broadcasting, at least more a more politically correct and locally useful partner.

Speaking of political correctness, almost everyone agrees that nuclear holocaust resulting from war is no longer the primary EBS mission. The self destruction of the USSR removed the last underpinnings from the tottering framework of a nuclear-driven EBS. However, few experts feel that a national alerting system should be scrapped. In fact, the Primary Entry Point (PEP) program, not yet fully operational, has been designed to do a better and faster job of national alerting should the need ever arise.[4]

FIXING EBS

Troubleshooting

Since broadcast engineers are troubleshooters, and EBS is so often thought of by programmers as an engineering function, our training can help focus attention on making EBS easier for government and broadcasters to activate. Let us troubleshoot the elements of EBS that still need to be addressed to get us to a more perfect world of user friendliness and EBS reliability:

Tests Are Not The Real Thing

" Tests good. Works bad." That sums up many print press and broadcaster EBS post-mortems for emergencies such as the Loma Prieta earthquake, the Oakland firestorm, and the 1992 riots in Los Angeles. Intrusive weekly tests have blunted the impact of the alerting function of the so-called two tone Attention Signal. Years of tests have led the public to a tune out before they hear if it heralds a test or a real alert. That same tone has hardened many operations personnel within broadcast facilities to turn off to EBS as well. Many operators do not wait to see if the Attention Signal from the CPCS-1 station they monitor precedes a test or a genuine alert before they reset the EBS monitor and complete their FCC-mandated log entry.

Decision Indecision

" In Case of Fire, Break Glass" signs are a classic example of a system that can cause hesitation at the wrong time, or what might be called Decision Indecision. The bigger the consequences of an emergency condition, the more an individual or group might balk before making a response decision. At the government end, where EBS activation requests are supposed to originate, Decision Indecision often delays activation requests. Broadcasters often have the information that triggered the activation request on the air as straight news, sometimes hours earlier. At broadcast facilities, Decision Indecision may delay honoring a government request for an EBS activation. Many staff people, in government or the electronic media, are reluctant to make judgement calls until they can get their manager on the phone.

Decision Indecision can keep information from people who need it most, at times when that information can do the most good. The National Weather Service, the Army Corps of Engineers, and the City of Los Angeles have been accused of a form of Decision Indecision when heavy rains inundated the Sepulveda Dam flood plain in 1992. Roads were not closed and information was not sent out to the public in time to prevent vehicles and their hapless occupants from being trapped in the very area designated as a flood plain. EBS was also not activated during the fire storm in Oakland that destroyed thousands of homes in 1992.

Cry Wolf There are usually no hard and fast protocols for what might trigger a proper request by government for EBS activation. Judgement is hard to translate to words in an operational manual. Its proper exercise requires more than just procedure and authority. It is also easy and legal for a government agency that does not plan and practice its path to sound judgement to request an EBS alert at the first hint of a cloudy sky. This means that there is always a potential for a "Cry Wolf" Syndrome, the opposite of Decision Indecision. The result is not in the public interest. "Crying Wolf" will ultimately backfire, leading to broadcasters becoming turned off to the trust and cooperation with local government that is vital during emergencies. It can also give the public more reasons to ignore EBS.

EBS Levels Have Never Been Integrated Systems engineering between the various EBS originators has been another missing ingredient since local emergencies became part of the EBS recipe. The technical and operational means now used to accomplish National, State, and Local alerting and information delivery lack coordination and consistency. The current EBS Rulemaking is an attempt by the FCC and industry to create a unified field theory for the technical part of EBS. Some would argue that the Rulemaking does not go far enough to address many EBS operational realities.

State and Local EBS Are Voluntary

National EBS is mandatory, state and local EBS are not. Per FCC Rules, stations who do not agree to participate in National EBS advance must leave the air if a National EBS Alert is declared. Not so with local emergency activations. The Weather Service, a state, or local government can only make activation REQUESTS. A station may legally choose not to honor such a request. It is this very point that leads to much confusion when a station either does not receive or honor a request for activation.

EBS Orange Book Issues

Terminology and the layout of the FCC's EBS Checklist, the "Orange Book" does not lend itself to easy and rapid use during the heat and panic of a State or Local activation request. Many areas of the country do not face routine floods, earthquakes, or other emergencies that provide real-time training for station personnel for EBS activations. Relying on the EBS Checklist, the "Orange Book" alone, ignores that each station may have:

- Different criteria (or no criteria) for honoring activation requests.

- Different ways (or no way) to get EBS announcements on the air.

- Minimum wage and/or minimum trained operators often left on their own to make (or not make) emergency decisions

- Panic even experienced operators feel when faced with duties and challenges not seen in routine operations.

These elements add up to the true state of EBS readiness in many stations. FCC inspectors never check to see if licensees have planned to deal with activation requests since there is no provision in the Rules for such a check. Everything is fine until an emergency occurs, an activation request is made, and the ball is dropped at the station level. Since most people do learn from their mistakes, things usually go better for the second earthquake, the second flood, or the second riot. What we really need is a system that should be able to bring us all to an instant state of proper actions when the first "surprise" happens. Does the current Orange Book help? Pages 11-12 of the EBS Checklist, the "Orange Book", deals with State and Local Level Instructions and Tests. Under ACTIVATION it says,

STATE LEVEL: A request for activation may be directed to the Originating Primary Relay Station by the Governor, his designated representative, the National Weather Service, the State Civil Defense or State Office of Emergency Services.

LOCAL LEVEL: A request for activation may be directed to the Common Program Control Station - One (CPCS-1) by the National Weather Service, local Civil Defense, local government or public safety officials.

Catch 22 Local EBS may also be activated by any AM, FM or TV broadcast station at management's discretion in connection with day-to-day emergencies posing a threat to the safety of life and property.[5]

While new EBS technology will hopefully make moot the need for Operators to really understand the above verbiage, we must make sure that words and phrases Operators do need to understand are well written, clear and obvious. No amount of training that the broadcast industry is currently capable of doing will ever get most operators to really understand what Originating Primary Relay Stations, or Common Program Control Station are.

Page 12 of the EBS Checklist speaks about IMPLEMENTATION. An Operator who may never have done an actual EBS alert, is supposed to calmly turn to this section in an emergency, and read announcements called for in items b) and d) verbatim. The operator then must do the following:

a) Record the emergency program material if possible.

b) Broadcast the following announcement

WE INTERRUPT THIS PROGRAM BECAUSE OF A (STATE OR LOCAL) EMERGENCY. IMPORTANT INFORMATION WILL FOLLOW.

c) Broadcast the two tone Attention Signal for 20 to 25 seconds using the EBS Encoder.... (See Sections 73.906 and 73.940 of the Rules).

d) Broadcast the following announcement:

WE INTERRUPT THIS PROGRAM TO ACTIVATE THE (NAME OF THE STATE OR EBS OPERATIONAL AREA) EMERGENCY BROADCAST SYSTEM AT THE REQUEST OF (ACTIVATING AGENCY OR OFFICIAL) AT (TIME)."

e) Broadcast the material from 3.a.[6]

One way to address the Orange Book problem is to simplify and package Operator decisions in advance. At KFWB in Los Angeles, the appropriate parts of the "Orange Book" for National, State and Local EBS are reduced to concise labels for a special set of tape cartridges kept both in the Master Control area and the Air Studio. For example, once a Local EBS activation request is received, an Operator has only to locate the correct three cartridges. Operators only have to read a few lines of clearly written instructions on the cartridge labels, and play the cartridges in the right order. National, State, and Local EBS sources are available on the station routing switcher so they are more likely to be found during the heat and panic of an emergency. While not specifically addressed in the Rulemaking, the FCC should consider a review and overhaul of the Orange Book.

Present EBS Concentrates More On Alerting Than Information Delivery

Weekly EBS test message makes the promise that broadcasters will "...keep you informed in the event of an emergency." Most EBS messages alert our audiences only to the bare facts of an emergency. We forget that each EBS activation ends with a termination notice that says:

THIS CONCLUDES OPERATIONS UNDER THE (NAME OF STATE OR EBS OPERATIONAL AREA) EMERGENCY BROADCAST SYSTEM. ALL BROADCAST STATIONS MAY NOW RESUME NORMAL BROADCAST OPERATIONS.[7]

An EBS premise, expressed in the test message, is that local activations have a short life cycle, and life quickly returns to 'normal'. While true for tornados, this is definitely not the case for many other types of regional emergencies. Many local emergencies go on for hours, if not days or weeks. EBS often only supplies a brief burst of information after the alerting function has been accomplished. The weekly test message promises the public a flow of information that lasts as long as emergency response and recovery dictate. EBS, as currently structured, often does not deliver on that promise to provide both the alerting function and deliver pertinent information for the duration.

We can learn a great deal from real emergencies about where people go for their information, and why they go there. Ratings during emergencies repeatedly show that people gravitate to stations that supply a sustained flow of information. Audiences living under emergency conditions are information-hungry. Few people really want to be returned to 'normal programming', as EBS philosophy has always assumed, when their safety is at stake. A sudden voracious appetite for information surfaces when audiences are alerted to emergencies that threaten their lives and property.

In Southern California, earthquakes awaken a craving for every tiny morsel of trivia and detail. Moments after the ground stops moving, broadcasters begin fulfillment of this need with just our experience of the tremor. Just like our audiences at that very moment, we know not if our loved ones are safe, if we will lose fresh water for two months, or if we have experienced an event that is just a prelude for an even larger seismic shaking. It is just this sort of emotional and incomplete information that broadcasters need to supplement with real facts, directives, and reassurance from public safety officials. The alerting function of EBS does not speak to this need.

The range of format options in most markets leaves most citizens with clear choices. What is missing is an EBS philosophy that embodies that concept of that steady flow of information to broadcasters from government and public safety officials, supplemented by information that broadcasters develop independently.

California, and regions like the Los Angeles County EBS Operational Area, have been working on just this problem. EBS, has been tried in the balance of major earthquakes, floods, and fires since the early 70's, and has been found wanting. Local government officials and broadcasters have, for the most part, stopped blaming each other for EBS failures, and are working together to devise systems that really work.

EBS Compliance: The Stick to Carrot Ratio Compliance with provisions of the FCC's Rules on EBS is most likely motivated more by the fear of fines and other penalties than by a true sense of public service. This is not the way to foster true voluntary cooperation.

ATTEMPTING REPAIRS

The Media/Public Safety Connection

How can we bring our troubleshooting skills to fix what is wrong with EBS? Let's begin by stating the obvious. The worst time to get to know local government officials who will be making local EBS activation requests is when the emergency happens. Public safety officials and staff need to know what makes broadcasters tick so they will be better prepared to request timely and effective activations. In many localities, it has been up to broadcast engineers to be the liaison agents. Their first task is to find the right officials in local government to talk to.

Both government and broadcasters sometimes miss the point that local and state EBS is voluntary. As pointed out earlier, a state, local Operational Area , or the National Weather Service can only request an EBS activation. There is no penalty for broadcasters NOT honoring activation requests, a double negative that impairs our ability to perform in the public interest during emergencies. Given the disruptive and somewhat burdensome FCC rules that come into play if we do decide to honor an activation request, most broadcast managers believe they may suffer perceived penalties if they do honor a request. This may be a classic case when they believe that no good deed goes unpunished.

Obviously there are rewards and benefits to honoring activation requests, but their perceived value to a given broadcaster may be format dependent. All News radio facilities get much more of a reward from going "All Emergency All The Time" than the average music station. Indeed, FCC records over the past 20 years show evidence that a significant number of CPCS-1 stations who have music formats have NOT activated EBS during emergencies when local or National Weather Service "requests" were made.

While some CPCS-1 stations with news formats have also not honored activation requests, a case can be made that they were acting in the public interest and convenience by continuing to do just what they do every day.[8] As stated earlier, the current FCC Rulemaking has been structured to deal primarily with the format and reliability of the alerting function. The Rulemaking does not fully address the additional issues of:

- Public safety officials and how they deal with the realities of issues related to deciding on and making activation requests

- Broadcasters honoring proper requests for activations in a timely manner to protect lives and property

- The relationship between public safety, government and media

While it can be argued that the FCC has no business telling local government how to conduct its business, the public has a right to some system-wide consistency. It can also be argued that the FCC must mandate whatever it takes to assure that a strong relationship is forged between originators of emergency information, and the media responsible to convey this information to the public. The strength of that relationship between public safety, government and media is the real key to providing effective and timely emergency information.

Emergencies like floods in the midwest show that a strong link between public safety and the media can help save lives and property, help hold the line on law and order, reduce panic, and, to some extent, shorten the response and recovery periods of an emergency. Such a link goes beyond the FCC's EBS current Rules. When emergencies suspend the everyday rules of electronic media competition, we are all placed in a common survival mode. A more universal understanding of this concept should lead to greater public safety and broadcast media cooperation. A lack of this strong link, as evidenced when EBS activation requests should have been made or should have been honored, proves the converse to be true. Public safety and broadcast media have the common denominator of both being franchised to act in the public interest. This premise should be the departure point for cooperation that will forget the trusting relationship that must be in place well before it is needed for mutual survival.

Both local government and broadcasters have another common denominator during major emergencies — fear. During a major emergency, when we share apprehension for our personal safety, and for the safety of our families and friends, we also share the feeling of being passengers in the same leaky boat. We need to work together for those special times when we need each other. At those times when survival is at stake, the operative mind set should be "we"; not "us" and "them".

Broadcast engineers have so far taken the lead to drag news and program managers into the process in many part of the country. Where this has not yet taken place, the first task is to get engineers, news management, news professional organizations, and local government emergency planners together to do prior planning. Major benefits begin to accrue once this process is started. By getting to know one another and sharing ideas, a sense of common purpose and cooperation is built. This shared sense of purpose is invaluable when the next emergency happens. The worst time for broadcasters and local emergency officials to meet for the first time is after an emergency begins. If this process has not been started in your Operational Area, the agenda for the first series of meetings might look like this:

• Review potential threats, and disasters

• Begin work on what you might expect to hear from government, and from whom

• Outline what audiences will hear when they hear of the emergency

• Identify government people/organizations who will make activation request decisions

• Identify key individuals who should be "pageable" in an emergency

• Identify which communications links to the media must be "hardened"

BEYOND EBS

Information Flow During Emergencies

In the Los Angeles County Operational Area, we are reaching the point with public safety and government where we are now addressing core issues of emergency information philosophy. We are beginning to focus more on information flow, not just on the alerting mechanism. There is a growing general agreement here that the public wants and deserves more than short term interruptions of programs during major emergencies. Conversely, for less significant emergencies that do not warrant an activation request, a priority framework is needed so those who are affected stay informed. We are rapidly reaching the point where these two concepts are not seen as requiring separate information delivery systems. Why not make them both part of the same system?

Message Priority Levels

One way to accomplish this is to view the flow of public information during any emergency as a planned escalation of data as it becomes available. The five message priority levels, as is in place now in California's EDIS (Emergency Digital Information Service) are:

TEST - Verification of equipment operation and training exercises

INFO - Advisories to the news media regarding upcoming or in-progress events to help coordinate media-government interactions

NEWS - Information the public needs to cope with an emergency or unusual situation

URGENT - Time Critical news and information regarding an emergency or unusual situation

FLASH - Immediate warnings of threats to life and property, usually requiring action to be taken by the public

Wireless Packet Delivery

A wireless packet delivery "wire service"using these priorities in a familiar and usable format has already become a valuable newsroom tool in several parts of California. EDIS is based on the American Newspaper Association (ANPA) wire service protocol. For example, with EDIS linked as a wire service to a radio or TV station's newsroom computer, editors can receive vital information instantly. The same system, again within the framework of the five EDIS priorities, can, during and after emergencies, easily and rapidly carry:

- Decisions citizens must know about
- Updates on changing situations
- Press briefings (especially when not all media can get to the briefing site)
- Background information

FEMA Involvement

This system is fast. During the Los Angeles brush fires in November, 1993, emergency information released by FEMA on EDIS was on the air on in just four minutes. FAXing more than 60 copies of this information would have taken hours, with no assurance that any newsroom decision makers would ever see the information. FEMA has already identified EDIS as a significant improvement in emergency communications. It is assisting California State OES to promote its use, and is considering EDIS for its own use during major regional emergencies.

A Los Angeles County low band VHF radio channel is used to distribute EDIS [9]. Another LA County radio voice channel will be used to originate live or taped voice feeds from the Los Angeles County and LA City Emergency Operations Centers (EOC). Government is eager to support this idea since it is the only way they can be assured that they can reach news decision makers when they most need to do so. Telephones go unanswered. Scanners are lost in the clutter of newsroom sounds. FAXed information, as mentioned before, takes too long to get out to a large number of broadcasters, and runs the risk of getting lost in FAX clutter. and But, the BULLETIN alarm in a newsroom computer always gets through.

EBS: THE EQUIPMENT

Ease of Use: Government and Media

An important element of user friendliness will be how well the new EBS equipment will actually work in the field. Its success depends on ease of use at both ends. The activation request end at the public safety activation point should be easy for this community of users to understand. The activation reception end at AM, FM, TV and cable companies should be designed for people who may not either know or care what an EOC is, or what a particular public safety acronym means. New EBS terminal equipment must be able to successfully manage and coordinate a variety of National, state, and local voice and data inputs. Station personnel will still have to exercise certain decisions once alert requests are sent.

The author has conducted an informal study on the type of computer interface that might best be suited for both ends of this system. It is based on the simple idea that almost everyone in today's society knows how to use a bank Automatic Teller terminal (ATM).

Such a device might look like a bank automatic teller terminal (ATM). A series of buttons could be located on each side of the screen. Depending on what each information screen says, and where it says it, the buttons would change function. Such a system would be self training. Once it had been programmed according to station management's desire for how their station will respond for various types of emergencies, operators would never need to refer to a separate Check List gathering dust on the wall. The Check List, the Orange Book, would be imbedded in the terminal. Since originators of activation requests in public safety have many of the same personnel training and equipment operation concerns we face in broadcasting, an ATM-like device would help them as well.

CONCLUSIONS

- The FCC overhaul of EBS alerting technology will not solve all of the problems of EBS.

- New EBS equipment will have to accommodate a spectrum of local needs that cannot be resolved into one simple electronic package. Broadcasters will have to work with local officials to plan how this new equipment can best be used.

- If broadcasters work at the local level with government and public safety officials, there is a much greater chance that the system will work when it is most needed.

- EBS should be made a much more integral part of all formats. Once this is done, it will be viewed as an cooperative partner, not as an interruption.

ACKNOWLEDGEMENTS

Thanks to Dane E. Ericksen, P.E., Chairman, SBE FCC Liaison Committee and Arthur Botterell, Telecommunications Engineer with the California State Office of Emergency Services for helpful suggestions during development of this paper.

ENDNOTES

1. CONELRAD stands for CONtrol of ELectromagnetic RADiation. Fearing that enemy bombers would use broadcast stations as navigation aids, most stations were required under FCC Rules to go off the air if an alert occurred. Specially equipped stations would stay on the air on 640 and 1240 kHz., but would not leave their carriers on long enough to provide an accurate "fix" for navigation.

2. Manufacturers provided CONELRAD capability for transmitters by including separate oscillators and tuning components. Relays switched oscillators, and contactors switched tuning elements. It should come as no surprise to broadcast engineers not old enough or unlucky enough to have worked with transmitters so equipped that they were temperamental at best, and prone to excessive failures at worst.

3. The other portion of the attention signal was a 1000 Hz. sine wave tone.

4. The Primary Entry Point (PEP) program involves 30 continental U.S. AM broadcast stations and seven additional off shore stations to cover non continental U.S. regions under FEMA. The system is designed to allow the President of the United States access to a calculated 90% of citizens. Stations in the PEP program have agreed to have their programming interrupted for such alerts. Remote controlled devices at transmitter sites can be commanded by equipment that travels with the President that is under control of the White House Communications Agency (WACA).

5. Broadcasters rarely activate EBS unilaterally. There are few cases where broadcast management is present during situations where they could or should act without consulting local public safety officials.

6. Page 12 continues with sections that deal with TERMINATION and RECORD KEEPING. While there are several blank pages for station notes, the Author has seen very few Orange Books where such notes have been written.

7. Page 10, EBS Checklist, as revised in 1987.

8. The phrase "In the public interest and convenience" appears in the Communications Act of 1934 and has been quoted often by the FCC to point broadcasters in directions that run parallel to the intent of that Act.

9. EDIS, the Emergency Digital Information Service, developed by the California State Office of Emergency Services. EDIS is a protocol for data transmission that follows the American Newspaper Publishers Association (ANPA) protocol. Using simple Amateur Radio Packet AX.25 technology, an EDIS feed can be easily connected to a personal computer, printer, or even directly to a newsroom computer. The EDIS protocol is open ended to change as the rapidly changing face of information technology changes. It will, for example, soon be transmitted throughout the state via satellite.

ALL CHANNEL ALERT: CANADIAN EMERGENCY BROADCAST SYSTEM

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Abstract

The All Channel Alert communication system was developed to receive messages from a remote source and to display the information on selected channels of a cable television system. The initial configuration of this modular system specifies for the capability to process up to 36 channels upgradable to 108 channels by connecting three fully loaded units in a daisy chain. Environment Canada will be the primary user of the ACA delivering severe weather warnings to local communities. At the same time, other federal departments will also have access to this Emergency Broadcast System (EBS) via an input centre located in Ottawa.

INTRODUCTION

The ACA was designed and developed by Pelmorex Communications Inc. to distribute messages to any area serviced by a cable television system. Its control is made through the company's existing communication infrastructure to transmit and receive messages.

The distribution of messages is on a system-by-system basis, from a single system or a group of systems to full national coverage. All ACA messages can also be displayed individually according to site or language, as is required. The insertion of messages is on an individual channel basis thus allowing for minimal interruption to the on-going programming.
This paper will explain briefly the design of the ACA system and how it will be implemented in Canada as the national Emergency Broadcast System.

THE ALL CHANNEL ALERT (ACA)

The ACA unit is located in a cable headend. The distribution of the field units will determine the coverage. The cable television infrastructure provides the link between the ACA system and the home viewer. As shown in Figure 1, the unit located at the cable headend is configured to receive information from the Message Originator via the space sector. The ACA Command Centre processes the information and oversees the administration and control of the remote units.

The *Message Originator* can be any body mandated to provide emergency information. In the first instance, Environment Canada's Atmospheric Environment Services (AES) will be the primary user of the ACA to broadcast severe weather warnings to affected communities. They have direct access to the Command Centre. Other authorities within the federal government will have access to broadcast emergency announcements via the national Government Emergency Operations Co-ordination Centre (GEOCC) at Emergency Preparedness Canada in Ottawa.

The *Command Centre* is the pre-processor of the emergency information and exercises control over the remote units. The centre uses the satellite signal to transmit control data and emergency messages to the ACA. It includes a central computer used to validate the messages and to ensure that they meet the required standards and addressing information for the remote units. Once the messages have been validated and addressed, the central computer formats and transmits the information according to the protocol supporting the ACA units.

Communications with the ACA Command Centre is by using any means available today. Except for telephone communications, the information leaving the Message Originator will be processed and displayed on the television screen

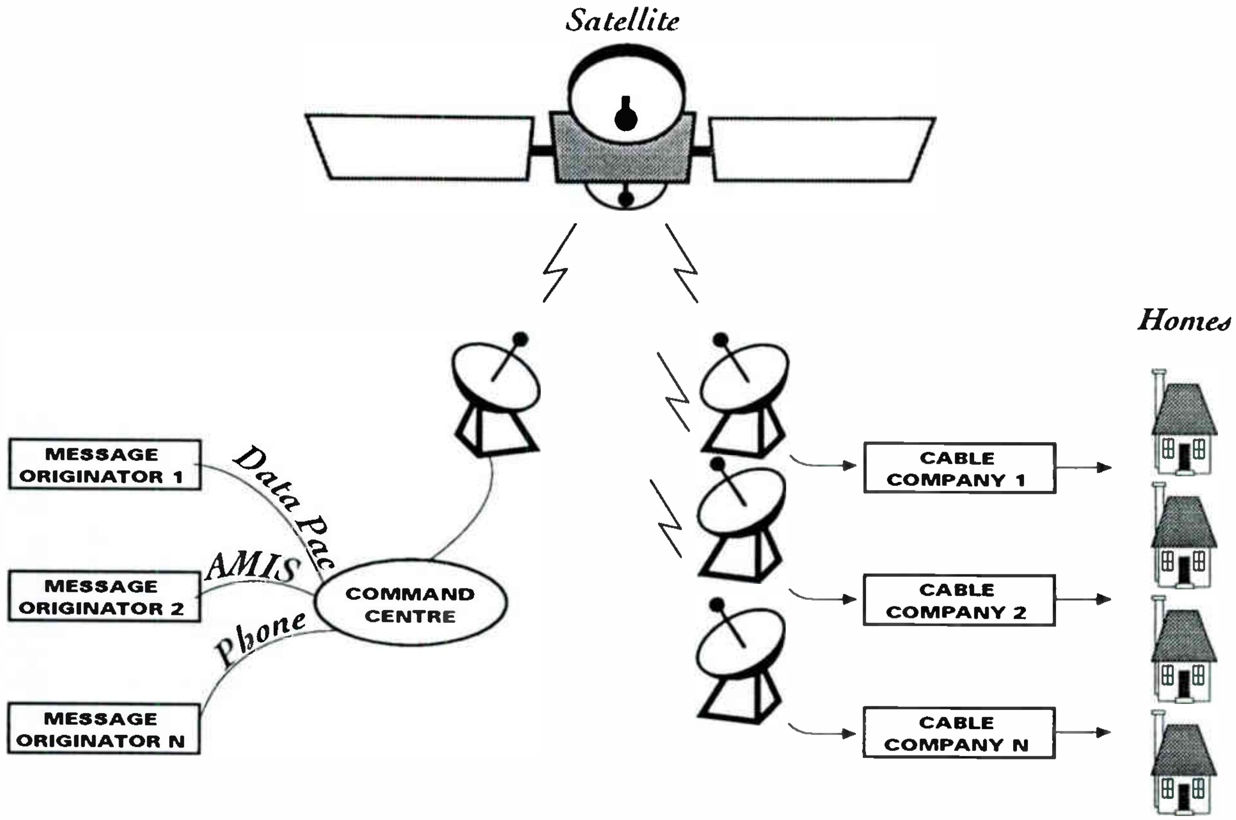

FIGURE 1: ACA SYSTEM OVERVIEW

in seconds, without human intervention. The mean time for the broadcast of AES weather warnings is currently four minutes using the existing system as they require manual addressing.

The ACA Chassis

The ACA chassis consists of three compartments in which the communications boards are housed separately from a PC-AT backplane and the power supply. Each component is removable as required to provide for future technology changes, for example from an analogue to digital environment.

Figure 2 illustrates a typical analogue installation at a participating cablesystem. A controller accepts from a receiver analogue audio and NTSC video for each source channel, it then modifies the signal prior to inserting it into the cable system's channel modulators.

The system establishes the level of intervention on each individual channel based on the priority level of the message. This functionality dictates how the message will be displayed on each of the channels on the cable system. Each ACA contains circuits to process up to 36 channels, organised as nine PCB's, each handling four channels; up to three populated ACA's may be daisy-chained to obtain 108 channels.

Each channel is provided with a character generator circuit capable of overlaying the video signal with one full page, or 240 characters of text, organised as 10 lines of 24 characters each. The character generator is fully synchronised with incoming video providing minimal disruption to the viewer.

Under normal conditions, the ACA unit is a passive device which is transparent to incoming video/audio signals. Under active conditions, upon receiving an input signal, the unit implements operations in response to commands and data sent from the Command Centre via the satellite link and controller.

The ACA supports the following functionalities:

(1) Superimpose text as a crawl message on the cable channel; the crawl message is synced to the incoming video.

(2) Superimpose text as a static text window (up to 10 lines of 24 characters per line) on the cable channel; the text message is synced to the incoming video.

(3) Replace the cable video signal with an override video signal. This switch is not synced.

(4) Superimpose text as a crawl message onto the override video signal; the text is synced to the override video.

(5) Superimpose text as a scrolled full text page onto the override video signal; the text is synced to the override video.

(6) Replace both channels of the cable audio signal with the override audio signal. This feature may be used independant or in connection with any of the above.

These features provide the ACA with the capability to display emergency messages with varying degrees of disruption to the cable television viewer. The least disruptive mode of operation is (1) which involves displaying a crawl text message, and the most disruptive is (5) which replaces the cable video with the override or replacement video and superimposes a full text page.

FEATURES OF THE ACA

The ACA was designed to provide maximum emergency notification with minimum interruption to the scheduled programming of a cable television station. This multi-leveled system enjoys such attributes as satellite communications and the ability to address one or a group of cable systems individually.

The target cost for a production version of the ACA system in volume is $50 CDN per channel. Therefore, a unit equipped to handle 36 channels will cost $1800 or less.

The ACA system allows the viewer to be alerted by a brief emergency announcement which directs the viewer to turn to a pre-determined channel for full details of the emergency warning. For example, if a severe thunderstorm is forecast for the viewing area of the cable station, the central computer will download a single line text message which is to be displayed on all or selected cable channels other than the one channel which will display full details.

The company's two networks, described below, currently broadcast all weather warnings issued by the AES of Environment Canada, always on a priority basis. Therefore in this example, they would be the designated stations to carry full details of the severe thunderstorm.

ROLE OF MÉTÉOMÉDIA/THE WEATHER NETWORK

Pelmorex Communications Inc. owns and operates two national satellite-to-cable networks. MétéoMédia (MM), broadcasting in French and The Weather Network (TWN), broadcasting in English across Canada, provide weather and environmental information to over 7 million households in Canada through more than 300 cable affiliates. This represents a penetration of almost 80% of cabled households.

As operators of the ACA system, MM/TWN are immediately prepared to act as the host broadcaster of emergency information by utilising their installed technology and broadcast experience. In their capacity, MM/TWN offers a fully bilingual 24 hour/7 day manned operations centre and existing communications infrastructure including a communications controller already installed in over 300 cable systems. The EBS will enjoy this ability to exercise real time control of data distribution and this experience of delivering to local communities the important emergency messages.

DISTRIBUTION REQUIREMENTS

Addressing the Messages

The distribution of weather warnings has in the past suffered from potential delays and inaccuracies. A universal scheme should be adapted across the country to allow for automatic dissemination and processing of important information. It is possible to distribute warnings locally with the use of the Canadian postal code, which allows for various zone sub-divisions and also provides a county and town hierarchial addressing structure.

It should be appreciated however that this scheme must be supported by a set of national standards to create a secure and inclusive operating environment.

Message Structure

All emergency messages sent to the Command Centre will include header information such as:

- Originator Identification
- Type of warning
- Affected zone(s)
- Message lifespan
- Priority level
- Language of broadcast

IMPLEMENTATION OF THE ACA

Studies have shown that national events triggering the use of a warning system are rare in comparison to local and regional use. However, the federal government must play an active role in organising national emergency operations standards as regards system access and message formats for all users of the system.

Environment Canada and Emergency Preparedness Canada (EPC) will have access to the ACA on the national level. As a second phase, provincial and regional bodies would be brought on line through EPC.

It should be appreciated that the ACA could reach up to 70% of the Canadian population. Therefore the support of the cable television and broadcasters associations is key to ensure the success of this project. The signal owners must be re-assured as to the low frequency of disruption and fail-safe features of the system to protect their viewers and advertisers.

CONCLUSION

Pelmorex Communications Inc. are prepared to act in collaboration with the various levels of governments as well as the broadcast and cable organisations to establish, install and operate the Canadian Emergency Broadcast System. This system would employ the existing satellite technology and facility of the company.

The system's product stream meets the requirements of both local and distant signal owners by synchronising with the incoming broadcast signal. Allowing for multiple levels of interference with normal broadcast operations, it ensures that the response is commensurate with the event. Furthermore, it allows for voluntary participation by channel and the status of this participation can change either in real time or only during "off-air" time. Finally, it allows for a real-time provision of signal to local broadcasters for inclusion in their "off-air" products.

The existence of a continuously manned operations centre and landline access from any point in Canada allows for a system which is open to support civil authorities for emergency measures. This is supplemented by the ability to accept a local video input and a local data input to further supplement this functionality.

The first field testing of the ACA system across Canada is targeted for early 1994. Through the cooperation of private enterprise and the public sector, the ACA system has evolved as a unique and innovative solution to the long-standing challenge of implementing an emergency broadcast system in Canada.

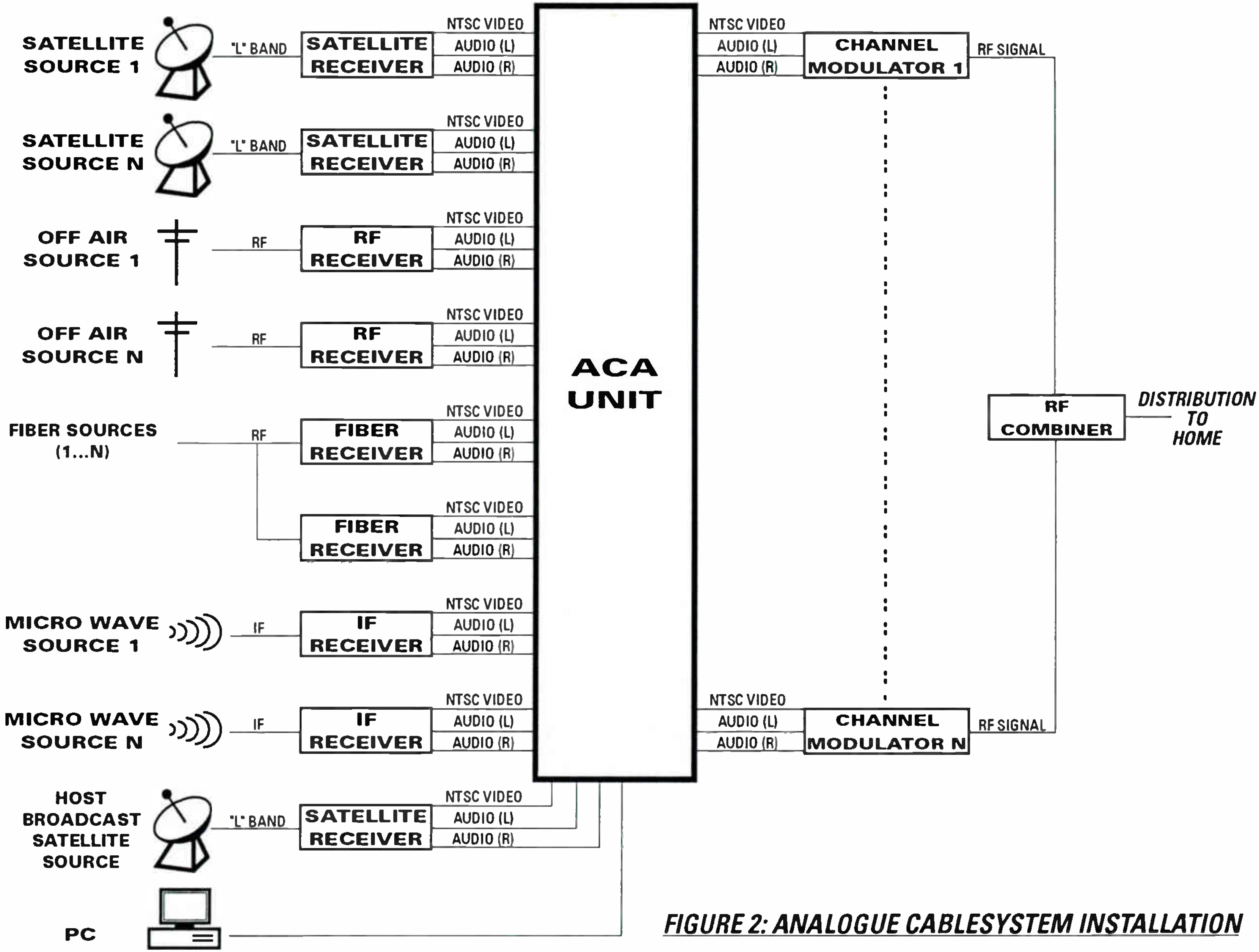

FIGURE 2: ANALOGUE CABLESYSTEM INSTALLATION

SBE

TV AUTOMATION

Tuesday, March 22, 1994

Moderator:
Jerry Butler, WETA, Washington, DC

***INTEGRATING AUTOMATED AND MANUAL OPERATIONS IN MASTER CONTROL**
John Wadle
ICA Systems Group
Clarksville, MD

***THE IMPACT OF DATA STORAGE SYSTEMS ON TV AUTOMATION**
Ray Baldock
Odetics Broadcast
Wayne, NJ

***WBNS—STATION AUTOMATION EXPERIENCES**
Marvin Born
WBNS TV-10
Columbus, OH

***TOTAL STATION AUTOMATION FOR THE '90S**
Douglas Hurrell
Alamar USA, Inc.
Campbell, CA

***WFMZ—STATION AUTOMATION EXPERIENCES**
Barry Fisher
WFMZ TV
Allentown, PA

***EMBEDDED ENCODING FOR POSITIVE EVENT IDENTIFICATION**
Daniel Wasserman
Cyphertech Systems, Inc.
Thornhill, Ontario

*Papers not available at the time of publication

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DIGITAL TV TEST AND MEASUREMENT WORKSHOP

Tuesday, March 22, 1994

Moderator:
Peter Smith, NBC, New York, NY

***DIGITAL TELEVISION IN THE STUDIO**
David Fibush
Tektronix
Beaverton, OR

***SIGNAL QUALITY ANALYSIS**
Dan McGee
Hewlett-Packard Company
Santa Clara, CA

***DIGITAL TELEVISION TRANSMISSION**
Dieter Scherer and Brian LeMay
Hewlett-Packard Company
Santa Clara, CA

*Papers not available at the time of publication

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THE BUSINESS OF CONTRACT ENGINEERING

Tuesday, March 22, 1994

Moderator:
Terry Baun, Criterion Broadcast Services, Milwaukee, WI

***AVOIDING THE PITFALLS IN CONTRACT ENGINEERING**
Dave Biondi
Broadcast Service Company
Houston, TX

***SOUND BUSINESS PRACTICES FOR CONTRACT ENGINEERS**
Paul Montoya
Broadcast Services of Colorado, Inc.
Lakewood, CO

***JUMPING THROUGH THE HOOPS: TIPS FOR KEEPING YOUR CONTRACTING BUSINESS LEGAL**
Christopher Imlay
Booth, Freret, and Imlay
Washington, DC

***BEING A GOOD CONTRACT ENGINEERING BUSINESS PERSON**
Mark Persons
M.W. Persons & Associates, Inc.
Minneapolis, MN

*Papers not available at the time of publication

FCC-INDUSTRY TECHNICAL PANEL

Wednesday, March 23, 1994

Moderator:
Michael Rau, NAB, Washington, DC

Panelists:

Harvey Arnold
The University of North Carolina
Research Triangle Park, NC

Dave Baylor
DirecTV/Hughes Communications
El Segundo, CA

Bill Hassinger
Federal Communications Commission
Washington, DC

Charles Kelly
Broadcast Electronics
Quincy, IL

E. Glynn Walden
Group W
Philadelphia, PA

*Papers not available at the time of publication

VIDEO COMPRESSION

Wednesday, March 23, 1994

Moderator:
Tony Uyttendaele, Capital Cities/ABC Inc., New York, NY

***VIDEO COMPRESSION TUTORIAL**
Dr. Didier Le Gall
C-Cube
San Jose, CA

***PRACTICAL IMPLEMENTATION OF MPEG VIDEO COMPRESSION PRODUCTS**
Mike Windram
NTL
Winchester, Hampshire, United Kingdom

***A VIDEO COMPRESSION EFFICIENCY ANALYSIS USING PROGRESSIVE AND INTERLACE SCANNING**
Eric D. Petajan
AT&T Bell Laboratories
Murray Hill, NJ

***VIOEO RESIZING—HOW TO MAKE THAT BIGGER/SMALLER IMAGE THE BEST IT CAN BE**
Jordan Du Val
Genesis Microchip Inc.
Markham, Ontario, Canada

***PC VIOEO COMPRESSION/OECOMPRESSION TECHNOLOGY**
Steve Patzer
Intel Indeo Video Group
Santa Clara, CA

*Papers not available at the time of publication

HDTV SYSTEM TESTING

Wednesday, March 23, 1994

Moderator:
Edmund Williams, PBS, Alexandria, VA

***DEVELOPING THE LABORATORY TEST PLANS FOR THE GRAND ALLIANCE ADVANCED TELEVISION SYSTEM**
Mark Richer
Public Broadcasting Service
Alexandria, VA

***LABORATORY TESTING THE GRAND ALLIANCE ADVANCED TELEVISION TRANSMISSION SYSTEM**
John Henderson
Hitachi America, Ltd.
Princeton, NJ

***DEVELOPING AND IMPLEMENTING THE TEST PLAN FOR FIELD TESTING THE ATV TRANSMISSION SYSTEM**
Jules Cohen
Jules Cohen and Associates. P.C.
Washington, DC

Edmund Williams
Public Broadcasting Company
Alexandria, VA

*Papers not available at the time of publication

HDTV FREQUENCY ALLOCATION ISSUES

Wednesday, March 23, 1994

Moderator:
Donald Jansky, Jansky, Barmat Telecommunications, Washington, DC

***ASSESSMENT OF CROSS POLARIZATION FOR INTERFERENCE REDUCTION IN BROADCAST SERVICES**
Victor Tawil
Association for Maximum Service Television
Washington, DC

SPECTRUM STUDIES FOR ADVANCED TELEVISION SERVICES IN THE U.S.
William Meintel
Datel Corporation
Chantilly, VA

VERTICAL POLARIZATION INTERFERENCE PATTERNS
Scott Lewis
PESA Micro Communications Inc.
Manchester, NH

*Papers not available at the time of publication

SPECTRUM STUDIES FOR ADVANCED TELEVISION SERVICE IN THE U.S.

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ABSTRACT

In 1990 the Federal Communications Commission made the decision that the implementation of an advanced television system (ATV) in the United States would to be within the existing television bands and that ATV channels would have the same 6 MHz bandwidth as the existing NTSC channels. Unlike color television, it was not feasible to design an ATV system that allows for transmission of NTSC and ATV on the same channel. In view of this, it became necessary to find an additional channel for each existing NTSC station so they will be able to begin transmission of ATV while continuing to provide NTSC service.

INTRODUCTION

The first efforts to determine the feasibility of developing a plan to implement ATV were carried out by Robert Eckert at the FCC. Soon thereafter a group of broadcast organizations coordinated by Maximum Service Television, Inc. undertook the funding of a project to develop computer modeling to asses the availability of spectrum for ATV.

The initial work of this project was an expansion of the FCC efforts whereby channel assignments were made on the basis of desired separation distance between stations. Improvements to the FCC's Assignment Model were made to allow the study of various separation criteria as well as the consideration of adjacent and UHF Taboo channels, plus a special criteria for co-located stations. Although this method created plans and allowed the determination of the number of existing stations that could be accommodated with a second channel, it did not provide any information concerning expected coverage and interference.

Because a nationwide plan for the implementation of ATV will have such a profound effect on the broadcast industry well into the next century, it is imperative that its design employ the best possible methods. Therefore efforts were undertaken to augment the Assignment Model software to address the issues of coverage and interference.

MODEL DEVELOPMENT

To determine the spectrum required for implementation of an ATV system, two computer models have been developed and integrated. The Assignment Model generates spectrum efficient channel assignment plans based on minimum separation distances. The Coverage and Interference Model (CIM) evaluates performance of plans generated by the Assignment Model providing expected coverage and service lost due to interference based on a set of user supplied system parameters. In addition, the CIM also has the capability to perform channel substitutions when its analysis indicates that such a change would yield an improved plan.

ASSIGNMENT MODEL

The Assignment Model uses minimum separation distances to determine the number of existing stations that can be accommodated with an additional ATV channel under different co-channel, adjacent channel and taboo channel distance separation criteria. Specifically, the model uses a heuristic approach to determine the "best" ATV accommodation statistics for a given set of input conditions. This is accomplished by first ordering the existing NTSC stations for a given area according to the apparent difficulty of finding a channel for them and then using a mathematical optimization method to find the largest number of stations that can be accommodated within that area. The output of the model is an assignment table that pairs existing NTSC stations with specific ATV channel assignments.

Two basic modules are used within the Assignment Model. The first module, the constraint generator, determines for each existing NTSC station all possible ATV channels that could be assigned to that station based on the user supplied separation criteria. The second module, the optimizer/evaluator, uses a number of mathematical algorithms to iteratively search for the lowest number of ATV channels that could be assigned to accommodate the entire set of NTSC stations.

Currently two techniques are being used to investigate the ATV assignment problem. The first technique employs an algorithm developed by Frank Box [1]. This technique is a heuristic assignment approach that operates on the principle that stations should be assigned channels in descending order of assignment difficulty. Through an iterative process, stations that repeatedly fail to be assigned channels, rise toward the top of the list of requirements for channels and are accommodated first. The successive reordering of stations by order of difficulty tends to improve accommodation statistics of a resulting solution or plan.

The second technique, developed by Robert Eckert at the FCC, uses a general algorithm, known as simulated annealing[2]. This method attempts to solve the assignment problem by starting with a random solution and repeatedly proposing small changes to the solution then replacing the solution each time an improved accommodation is obtained.

Both of these techniques have been used to investigate the assignment question and Eckert has indicated that the simulated annealing approach yields somewhat better results[3].

COVERAGE AND INTERFERENCE MODEL

Originally, software was to be developed that would provide coverage and interference area determinations for a single station with an option to graphically display the results. Computations were to be made using the FCC R-6602 propagation curves and user specified station data. From this basic concept, a highly sophisticated computer model has evolved. This enhanced model provides the capability to analyze nationwide ATV assignment plans, with flexibility to vary station and system parameters, select from any of three propagation models and furnish statistics on population density within coverage and interference areas for individual stations and for the entire country. The model has also been improved to employ actual station data from the FCC's Television Engineering and Directional Antenna data bases as well as the ability to use three second terrain data as needed for propagation modeling. In addition to the many computational features provided, software has also been developed that allows for optimization of ATV assignment plans.

Coverage and Interference Computation

The extent of the noise limited service contour of a station is determined using the FCC propagation curves. The F50/50 curves are used when computing NTSC service and at the user's option either F50/50 or F50/90 curves are used for ATV computations. The desired dBu value of the noise limited contour is a user specified value with different values permitted for each system (NTSC or ATV) and TV band. If the user has selected the FCC curves as the propagation model then it is assumed that field strength values will equal or exceed the noise limited value everywhere within the computed contour. However, if either of the optional propagation models (Longley-Rice [4] or TIREM [5]) is selected, then a further analysis is made to predict within the contour where noise limited service will exist. This is required since predicted signal levels using these alternative propagation models takes into consideration the terrain features along the path between the station and the point being evaluated, whereas use of the FCC propagation curves assumes an average height above the surrounding terrain in all directions.

Included in the computations are the stations's power and antenna height above the surrounding terrain, and at the user's option, the station's antenna pattern. Power and height can be either that found in the FCC data base or set to a fixed value for each band and system. When analyzing ATV assignments contained in an assignment plan, the user may use the fixed power and height option or have the station power computed so that the ATV noise limited service contour will match that of its paired NTSC station, using the paired station's height.

Interference calculations are made using the same station parameters and propagation model criteria used to determine noise limited service. If the FCC curves are selected as the propagation model then the F50/10 curves are used to compute interfering signal levels. If either the Longley-Rice or the TIREM model is selected then the user is requested to specify the desired location and time probabilities and in the case of Longley-Rice the level of confidence.

When determining the existence of interference, the user may also specify the channel relationships to be considered. Relationships that may be included are co-channel, upper and/or lower adjacent channels and the UHF "TABOO" channels. Different channel relationship considerations may be specified for NTSC to NTSC, NTSC to ATV, ATV to NTSC and ATV to ATV computations. Likewise the user may specify the acceptable ratio of desired-to-undesired signal level for each channel relationship that will be considered.

In order to evaluate the impact of interference from individual stations as well as overall interference, and to asses the amount of new interference caused to NTSC stations by the introduction of ATV service, a service matrix concept is employed. This concept divides the area within the service contour of a station into a matrix of cells. A determination is then made of the cells within the service area where interference is present. The total area within the contour receiving interference is then determined by adding up the area of the cells where interference was found to exist. By overlying the matrices that define the areas of interference caused by individual stations it is possible to determine the overall area of interference. Likewise by determining the difference between a matrix of the interference caused by other NTSC stations and a matrix of ATV only interference to an NTSC station it is possible to determine the amount of new interference that would be caused by the introduction of ATV stations. Similarly it is possible to show the extent of interference that will be received by an ATV station when NTSC stations are no longer operating.

The same matrix concept is used to determine noise limited service when the Longley-Rice or TIREM propagation models are used. The area of the cells where the computed signal level is below the noise limited threshold are subtracted from the total area within the noise limited contour to determine noise limited service area.

Determination of Population Densities

To be able to ascertain the population within service and interference areas, a data base of the 1990 census at the block level has been established. This data base contains the population and reference coordinates of each census block in the United States. After the service contour of a station has been established, the population within each matrix cell within the contour is computed by determining the total of the population of the census blocks that fall within the cell. Once this population matrix has been established, it is then used in the same manner as the area matrix to compute the population statistics.

Optimization software

In addition to providing the capability to determine coverage and interference statistics for an ATV assignment plan, the model also has two different options that attempt to optimize channel assignments by matching ATV interference areas with those of the paired NTSC station. The first option attempts to determine the "best" assignment scheme for stations located at a common site. A common site can be defined as a single location or locations that are within a specified distance of each other. The second option attempts to improve the plan by substituting other channels when such channels are determined to provide improved interference matching.

The first option analyzes all channels assigned at a "co-located" site for each of the stations at the site and creates a matrix of the mismatch between the existing NTSC interference area and ATV interference area. The reason to analyze each channel for each station is that this option assumes that the ATV assignment service contours are to duplicate those of the paired NTSC stations; therefore, the coverage and interference of each ATV station will depend on that of the paired NTSC facility. After all the stations and channels have been examined, the matrix is then analyzed. The station with the poorest overall interference area match (including all channels) is assigned the channel that had the best match. That station and channel are then removed from the matrix and the process is repeated until all stations have been assigned a channel. This system attempts to insure that matching is evenly distributed among the stations at the common site.

The second optimization option analyzes other available channels in an attempt to find a channel that either does not receive any interference or whose interference area better matches that of its paired NTSC station. When an ATV assignment plan is created based on specified separation distances, many of the locations may still have other channels that could have been assigned. This option first analyzes the planned ATV channel assignment to determine the amount of interference it receives from other ATV stations and existing NTSC stations. If no interference is received then no further analysis of the ATV assignment is made. However, if interference is received, the area where the interference occurs is compared to the interference area of the paired NTSC station. If the interference areas match exactly (no interference is caused to the ATV station outside the area where the paired NTSC station receives interference) no further analysis is performed, otherwise, each of the other available channels is analyzed searching for either a channel that receives no interference or one where the interference area better matches that of the paired NTSC station. If an improvement is found, that channel is substituted for the channel originally assigned. The list of additional channels that are available at other sites not yet analyzed are then checked to determine if this assignment will conflict with them. If a conflict is found the list for the other sites is modified to eliminate the conflict. Conflicts are determined on the basis of the spacing requirements that were used to create the assignment plan. It is noted that before the list of available channels is analyzed by the model, the channels are ranked by distance to the nearest co-channel NTSC station or ATV assignment and those with the greatest distance separation are analyzed first.

Special options

In addition to the originally planned model options discussed above, it has been necessary to create several others to study the effect of certain planning concepts not originally anticipated. Several of the major options included for these special studies are discussed below.

When attempting to optimize an assignment plan by substituting other channels an option has been added that will eliminate from consideration those channels that are within a specified distance range of an adjacent channel NTSC station.

Another special option allows for computation, when using the FCC propagation model, of undesired signals from stations that are less than 10 km from the desired station to be made using the same propagation curve as the desired signal rather than the F50/10 curve. This option was developed since it is expected that the signals from stations at a common site will propagate in a similar manner.

A further option computes undesired signals of stations on adjacent channels to desired ATV stations using F50/50 curves if the site to site distance is greater than or equal to a specified distance and F50/90 curves if the distance is less the specified distance. Computation of undesired signals on channels adjacent to a desired NTSC station are made using F50/50 curves if the site-to-site distance is less than the specified distance and F50/10 curves if the distance is equal to or greater than the specified distance.

And still another option caps the distance to the ATV service contour at specified distance if the noise limited contour of the paired NTSC station is greater than the specified distance. This option is used to perform evaluations of a plan where ATV service would not be provided beyond a specified distance.

When analyzing an ATV assignment plan using the equivalent service area option (duplicating NTSC service), an option has been provided to iteratively reduce (if necessary) the power of ATV assignments until the new interference caused to individual NTSC stations is no more than a certain percentage of the NTSC station's service area. This option allows for a determinations of the ATV power reductions that would be needed to minimize new NTSC interference.

Form of results

As shown in Figure 1, the results of the coverage and interference analysis include a considerable amount of information. The city, state, call sign, channel, power, and antenna height above average terrain is listed for both the desired station and any stations that cause interference to the desired station.

```
Interference analysis of NTSC station
WXXA  ALBANY NY                  Channel 23
ERP     3020.000 kw   34.80 dBk  HAAT:  366.0 meters

Service area equals   19054.0 square km
Service contour:  64.0 dBu  84.8 km

Population within service area  1310494

    Interference received from NTSC station
    WWLP  SPRINGFIELD MA              Channel 22
    ERP     3390.000 kw   35.30 dBk  HAAT:  268.0 meters

    Site to site azimuth :  118.4 degrees
    Site to site distance:   122.7 km

    Interference caused to:    618.8 square km of service area
    Using protection ratio of:   -6.0 dB

    Interference occurs between  83.1 and 147.5 degrees
    Width of interference area along the site to site azimuth:   10.5 km

    Interference received from ATV station
    WHSH  MARLBOROUGH MA              Channel 23
    ERP     250.67 kw   23.99 dBk  HAAT:  326.0 meters

    Site to site azimuth :   96.3 degrees
    Site to site distance:   208.1 km

    Interference caused to:   2340.6 square km of service area
    Using protection ratio of:   34.0 dB

    Interference occurs between  19.4 and 172.9 degrees
    Width of interference area along the site to site azimuth:   18.2 km
NTSC baseline interference:    618.8 square km
Percent of service area where interference is present:   3.2%

NTSC baseline service population  1238081

Portion of service area receiving interference:   2340.6 square km
Percent of service area where interference is present:  12.3%

Interference limited population  1154842

New IX to NTSC:   1721.7 square km
Percent of service area where interference is present:   9.0%

New Interference limited population    83239
```

Figure 1

In addition, the distance to the desired station's noise limited contour (maximum distance where a directional antenna is employed) and the area within the contour is given along with the population within the contour. If the Longley-Rice or Tirem propagation model was used then the noise limited service area and population is also provided. If any interference was found, the area of service and population lost to interference is given both for the individual interfering stations and the cumulative effect of all interfering stations. Separate statistics are also given for new interference caused to NTSC stations and for ATV only interference to ATV stations. Also provided for each interfering station is the arc of the desired station over which the interference occurs, the site to site azimuth and distance as well as the width of the interference along that azimuth.

When a nationwide analysis is performed a summarized version of the above, as shown in Figure 2, can also be provided. This summary provides a one line listing for each station. The listing identifies the station by location, call sign, channel, power and height and provides the cumulative totals for service and interference areas. A summary of the overall statistics on nationwide service and interference is can also be provided.

The user also has to option to display the results of an analysis graphically. Although this option is most often used to show the coverage and interference of individual stations, it is possible to display all analyzed stations within an area defined by the user. These graphical presentations use different colors to portray the various pieces of information. Service is shown in one color whereas interference areas are displayed in different colors depending on the channel relationship between the desired and undesired signals. The user also is given the option of displaying a geographical coordinate grid and/or a map of the area. All plots are accompanied by a legend and labels that provides a distance scale, the geographic coordinates of the four corners of the area, the desired stations' city, state and call sign, the date the plot was produced and up to three lines of user supplied description. Figure 3 is a graphical representation of the results of the coverage and interference analysis whose details are given in Figure 1, and Figure 4 shows projected loss of coverage when terrain features are considered by the Longley-Rice propagation model.

PAGE 2

CALL CITY - STATE	CH	PWR (KW)	HAAT (M)	SERVICE AREA (SQ KM)	POPULATION	DISTANCE TO CONTOUR (KM)	IX AREA (SQ KM)	% of SERVICE AREA	POPULATION LOST	ATV IX AREA (SQ KM)	% of SERVICE AREA	POPULATION LOST	ATV - NTSC (SQ KM)	% MISMATCH	POPULATION	%IMPROVE in SVC 1	%IMPROVE in SVC 2
WALA MOBILE AL	8A	12.96	381.0	33877.7	1020222	102.2	0.0	0.0	0	0.0	0.0	0	-408.7	0.0	0	1.3	1.2
WPMI MOBILE AL	17A	348.83	521.0	33477.2	909906	101.7	0.0	0.0	0	0.0	0.0	0	-668.2	0.0	0	2.9	2.8
WMPV MOBILE AL	62A	251.29	436.0	16057.1	819432	74.9	0.0	0.0	0	0.0	0.0	0	-33.1	0.0	0	0.2	0.2
WEIQ MOBILE AL	55A	36.12	183.0	11998.5	577191	51.8	0.0	0.0	0	0.0	0.0	0	-451.4	0.0	0	3.9	3.8
WSFA MONTGOMERY AL	28A	1651.40	610.0	45465.4	936231	120.3	0.0	0.0	0	0.0	0.0	0	-1344.5	0.0	0	3.1	3.0
WCOV MONTGOMERY AL	14A	15.79	224.0	11613.3	356265	40.8	0.0	0.0	0	0.0	0.0	0	-532.6	0.0	0	4.4	4.4
WAIQ MONTGOMERY AL	65A	43.10	183.0	12469.0	370913	43.0	0.0	0.0	0	0.0	0.0	0	0.0	0.0	0	0.0	0.0
WHOA MONTGOMERY AL	47A	304.43	545.0	28174.5	529128	102.5	294.1	1.0	5206	294.1	1.0	5206	160.5	1.0	5206	-0.6	-0.6
WMCF MONTGOMERY AL	49A	16.65	308.0	13609.5	398019	66.3	0.0	0.0	0	0.0	0.0	0	0.0	0.0	0	0.0	0.0
WCIQ MOUNT CHEAHA AL	52A	1451 60	610.0	45465.4	2231126	120.3	415.3	0.9	103652	415.2	0.9	103652	-1090.0	0.9	103652	7.4	6.8
WSWS OPELIKA AL	15A	21 33	207.0	11728.2	488433	61.1	0.0	0.0	0	0.0	0.0	0	-190.0	0.0	0	1.7	1.7
WDAU OZARK AL	57A	31.16	142.0	8787.1	229089	58.5	0.0	0.0	0	0.0	0.0	0	0.0	0.0	0	0.0	0.0
WAKA SELMA AL	31A	1308.39	515.0	40613.5	681584	113.7	1256.5	3.1	19293	1256.5	3.1	19293	-1266.9	0.1	113	9.1	8.0
WRJM TROY AL	36A	110.16	588.0	23510.9	536114	105.2	0.0	0.0	0	0.0	0.0	0	-1577.1	0.0	0	7.2	6.7
WDBB TUSCALOOSA AL	63A	238.33	674.0	29843.9	1269753	106.6	121.4	0.4	2502	121.4	0.4	2502	-2984.1	0.0	0	11.2	10.0
WCFT TUSCALOOSA AL	38A	31.93	165.0	11433.1	319765	60.3	0.0	0.0	0	0.0	0.0	0	-453.1	0.0	0	8.1	7.5
NEW TUSKEGEE AL	23A	340.50	610.0	35968.1	1181476	107.0	1093.3	3.0	73387	0.0	0.0	0	-2676.1	2.7	67419	8.3	7.4
KETG ARKADELPHIA AR	22A	1362.46	326.0	29066.6	386954	97.5	90.2	0.3	520	90.2	0.3	520	-3700.6	0.3	520	14.5	12.7
KTVE EL DORADO AR	34A	1446.43	605.0	45328.9	660008	120.0	0.0	0.0	0	0.0	0.0	0	-14365.4	0.0	0	45.6	31.3
KAFT FAYETTEVILLE AR	46A	1307.43	506.0	40186.0	750255	113.1	408.6	1.0	5262	408.6	1.0	5162	-4666.7	0.1	1269	13.3	11.6
KHOG FAYETTEVILLE AR	69A	19.32	271.0	13191.7	292258	64.8	0.0	0.0	0	0.0	0.0	0	0.0	0.0	0	0.0	0.0
KFSM FORT SMITH AR	48A	2653.97	384.0	38013.3	707045	110.0	0.0	0.0	0	0.0	0.0	0	-6099.7	0.0	0	14.4	15.5
KPOM FORT SMITH AR	18A	117.98	317.0	15147.4	428020	78.2	0.0	0.0	0	0.0	0.0	0	-521.0	0.0	0	3.6	2.4
KHBS FORT SMITH AR	36A	104.24	610.0	26167.6	336048	101.5	0.0	0.0	0	0.0	0.0	0	-237.7	0.0	0	0.5	0.9
KRZB HOT SPRINGS AR	58A	5.32	267.0	7887.8	168483	59.0	0.0	0.0	0	0.0	0.0	0	-766.7	0.0	0	11.2	10.1
KAIT JONESBORO AR	56A	1328.33	533.0	41847.6	713481	115.0	0.0	0.0	0	0.0	0.0	0	-5017.4	0.0	0	13.7	12.1
KTEJ JONESBORO AR	59A	43.88	311.0	18195.7	227612	71.0	0.0	0.0	0	0.0	0.0	0	-26.6	0.0	0	0.2	0.2
NEW JONESBORO AR	65A	119.99	142.0	16880.4	242699	80.7	0.0	0.0	0	0.0	0.0	0	0.0	0.0	0	0.0	0.0
KETS LITTLE ROCK AR	52A	3161.60	643.0	48205.1	1037861	124.0	0.0	0.0	0	0.0	0.0	0	-5969.0	0.0	0	14.1	12.4
KARK LITTLE ROCK AR	67A	2937.02	501.0	45760.3	1019465	120.7	0.0	0.0	0	0.0	0.0	0	-2319.6	0.0	0	5.3	5.1
KATV LITTLE ROCK AR	63A	1425.06	591.0	44562.9	992934	119.1	0.0	0.0	0	0.0	0.0	0	-665.0	0.0	0	1.9	1.9
KTHV LITTLE ROCK AR	50A	1310.12	521.0	40899.8	977125	116.1	0.0	0.0	0	0.0	0.0	0	-4880.7	0.0	0	12.9	11.4
KLRT LITTLE ROCK AR	40A	242.63	539.0	27390.2	447165	103.1	230.1	0.0	1116	230.1	0.8	1116	230.1	0.8	1116	-0.8	-0.8
KVUT LITTLE ROCK AR	43A	223.38	156.0	14957.1	613964	69.0	0.0	0.0	0	0.0	0.0	0	0.0	0.0	0	0.0	0.0
KENV MOUNTAIN VIEW AR	39A	2578.12	424.0	40470.8	609218	113.5	67.5	0.2	35290	0.0	0.0	0	-10444.1	0.0	0	36.9	26.9
KLEP NEWARK AR	38A	0.17	162.0	3631.7	52836	34.0	0.0	0.0	0	0.0	0.0	0	-97.2	0.0	0	2.0	2.7
KVTN PINE BLUFF AR	31A	189.33	182.0	11216.2	576604	70.2	0.0	0.0	0	0.0	0.0	0	-95.2	0.0	0	0.9	0.4
KASN PINE BLUFF AR	46A	336 88	593.0	36255.4	809522	105.8	0.0	0.0	0	0.0	0.0	0	-1253.7	0.0	0	5.0	4.4
KFAA ROGERS AR	64A	1 15	145.0	5738.0	230677	42.7	0.0	0.0	0	0.0	0.0	0	0.0	0.0	0	0.0	0.0
KNAZ FLAGSTAFF AZ	69A	2828.10	488.0	64712.7	301082	119.3	0.0	0.0	0	0.0	0.0	0	-1843.3	0.0	0	3.6	3.5

Figure 2 Sample of summary output from the CIM

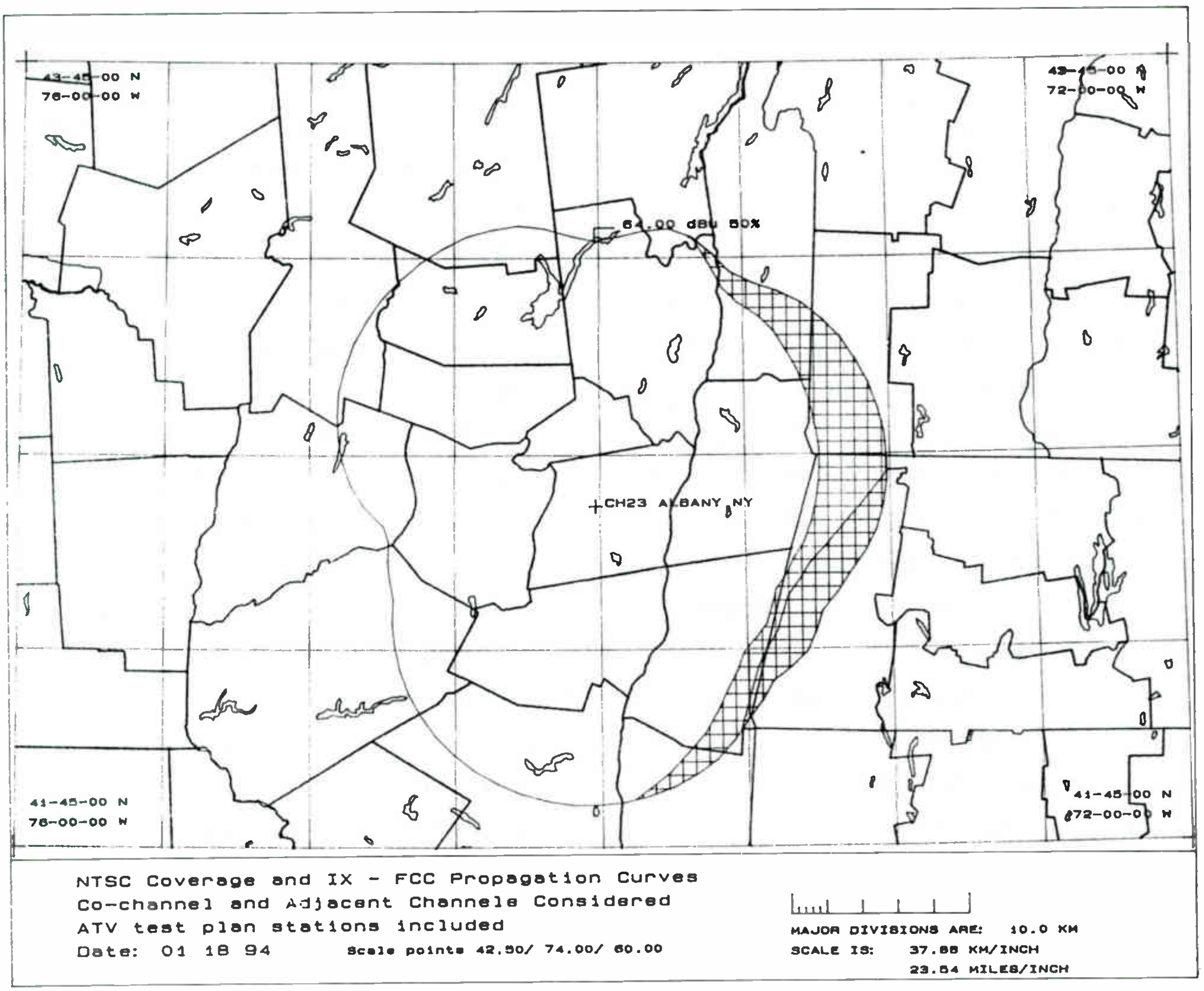

Figure 3

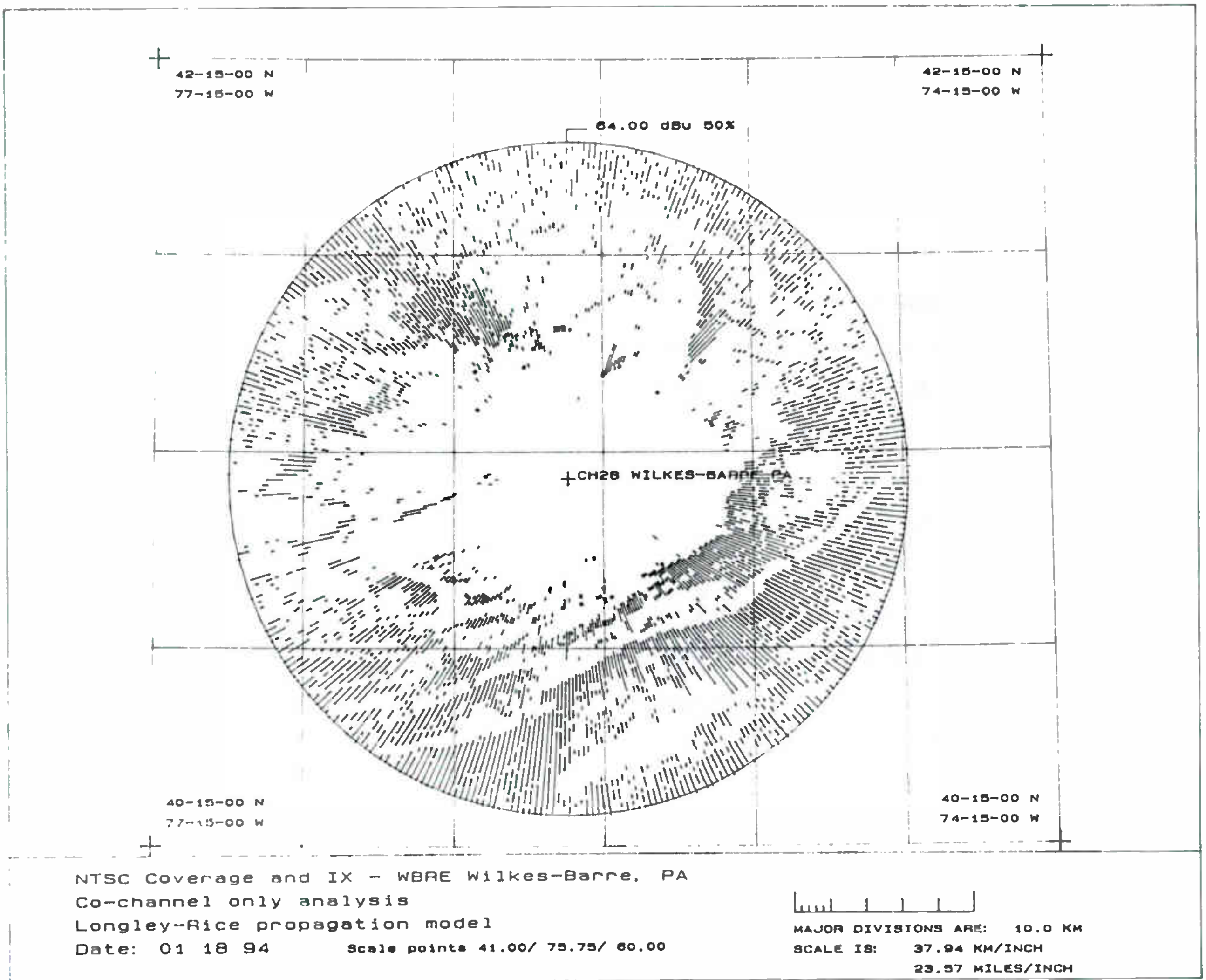

Figure 4

The CIM Model in use

Over the past year and a half this model has been used extensively to evaluate the coverage and interference characteristics of the various proposed systems under consideration by the FCC Advisory Committee on ATV. It has also been used to analyze the ATV channel assignment plan put forth by the FCC as well as being used to develop and analyze several alternative plans. More recently the model was used to assist the Grand Alliance in their efforts to achieve consensus on a single method for the transmission of ATV. Over the next several months the model is expected to play a major role in the development of the nation's ATV channel assignment plan.

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VERTICAL POLARIZATION INTERFERENCE PATTERNS

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I. Introduction

In developing the parameters for digital TV, consideration has been given to using vertical polarization to reduce the interference from HDTV transmission into existing NTSC receivers.

Since all FM and TV broadcast in this country use horizontal polarization, little information is available on "how" vertically polarized waves over lossy ground combine at the receiving antenna.

The analysis here considers only a direct wave combining with a single reflected wave for the two channels (53 and 6) to be used in over the air testing of the Grand Alliance system. This paper does not consider surface variability factors which are included in the FCC 50/50 curves.

The interference pattern occurs when the reflected signal combines in and out of phase with the direct signal as the receiving antenna is raised to the 30 Ft. height required by the FCC in making field strength measurements.

The NTSC signal is essentially a single frequency whereas the HDTV is a uniform signal over a 6 MHz band.

The nulls shown in the graphs of signal variation as a function of distance are due to the interference patterns and not due to nulls in the antenna patterns, since an isotropic pattern was used as the source in this theoretical model.

Field tests are a vital part of the Advanced Television (ATTC, SS/WP-2) evaluation process. The test on the final system is to take place in Charlotte, NC from an antenna on a 1300 Ft. tower at Channels 6 and 53.

The purpose of this paper is to consider transmission of vertically polarized waves between elevated antennas over lossy plane earth. Although the discussion is limited to Ch. 6 and Ch. 53, the results developed here are generally applicable to all high and low TV bands.

The first analytic treatment of the radiation field of a dipole source above a lossy plane earth was given by Sommerfeld {1} in 1909. Since then, a large number of investigators (e.g. {2}-{4}) have refined Sommerfeld's early work, in particular with respect to the rigorous analytic treatment of the various integrals arising in the solution. For general engineering purposes, the most useful analytic results were obtained by Norton ({5}-{7}) wherein it was demonstrated that the total field due to a general radiator near a lossy half-space is constructed as the sum of direct, reflected, and surface wave contributions. As shown in Fig. 2 the direct wave does not "see" the earth; the reflected wave arrives at the reflection point from an angle ψ with respect to the plane of the interface and leaves at the same angle, having suffered amplitude diminution and phase alteration; and the surface wave propagates (and attenuates) along the interface and attenuates away from the interface. For sufficiently large receiving antenna heights, the surface wave may be ignored.

For standard TV and FM antennas, the ground-reflected signal gives a fundamental contribution at large antenna heights. Horizontally and vertically polarized signals have dissimilar reflection characteristics at the dielectric interfaces.

An important characteristic of interfering waves is that when path lengths are not greatly different (in an absolute sense) the signal may vary from nearly complete cancellation to 6 dB above the free-space signal. For transmitting antenna heights on the order of 1000 ft. and separations of beyond 5 mi. variations repeat for increases in receiving antenna height.

A small hill may then account for a significant increase or decrease in received signal. While it is generally true that horizontally polarized waves are slightly stronger than vertically polarized waves, in the region of the interference pattern null the vertically polarized wave can be stronger by many dB.

In Section II, the characteristics of transmission between elevated antennas are discussed for linear polarization. In Section III, curves are presented for determining the distance between receiving antenna and reflection point as a function of antenna height and distance. In Section IV, numerous examples are given to illustrate the effects of antenna height, path length, and soil constants on signal strength at elevated receiving points.

II. Transmission Between Elevated Antennas

For short transmission paths, it is sufficient to consider the earth as a plane interface with the heights of the antennas determined with respect to the average terrain, as shown schematically in Fig. 1. The interface is assumed to separate two semi-infinite half-spaces. The lower half-space is a lossy dielectric, characteristic of typical ground, with relative dielectric constant ε and bulk conductivity σ (in millimhos/meter). The upper half-space is vacuum. And while such a discontinuity results in a pole in the general analysis (thereby predicting a surface wave), it is assumed that the antenna heights are sufficiently great such that the surface wave may be neglected.

The general problem of interference between the direct and ground-reflected waves is then reduced to the relatively simple problem of determining the phase and amplitude relation between the two signals. Fig. 2 represents the geometry of the specialized problem. Intuitively, if the interface is replaced by a perfect conductor and the path lengths D_1 and D_2 are not greatly different in magnitude, it is expected that the signal at the receiving antenna will periodically undergo variations from no signal to about 6 dB above the signal that would be received in an infinite space as the height of either antenna is continuously increased or decreased.

Since the earth is not a perfect conductor (or even a fair conductor), the reflected signal must experience both phase and amplitude degradation at the interface. The effects of this degradation will alter the interference pattern with respect to amplitude (due primarily to phase alteration upon reflection). The departure from the ideal situation will then be great or minor depending upon how closely the earth approximates a perfect conductor for the given geometry. However, as the magnitude of the reflection coefficient is comparable to 1, there will result significant signal gain or loss with respect to free-space propagation as the antenna heights are varied.

The degree of departure from the ideal case above is also dependent on the polarization of the transmitted signal. Reflection at the boundary between half-spaces does not produce depolarization, and it can be expected then, as with the reflection at the interface between lossless dielectrics, that the horizontal and vertical polarization reflection coefficients will differ widely in both magnitude and phase.

For frequencies in the TV band and coil constraints typical of the United States, the magnitude of the reflection is approximately 1.0 for horizontal polarization and the phase of R_H is in the neighborhood of -180° for all angles of incidence. For vertical polarization however, the magnitude and phase of the reflection coefficient R_v can vary greatly (5).

Consider the situation depicted in Fig. 2. The total space wave seen by the omnidirectional receiving antenna due to the omnidirectional source is proportional to the sum of the direct wave and the reflected wave. Therefore, for either polarization the field strength behaves as:

$$|E|_p = C\left|\frac{e^{-jkD_1}}{D_1} + R\,\frac{e^{-jkD_2}}{D_2}\right|$$

$$|E|_p = \frac{C}{D_1}\left|1 + R\,\frac{e^{-jk(D2-D1)}}{D_2/D_1}\right| \qquad (1)$$

where R is either R_H or R_V, C is a proportionality constant, and k is the free-space wave number $2\pi/\lambda$. For $D >> (D_2 - D_1)$, the exponent in (1) may be replaced by $-jh_1h_2/D$. As (D_2-D_1) approaches an odd multiple of $\lambda/2$

$$|E|_p = \frac{C}{D_1}\left|1 - R(D_1/D_2)\right| \qquad (2)$$

where

D_2 = total path length reflected wave;
D_1 = total path length direct wave;

and, as (D_2-D_1) approaches an even multiple of $\lambda/2$

$$\left|E\right|_p = \frac{C}{D_1}\left|1 + R(D_1/D_2)\right| \qquad (3)$$

If d >> (D_2-D_1), it is possible to determine the approximate relative heights for which the interference maxima and minima will occur. Let ϕ be the phase of the reflection coefficient R. Then minima will occur for

$$(2n - 1)\,\pi = k(D_2 - D_1) - \phi$$

$$\approx \frac{2k\,h_1\,h_2 - \phi}{d}$$

$$h_2 \approx \frac{(2n - 1 + \phi / 180°)\,\lambda d}{4h_1}, \qquad (4)$$

$$n = 1, 2, 3 \ldots \infty$$

and maxima will occur for

$$h_2 \cong \frac{(2n + \phi / 180°)\,\lambda d}{4h_1}, \qquad (5)$$

$$n = 1, 2, 3, \ldots \infty$$

where d is the horizontal separation between the two antennas. In the region for which ϕ does not greatly differ from -180°, as is the case for horizontal polarization, the first maximum occurs at

$$h_2 \approx \frac{\lambda d}{4h_1} \text{ (for first maximum)} \qquad (4a)$$

and the second minimum (the first being at $h_2 = 0$) occurs for

$$h_2 \approx \frac{\lambda d}{2h_1} \text{ (for second minimum)} \qquad (5a)$$

It is of interest to note that for standard soil (ε = 15, δ = 15 mmhos/m) the phase ϕ of both horizontal and vertical reflection coefficients are within ± 10° of -180°, respectively, for small elevation angles (ψ < 10°). Since practical towers seldom exceed 2000 ft., the occurrence of maxima and minima for vertical polarization should coincide closely with that for horizontal polarization. However, when the point of reflection is a highly conducting surface, such as sea water, the phase of R_V is nearly 0°, while the phase of R_H is nearly -180°. Maxima for vertically polarized waves will then occur in the vicinity of minima for horizontally polarized waves, and conversely.

III. Effective Height of Receiving Antenna

The validity of this data depends on an understanding of the magnitude of the reflected signal. Throughout this paper we assume flat terrain with ground properties as specified. In many locations the signal will be scattered by local reflection such as home, buildings, etc., and the magnitude will be different from that calculated.

It is therefore necessary to have an estimate as to the distance to the point of reflection from the receiving antenna location. The distance from the receiving antenna to the point of reflecting is

$$L = \frac{D(5280)}{1 + h_1/h_2} \qquad (6)$$

where

L	distance feet, to point of reflection from receiving antennas;
D	distance miles, between transmitting and receiving antennas;
h_1/h_2	ratio of transmitter to receiver height.

Equation (6) is plotted in Fig. 3 for distances of 5, 15, and 30 mi. If there is no reflected signal, (e.g., it is completely scattered and there is a line of sight path from the transmitter to receiver) the received signal would be equal to the inverse field (free space loss) at the point of measurement regardless of height.

IV. Tabulated Results

The data shown in Fig. 4 is the signal level in mV/m and dBu as a function of height above ground.

This is the signal that would be received by a horizontal or vertical dipole as the antenna is

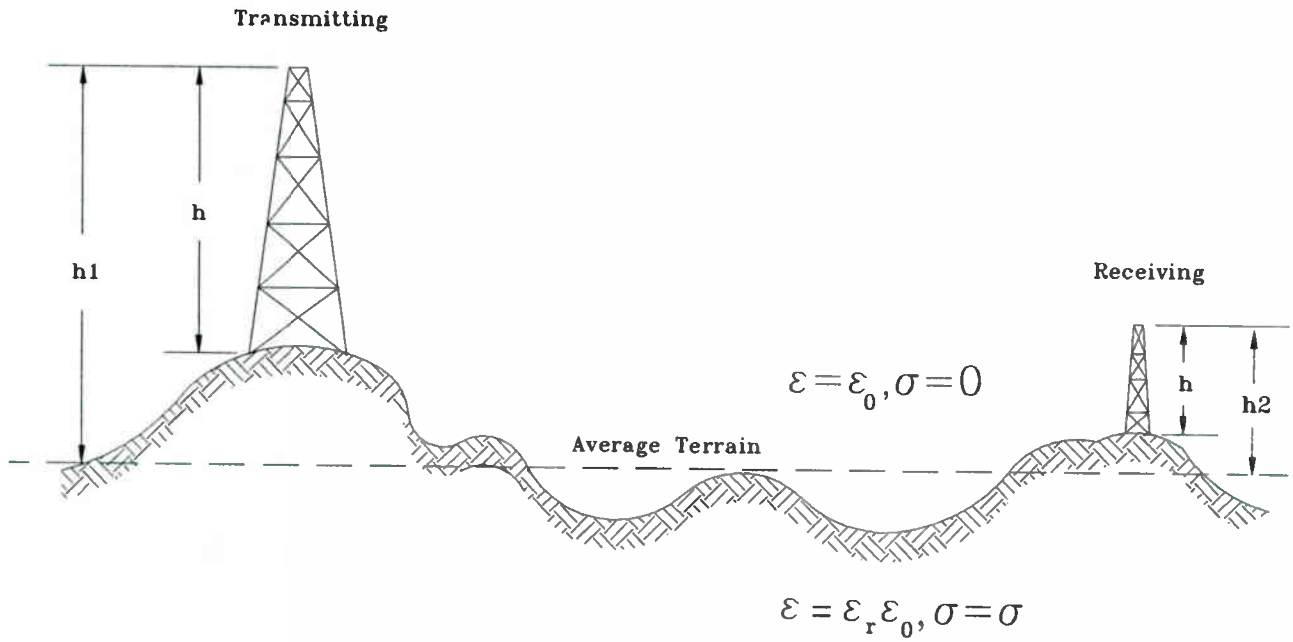

Figure 1. Antenna heights relative to terrain

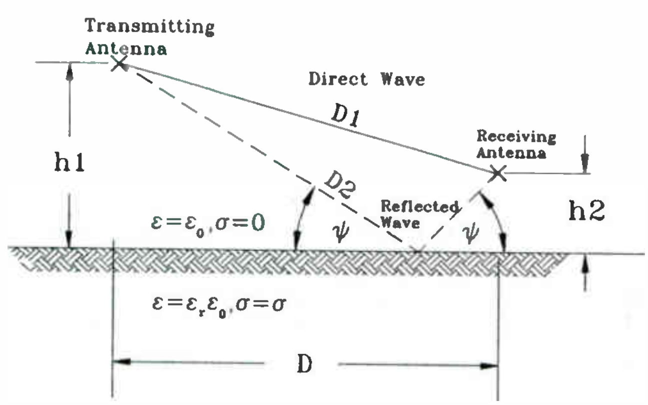

Figure 2. Geometry for direct and ground-reflected wave

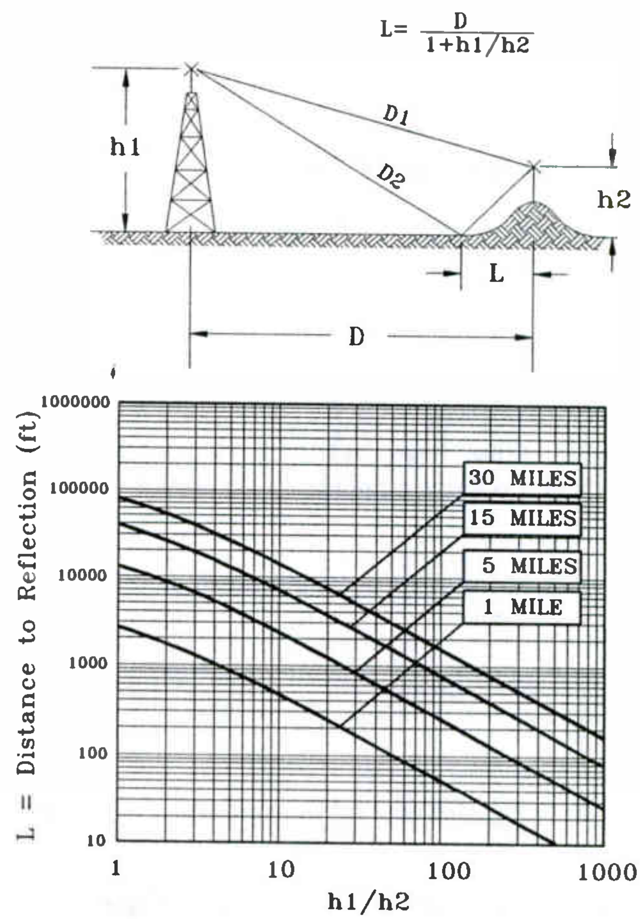

Figure 3. Distance to point of reflection vs. h1/h2

raised from ground zero to the height shown. The antenna is assumed to be an isotropic radiator with an input power of one kilowatt at a height of 1300 ft.

The data is for receiving locations of 5, 15, and 30 miles from the transmitting antenna for Channel 53 (705.25 MHz), the channel selected for ATV field tests. At the point of reflection, the ground properties are $\varepsilon = 15$, $\sigma = 15$ mmhos/m representing pastoral land.

The interference pattern is a cyclic function of the difference between the direct and reflected wave.

The solid line represents the free space loss (inverse field) at each distance. Also shown is the predicted signal from the FCC 50/50 curves.

The results show that for vertical polarization the signal is slightly less at the peak, but considerably greater at the nulls.

The data was calculated for Ch. 53 at f = 705.25 MHz (Fig. 4, 5, 6) and Ch. 6, f = 83.25 MHz (Fig. 7, 8, 9), the two channels to be tested and for the following ground properties:

Pastoral Land	$\varepsilon = 15, \sigma = 15$	mmho/m
Industrial land	$\varepsilon = 3, \sigma = 0.1$	mmho/m
Sea Water	$\varepsilon = 80, \sigma = 5.000$	mmho/m

The difference between the peak and null of the interference pattern relative to the FCC 50/50 curves can be ± 20 dB, except for vertical polarization over sea water where the reflection coefficient is very small so the resultant signal approaches the free space loss.

The signal variation at a receiver height of 30 ft. as a function distance from the transmitting antenna is shown in Fig. 10, 11, 12 for Ch. 53 and 13, 14, 15 for Ch. 6 .

On each curve the signal for both polarizations is shown as compared to the free space loss and the FCC 50/50 curves.

The "near in" nulls are a result of the interference pattern and not the antenna pattern since a unit isotropic source was used as the transmitting antenna.

Of interest was the variation over a 6 Mhz band as a function of height for Ch. 53 and Ch. 6 (Figures 16,17).

The effective reduction in signal is greater at a distance close to the tower, the reason being that for tall transmitter antenna locations we are already close to the peak of the interference pattern and no matter how high the transmitting antenna is increased beyond that point there can be no further increase in signal. The above data was calculated for Channels 53 and 6 with an antenna input power of 1kW. The data can be corrected for any power by

$$E = \sqrt{\frac{P_{ACTUAL}}{P_{1KW}}}$$

V. Conclusion

This paper shows that when making field intensity measurements from hills or high structures, it is necessary to consider the effective height as above the point of reflection and not the point of ground on which the receiving antenna is located. An interference pattern will be produced, the peak of which can be 6 dB higher than a straight inverse field calculation. It is also possible depending on distance and heights, for the signal to decrease with increasing receiver height.

References

{1} A Sommerfeld,"The propagation of waves in wireless telegraphy." Ann. Phys., vol. 28, p. 665, 1909.

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{5} K. A. Norton, "The propagation of radio waves over the surface of the earth and in the upper atmosphere-Part I, " Proc. IRE, vol. 24 pp. 1367-1387, Oct. 1936.

{6} ___,"The propagation of radio waves over the surface of the earth and in the upper atmosphere-Part II." Proc. IRE, vol. 25, pp. 1203-1236, Sept. 1937.

{7} ___,"The calculations of ground-wave field intensity over a finitely conducting spherical earth," Proc. IRE, vol. 29, pp. 623-639, Dec. 1941.

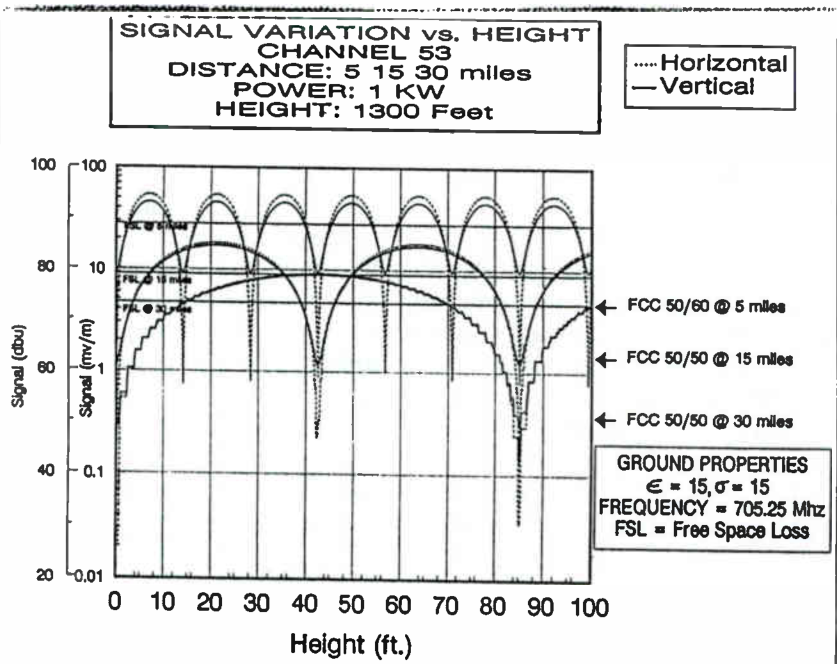


FIGURE 4

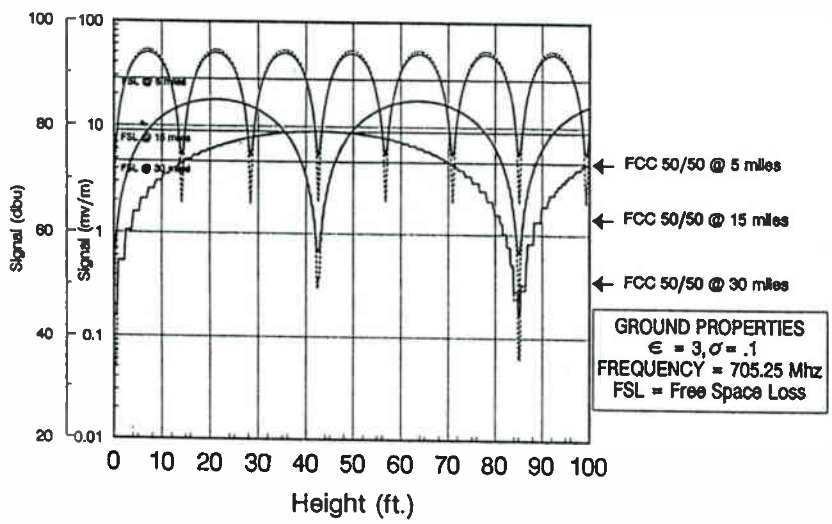
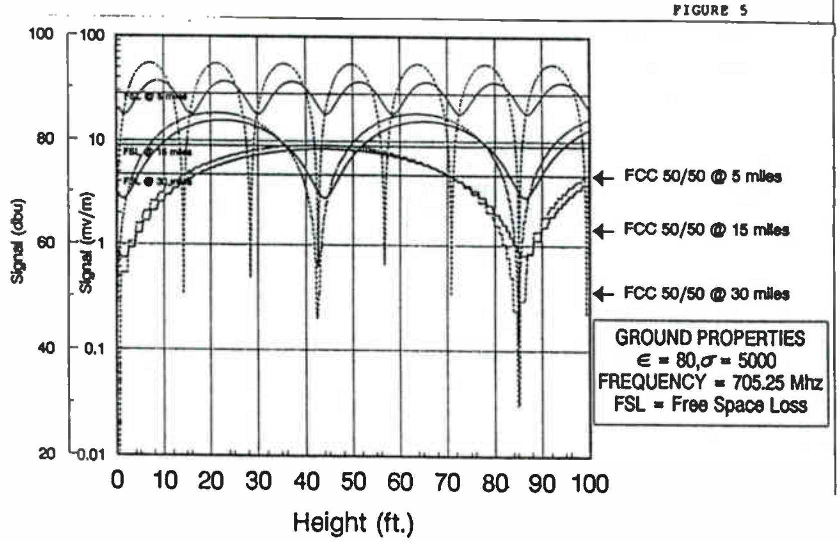


FIGURE 5



SIGNAL VARIATION vs. HEIGHT
CHANNEL 6
DISTANCE: 5 15 30 miles
POWER: 1 KW
HEIGHT: 1300 Feet

.....Horizontal
—Vertical

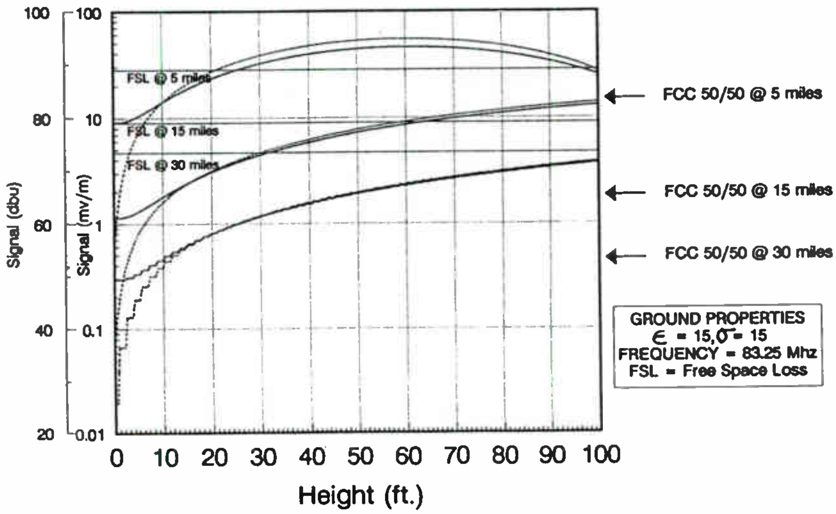


FIGURE 7A

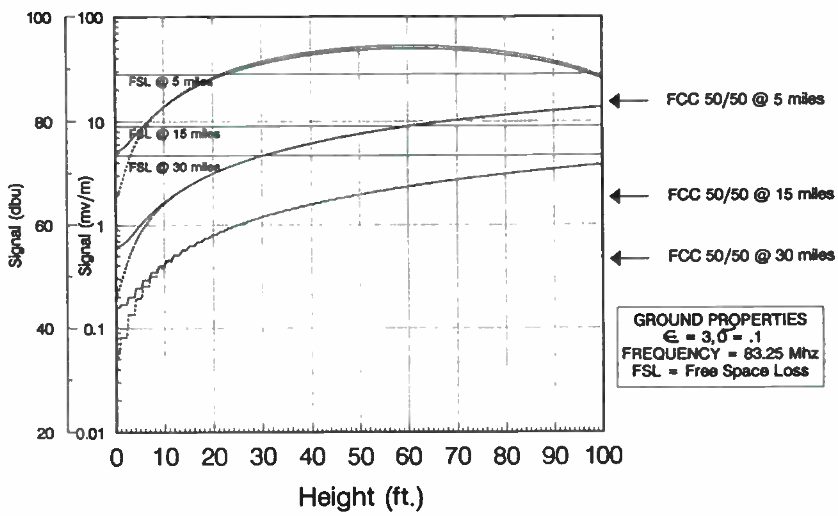


FIGURE 8A

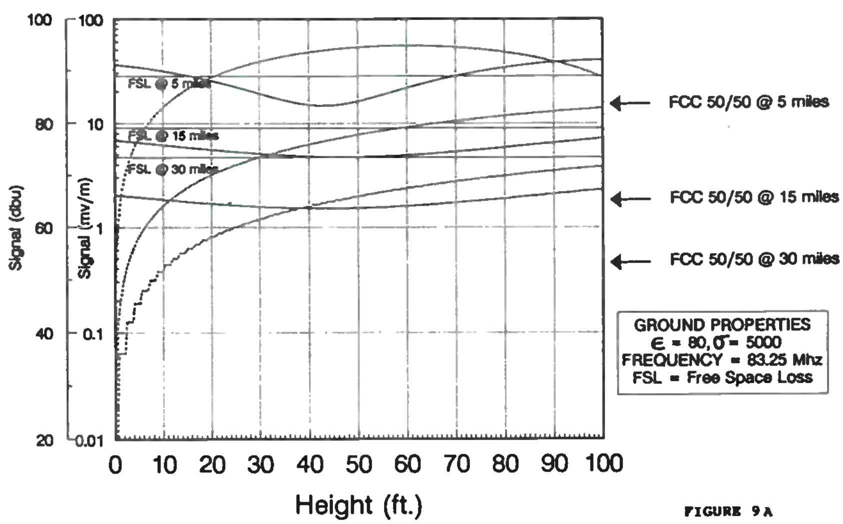


FIGURE 9A

SIGNAL VARIATION vs. HEIGHT
CHANNEL 6
DISTANCE: 5 15 30 miles
POWER: 1 KW
HEIGHT: 1300 Feet

·····Horizontal
—Vertical

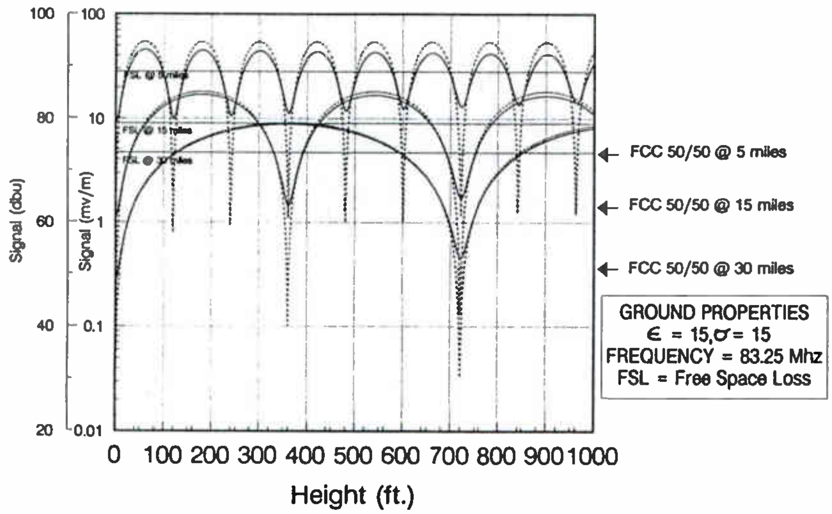


FIGURE 7B

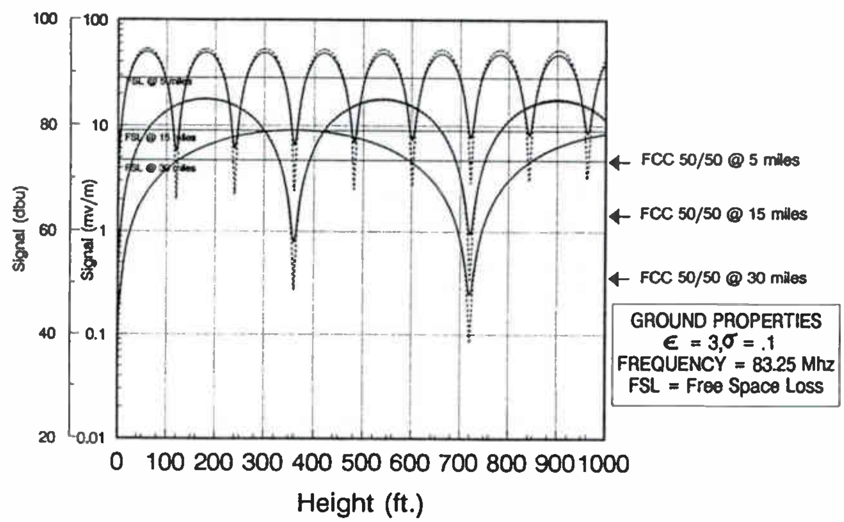


FIGURE 8 B

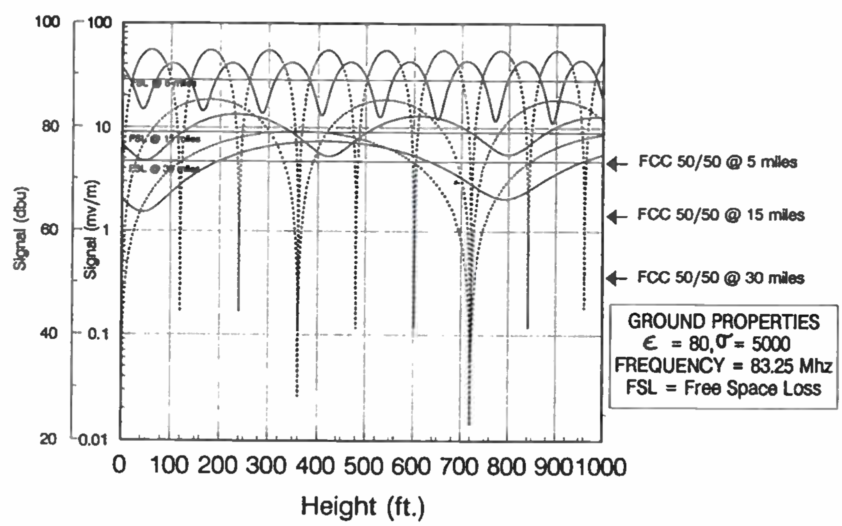


FIGURE 9B

SIGNAL VARIATION vs. DISTANCE
CHANNEL 53
HEIGHT: 1300 Feet
POWER: 1 KW
RECEIVER HEIGHT: 30 Feet

..... Horizontal
— Vertical

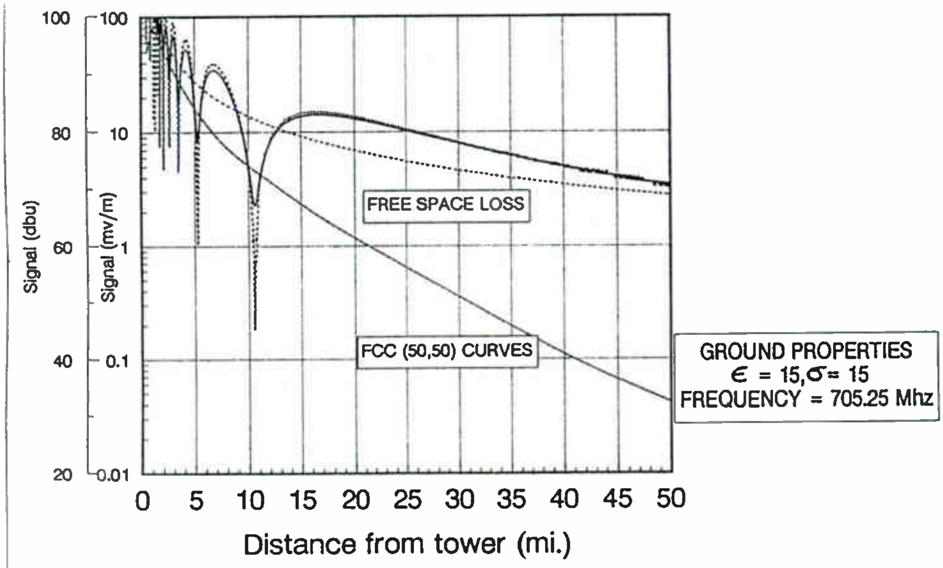


FIGURE 10

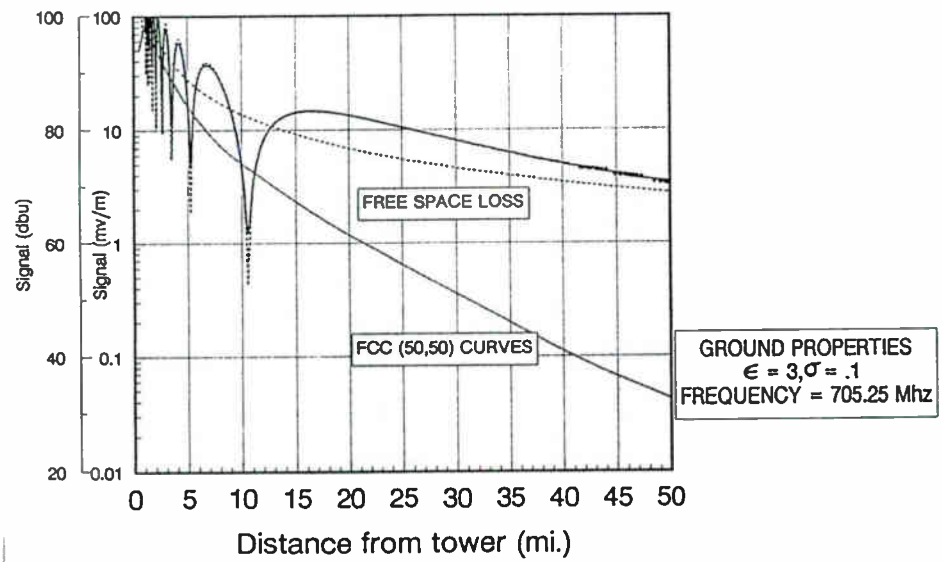


FIGURE 11

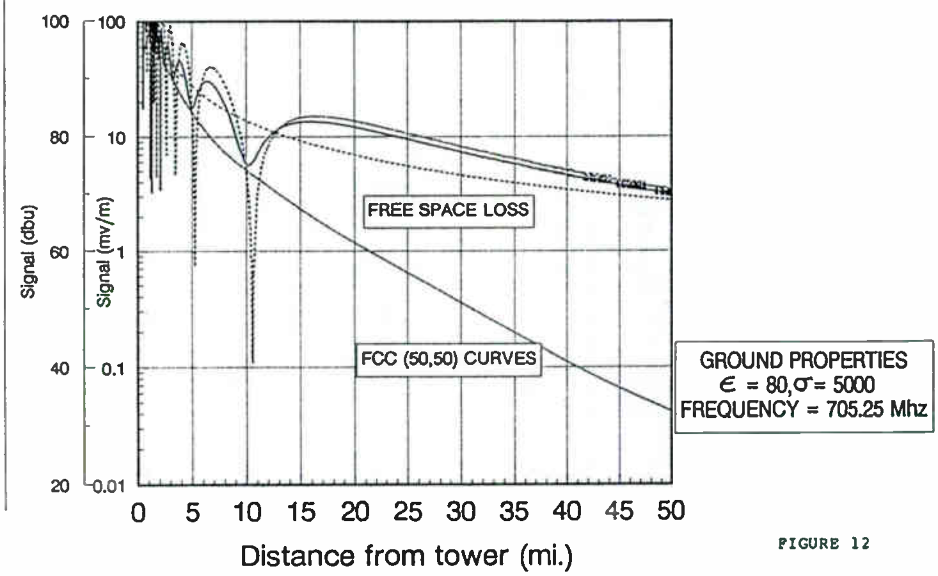


FIGURE 12

SIGNAL VARIATION vs. DISTANCE
CHANNEL 6
HEIGHT: 1300 Feet
POWER: 1 KW
RECEIVER HEIGHT: 30 Feet

..... Horizontal
— Vertical

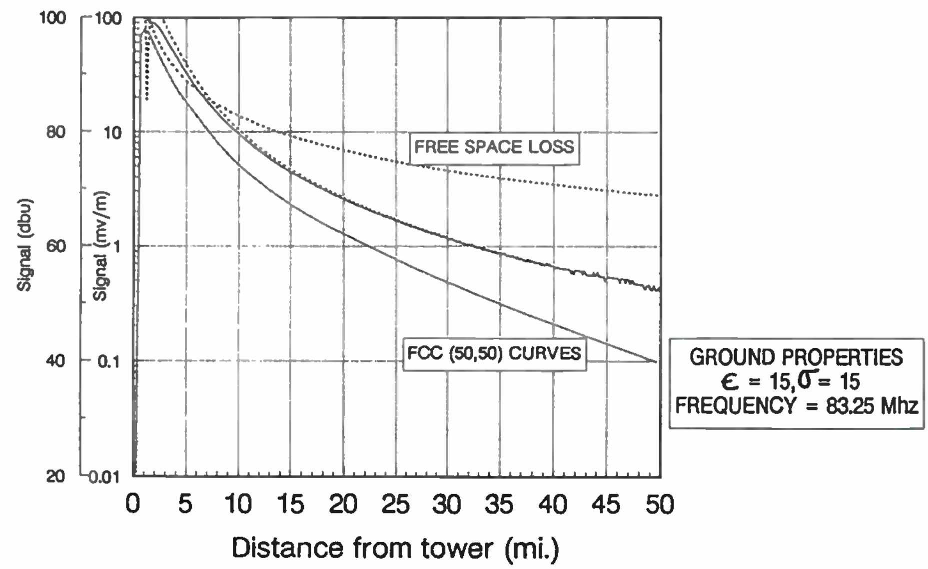


FIGURE 13

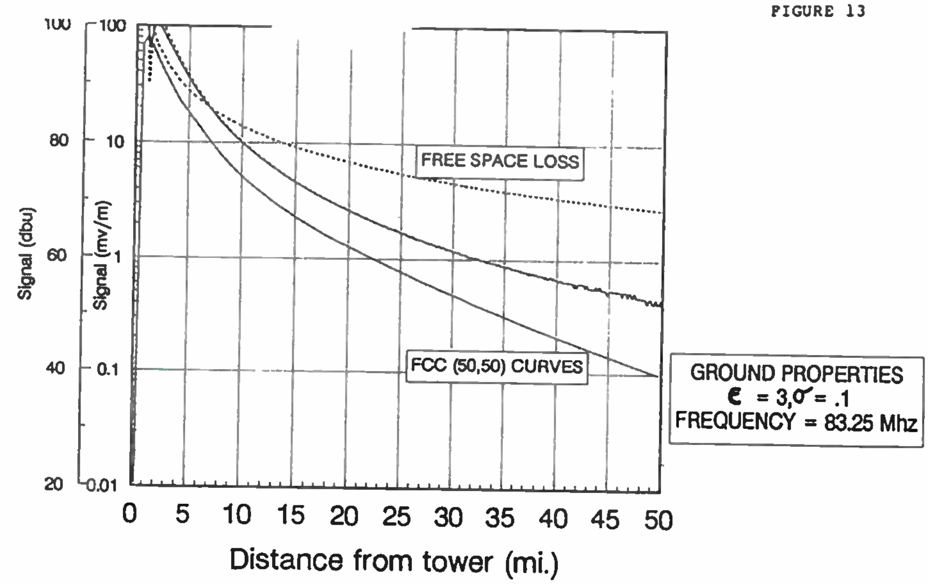


FIGURE 14

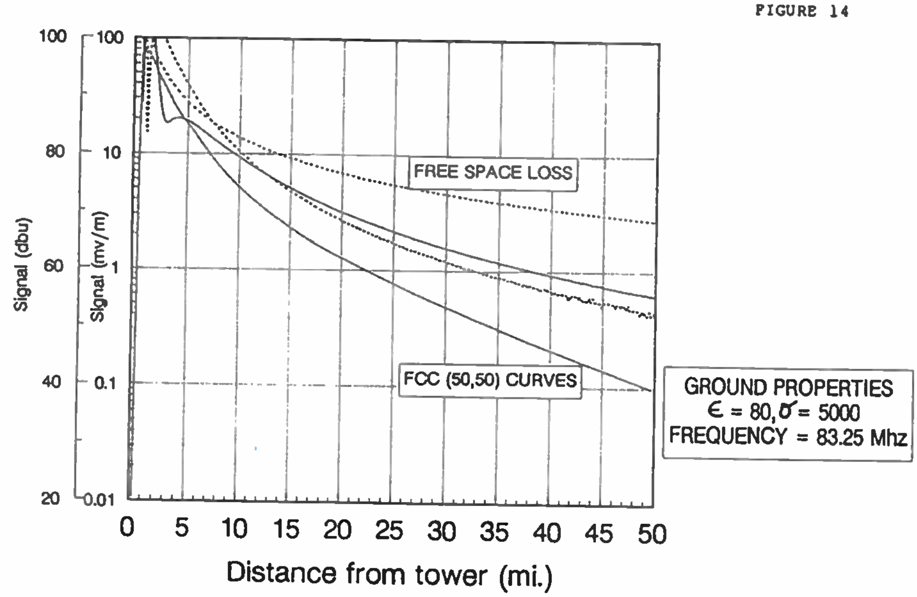


FIGURE 15

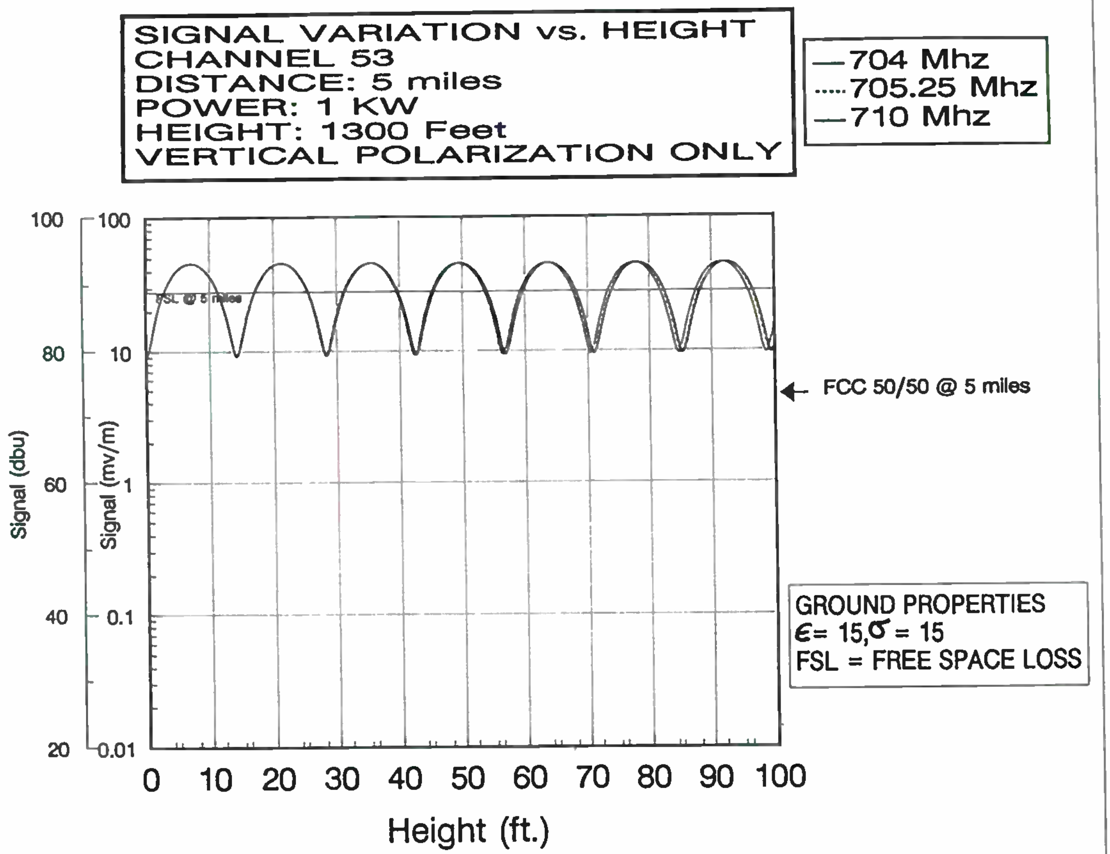


FIGURE 16

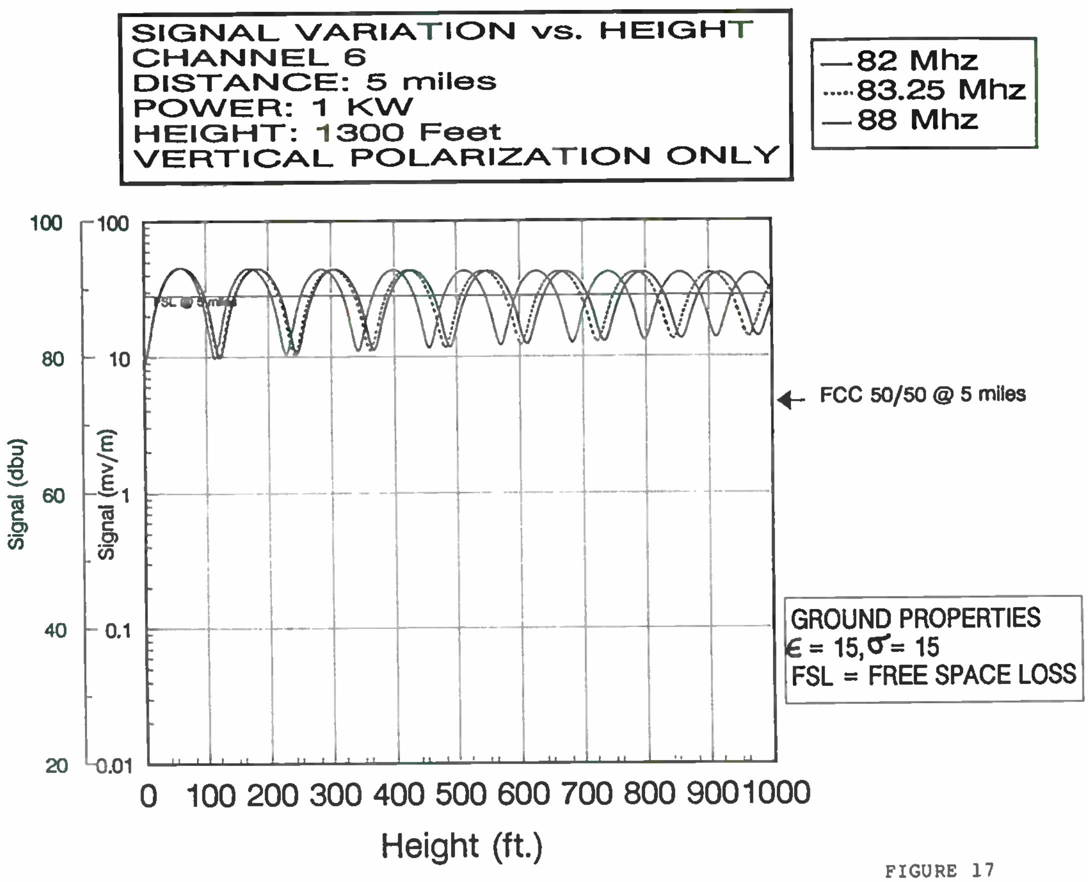


FIGURE 17

DAB—I

Wednesday, March 23, 1994

Moderator:
Ken Springer, NAB, Washington, DC

THE EIA DIGITAL AUDIO SUBCOMMITTEE VHF CHANNEL CHARACTERIZATION TEST
Robert Culver, P.E.
Lohnes and Culver Consulting Engineers
Laurel, MD

***NRSC/EIA DAB TEST REPORT**
Thomas Keller
Consultant
Springfield, VA

ASSESSING THE SUBJECTIVE QUALITY OF DAR SYSTEMS
Gerald Chouinard
Communications Research Centre
Ottawa, Ontario, Canada

THE EIA DIGITAL AUDIO SUBCOMMITTEE VHF CHANNEL CHARACTERIZATION TEST

Robert D. Culver, P.E.
Lohnes and Culver Consulting Engineers
Laurel, Maryland

ABSTRACT

The Electronic Industries Association (EIA) DAR Subcommittee is evaluating proposed Digital Audio Radio (DAR) systems for the United States. As a part of that process, the Subcommittee is conducting a "VHF Radio Channel Characterization Test". The results of the test will be applied directly to the laboratory testing of the DAR systems and will also find general use within the radio communications industry.

INTRODUCTION

The EIA DAR Subcommittee was organized in October 1991 for the purpose of documenting and testing all potential United States DAR systems in a uniform and impartial forum. The testing will take place in both a laboratory and the field. The VHF channel characterization test program is a recent addition to the EIA-DAR system tests for the purpose of quantifying VHF multipath propagation characteristics for the FM channel proponent laboratory tests.

The test system consists of a wide band pulse transmitter, a mobile data collection van with receiver and off-line data processing. The test signal consists of a series of pseudo-random pulses, transmitted at a pulse rate of 2 MHz, yielding a 4 MHz wide test signal centered in the television band at Channel 6, 85 MHz. Transmission is made with a circularly polarized antenna for reception on a specially designed and built dual-independent polarized antenna. Data collection is triggered at fixed intervals along a path, for this test every 1/10 wavelength. By choosing the path to cover a representative sample of an area, the channel characteristics of that type of area can be reported.

The data is analyzed to reveal the delay times and magnitudes of the transmitted reflected signal, relative to the direct signal. This data is used in simulating the multipath environment for laboratory testing. From that information the frequency domain channel characteristics are then calculated and presented. The data can be analyzed in a variety of formats for presentation to specific users relative to their particular needs. A sample of the analyzed data is presented in this report supported by a series of tables, charts, graphs and photographs.

The Need for "Characterization Tests"

Laboratory testing will employ a multi-channel multipath simulator, operating over the full range of multipath propagation delays and reflections expected in the real world. Unfortunately, the current literature does not clearly define the multipath characteristics in relative time delay and reflected signal level to be found at FM frequencies. As a result of the proponent's questions regarding the range of multipath testing, the EIA-DAR test program was expanded to conduct a channel characterization test. The Advanced Television (ATV) test being conducted by PBS in Charlotte, N.C. had an existing FCC authorization to conduct experimental ATV broadcast tests which presented an opportunity to experiment with the DAR Channel test system. Data collection was later conducted in Salt Lake City. This paper describes the progress of the

EIA-DAR Channel Test as conducted to date and the forthcoming data reduction and reporting.

THE CHANNEL TEST SYSTEM

The system used for the EIA DAR Channel test, much like that described by Herman[1] and Moriyama [2], was built by engineers at the DELCO Electronics Audio Advanced Development Laboratory at Kokomo, Indiana. It consists of a pseudo-random pulse generator driving a wide band (television) transmitter as shown in the system block diagram in Figure 1. A pseudo random pulse string, 255 pulses (bits or chips) long, is generated at the pulse repetition rate (chip rate) of 2MHz. The string of pulses is random in time over the 255 pulse string which is then repeated continuously, hence the pseudo-random nature of the pulses. The repeated 255 pulse string, called an "instance", is long enough to allow for calculation of over 40dB of level resolution but short enough to assure identification and capturing of one complete string in a relatively short time with a reasonable amount of data. The 2MHz pulse rate and the method of computer analysis used, allows resolution of signal reflection delay times as short as 250 nanoseconds with a finite probability of error, and as long as 127.5 microseconds delay time.

Transmission

The transmitter begins with a pulsegenerator providing a single-bit, shift register based, Pseudo-random Number (PN) sequence, with accuracy derived from a 25ppm crystal oscillator. The natural frequency side lobes of the sequence decay in level with frequency and they are subject to an additional attenuation of 20 dB by filtering in the signal generator to achieve 30 dB or more attenuation with respect to the main lobe. The PN sequence is 255 pulses long over which its timing is random. The sequence is continuously repeated, hence the pseudo-random nature of the signal. This pulse train phase modulates a high stability (1ppm) oscillator, resulting in the BPSK (suppressed carrier) signal at approximately 44 MHz. The actual frequency is adjustable over a moderate range, within the range acceptable to most television transmitter exciters. The transmitter exciter provides additional filtering of the sidelobe out-of-band components in a Surface Acoustic Wave (SAW) filter, and provides a constant amplitude signal to drive the transmitter power amplifier. The power amplifier is capable of a significant television peak of sync power but it is driven only to an output power of a few dB below the power compression point to maintain good amplitude linearity.

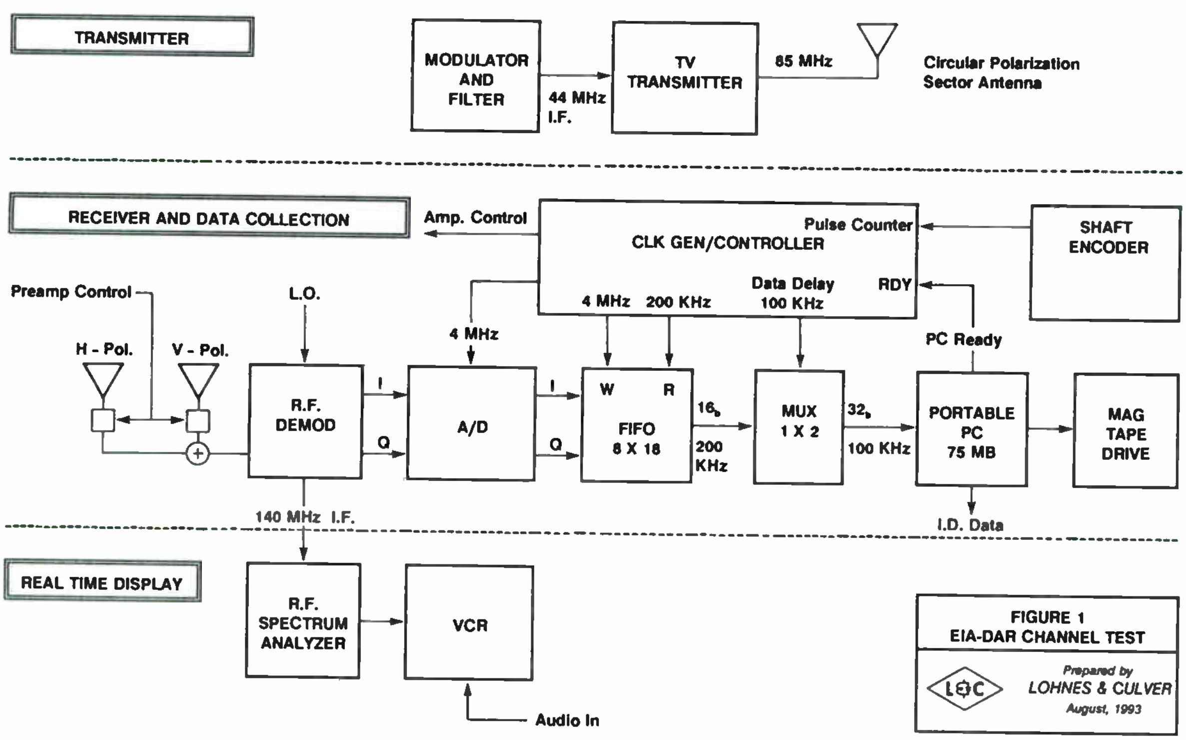
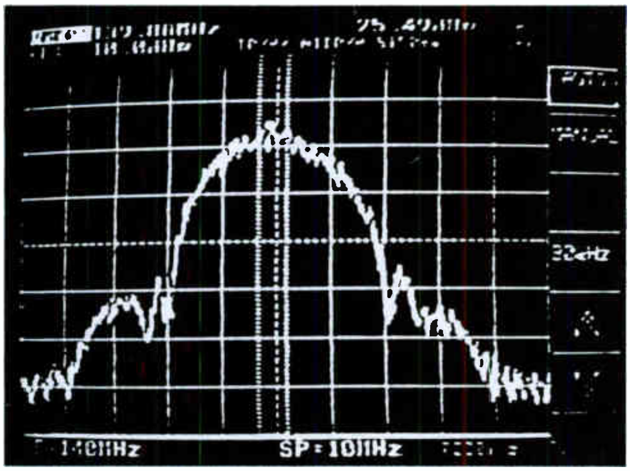


FIGURE 1
EIA-DAR CHANNEL TEST

The 2MHz pulse rate and BPSK (suppressed carrier) modulation generates a 4 MHz wide R.F. spectrum with its first frequency nulls at ±2 MHz and decreasing side lobes, each 2 MHz wide. The output is filtered as described above so that the transmitted signal appears as a 4MHz wide Gaussian distributed noise signal with rapidly decreasing side lobes as shown in the photograph of Figure 2.

FIGURE 2
Channel Characterization Test Signal



This spectrograph of the receiver I.F. output was taken with the test transmitter signal connected directly to the receiver antenna input. The horizontal span is 10 MHz, a scale of 1 MHz per division, with a vertical span of 80 dB, a scale of 10 dB per division. The test signal was transmitted with a center frequency of 85 MHz, the center of the television Channel 6 band, a relatively accessible frequency where ever a television Channel 6 station is not operating. This frequency is also sufficiently close to the FM band to accurately characterize it. Use of high stability reference oscillators (1 part per million) in both the transmitter and receiver allows for demodulation without frequency lock.

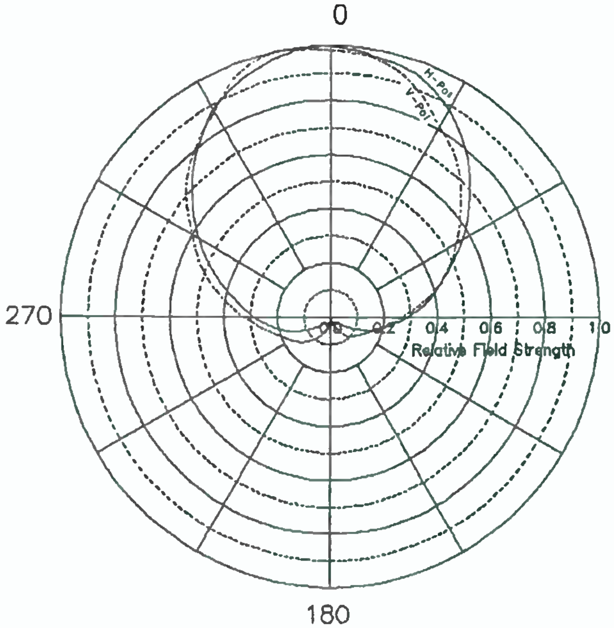
In Salt Lake City, a 6 kW Larcan transmitter was used with a two element LDL Spearhead Panel antenna configured with two elements stacked vertically and splayed ±30 degrees to provide a narrow azimuth pattern oriented east over the city and toward the mountains beyond. The resulting horizontal plane relative field patterns are shown in Figure 3.

The antenna was mounted at approximately 9100 feet AMSL or 4500 feet above the valley floor at the Bonneville Broadcasting KSL(TV) transmitter site on Farnsworth Peak, 18 miles west of Salt Lake City. The resulting ERP of 7.6 kW average power is sufficient to cover the city and surrounding area including the mountains up to 25 miles away.

Receiver

The receiver, using Superheterodyne radio frequency design, was also built by Delco. It was carried in a mini-van along with other associated equipment including a specially built "reference" antenna. Each of the two independently polarized antennas feed modular amplifiers and each amplifier output is combined in a power coupler into the single channel receiver. Under control of the test computer the antenna amplifiers are alternately turned on, completing one path and rejecting the other by over 40dB. In this combined R.F. output channel several tunable notch filters are available to remove strong out-of-band components without perturbing the pass band signal. The filtered signal is then mixed with a 140 MHz Local Oscillator (LO) to generate the 140 MHz I.F. and further filtered by a Surface Acoustic Wave (SAW) filter to provide up to an additional 40dB of out-of-band rejection.

FIGURE 3
Test Antenna Azimuth Pattern



At this point the signal is sampled for several monitoring functions. First, the IF is sampled to display the RF component on a spectrum analyzer with that display recorded on a video recorder. The IF is also mixed to baseband for monitoring by an oscilloscope to facilitate tuning the LO to zero frequency offset with the transmitter. In practice, approximately a 100 Hz drift occurs during one measurement run which has no significant effect on the data that is recorded.

Next, both the I and Q signal components are sampled by an 8-bit A/D converter. The resulting half-Least Significant Bit (LSB) uncertainty error is more than 40 dB below full level, limiting the dynamic range of the A/D conversion to approximately 40dB. A manual attenuator with a 50 dB range is adjusted on each data collection run to set the signal in the optimum digitizing range. The system has an effective sensitivity of better than -69 dBm in the presence of anticipated interference from strong nearby adjacent channel stations.

The received waveform is sampled at 4 MHz, twice the transmitted pulse rate (two times oversampled). The need for bit synchronization circuitry is eliminated by averaging every pair of data points. Data acquisition is controlled by an enable signal from the computer for each sample point. At each sample point the first polarization waveform is digitized and loaded into First-In-First-Out (FIFO) memory. Then the antenna amplifiers are switched and after a 1 microsecond settling time, the opposite polarization signal is digitized and transferred to FIFO. The antenna amplifiers are then switched back to their normal state to wait for the next sample point. The system remains in this position for at least 14 milliseconds while data in FIFO is multiplexed and loaded into the computer memory. This measurement cycle is dependent on the basic data transfer rate of 100kHz, from data acquisition board to computer, setting the maximum speed of data collection. That, in turn, sets the minimum computer cycle time which determines the maximum vehicle speed at the minimum sample point spacing chosen for this test. After collection of one data file, 1 M byte of data, the system pauses briefly to save the file to disc, clears the acquisition board and resumes collection.

Receiving Antenna

Several discussions within the EIA-DAR testing group, regarding the type of antenna to be used for reception testing, led to the conclusion that some type of "reference" antenna must be used. Considering the many types of vehicles and specific antennas available for each, literally thousands of possible combinations exist; no one of which could be considered standard.

The basic antenna requirements were chosen to provide omnidirectional horizontal plane reception with good bandwidth. Furthermore, the antenna must provide dual and independent outputs for the horizontal and vertical polarization components. The standard test antenna was designed in a joint effort by Delco, Lohnes and Culver and Harris. It is designed to receive both the horizontally and vertically transmitted components, but to receive them separately and independently with little interaction. Tests show that the antennas have a horizontal plane circularity of better than ±2.5 dB and a cross polarization self discrimination of at least 10 dB as shown in Figures 4 and 5 on the next page.

Figure 6 shows the test antenna mounted over a ground plane atop the van with its center approximately 3 meters above ground level. A monopole of approximately 1/4 wavelength is mounted over the ground plane described above. The precise monopole length and diameter was chosen to provide reasonable impedance and wide band performance. The vertical monopole antenna is surrounded by a dielectric structural tube which supports the top mounted horizontally polarized antenna. That antenna is designed to provide a nearly circular horizontal plane pattern with minimum interaction with the vertical monopole or the ground plane. The entire antenna and ground plane assembly was constructed and tested at the Delco facility in Kokomo, Indiana, with further testing at the Harris antenna facility at Palmyra, Mo.

The separate antenna lines were connected to medium gain amplifiers in the receiver, as described above, which could be biased on or off under control of the test computer, discussed below.

FIGURE 4
Standard Receive Antenna Horizontal Polarization

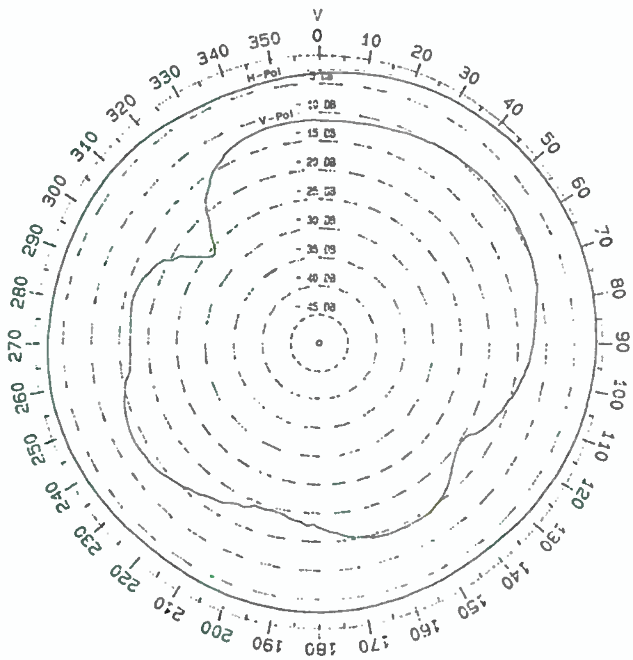


FIGURE 5
Standard Receive Antenna Vertical Polarization

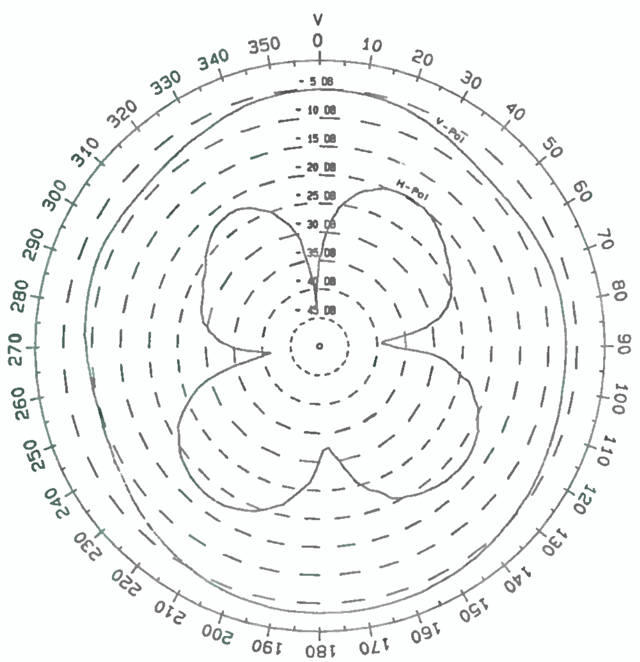


FIGURE 6
Test Van and Antenna

<u>Computer</u>

The test signal is received in the mobile van driven along a desired route. Uniform spacing of measuring points is assured by use of a trigger counter, driven by a shaft encoder on the van drive train. At the start of a measuring route the trigger counter and computer program automatically controls data collection. The crew in the van make notes of the run number, location and environment description, starting parameters such as attenuator setting and LO frequency, the time and date. The shaft encoder generates a string of pulses with vehicle motion and sends them to a programmable counter in the computer which, depending on its programming, sets the linear spacing of the data samples along a path. For this test, the sample interval was set at 0.35 meters or 1/10 wavelength at the 85 MHz center frequency, close enough to define the spacial distribution of multipath characteristics along the path and hence over an area.

At each data collection point, the on-board computer executes a routine which collects data over a span of 3 continuous PN sequences to ensure that after analysis at least one full instance is available. The received wave form is digitized to 8 bit resolution for the default polarization and then immediately repeated for the other polarization as described above. Data collection is continued along a path for up to 4.4 kilometers, 75 MB of data, until the computer hard drive is full and halts collection or when a convenient stopping landmark is reached prior to that.

At that point the 75MB data file is transferred to tape and ending parameters are noted such as; path end landmark, time, file length and LO frequency. After approximately 15 minutes required to transfer data to tpae storage, measurements can be resumed.

The collected data is grouped into 1 MB files each containing 160 samples of approximately 6KB each and covering 56 meters. A one kilometer path requires approximately 17 MB of data and a full 75 MB disk covers approximately 4.4 kilometers of path before data transfer to tape is required. The point spacing and the speed of the computer system allows for data to be collected at up to 60 KPH. However, slower speeds make data collection easier for the crew in the van.

At each sample point <u>both</u> the horizontal and vertical polarized signal were received and recorded. To do this the computer controlled the antenna amplifiers as described previously. The transition time between amplifiers is short enough to limit the distance moved between the horizontal and vertical samples to less than 1/100 wavelength at the maximum data collection speed. In reality, at lower speeds, the interval is less and the two samples are collected at nearly the same location.

Other System Equipment

As data was collected the computer displayed the 1 M byte sequential file number last written. At easily identified landmarks, such as cross streets, the file number, the landmark and other data, such as time, were written down by the van crew. Later, by plotting the measurement path and the file numbers on detailed topographic maps, the exact position at which data was recorded can be determined by linear distance interpolation along a path. Also on board were a spectrum analyzer and a video recorder. The analyzer, connected to the receiver I.F. output, displayed the signal spectrum after conversion to the 140 MHz I.F. The spectrographs attached as Figures 7A through 7D again show the reference signal and several typical multipath faded channels displayed by the analyzer.

The receiver sampled both the horizontal and vertical polarizations before conversion to I.F. but, because of the strong time bias toward one antenna, the predominant signal displayed is the default polarization. The spectrum analyzer has an update rate of less than five times per second which presents only a "snapshots" of the spectrum when vehicle speed is relatively high. At very slow speeds, however, even this slow update is fast enough to show a slowly changing spectrum.

FIGURE 7-A
Direct Input Test

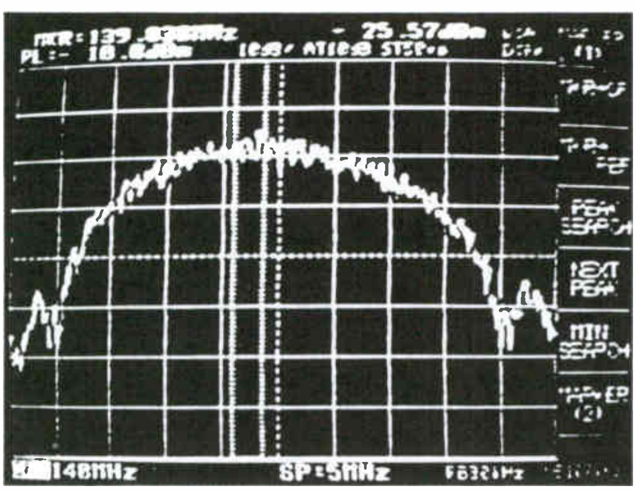


FIGURE 7-B
Sample Spectrograph

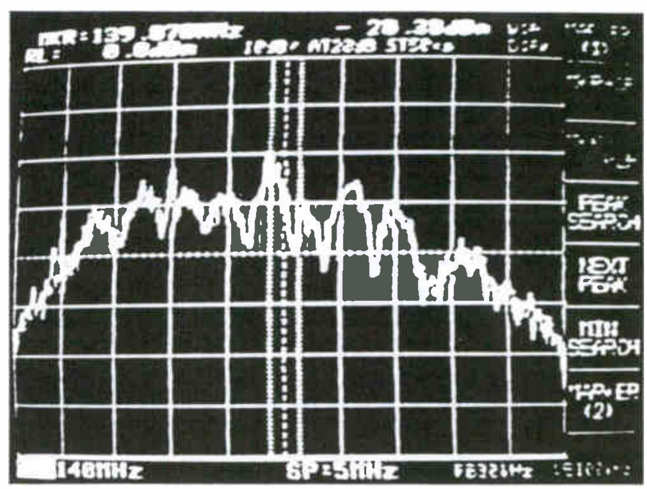


FIGURE 7-C
Sample Spectrograph

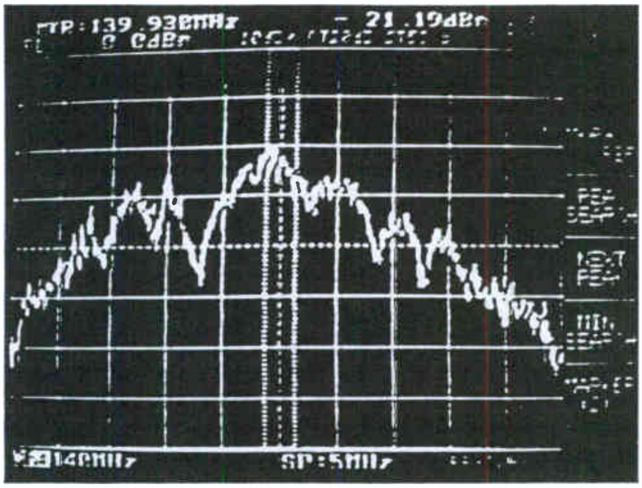
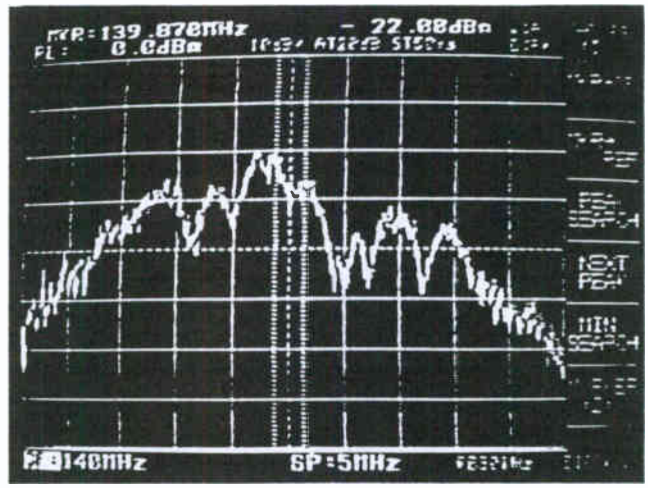


FIGURE 7-D
Sample Spectrograph



The recording of the spectrum analyzer display and the audio from a microphone in the van gives a continuous update on activities in the van. Thus, as each landmark was called out, the tape would contain narration describing the path and measurement positions.

DATA ANALYSIS AND REPORTING

The data that is generated by the test system is prodigious. At 17 M byte per kilometer, over 500 M bytes of data was collected in one day and nearly 3 Gigabytes of data was collected in one week. Data analysis requires identification of one full pulse instance and de-correlation of the data. Delco electronics and SEIKO Telecommunication Systems, Inc. of Beaverton, Oregon have contributed their time and computer facilities to perform the data reduction using the Math Works "MATLAB" software. The extensive matrix math operations involved in the data reduction require significant computer speed. The computers available included an IBM model RS/6000 and HP series 200 work station. Data analysis and presentation after bulk reduction is possible on a fast PC.

The data is processed in two logical sections. First, the data is averaged in a 2 bit moving window, decorrelated and frame synchronized to extract the single I/Q frames for the two polarizations at each sample point. This processed raw data is in a convenient format for further analysis. The second processing section analyzes the data for this study, extracting the parameters that will be used to program the multipath simulator and determining the statistics of the multipath fades in both time and frequency domains.

The first step in analyzing the processed data is to extract the relative time and magnitude of the five primary reflections in the channel. These delay values (plus the direct signal) will be programmed into the six channel multipath simulator. Knowing the time delay and the complex magnitude of the reflections allows the frequency domain effects to be calculated. This method of transforming the data from time to frequency is used to determine the significance of a reflection by assessing its frequency domain impact. Each of the significant reflections are shown by the cross tic-marks (+) on the time domain graphs. The corresponding frequency domain is shown as a "smooth" curve for the 6 or fewer significant points and as the "rough" curve for all reflection points in each of the 250 nanosecond bins.

The analysis is done by a complex algorithm based on linear programming, using the modified forward-backward linear prediction method described by Tufts [3] and suggested by Herman. It is a powerful algorithm for resolving reflectors spaced closer than the time between two transmitted pulses (chips), a single timing bin, in this case 500 nanoseconds, and in the presence of noise. With this method reflectors spaced less than half of a timing bin can be resolved at the minimum receiver SNR of 10 dB. In this method the channel is modeled as a Finite Impulse Response (FIR) filter with the number of filter elements is set at twenty. This allows the algorithm to perform nicely without using undue processing time that would be required for a larger filter order.

PRELIMINARY SALT LAKE CITY TEST RESULTS

The parameters of delay time and magnitude, over a set of instances that are grouped by physical environment, provide the path information to be programmed into the multipath simulator. Samples of this extracted data are shown on the following pages in Figures 8.A.1 & 2 through 8.D.1 & 2, presenting data for only one instance in each of the four chosen "environments" A,B,C and D described later. Samples of the time domain data transformed to frequency domain are also shown on those Figures.

A visual representation of time and magnitude data over one full set of 160 samples is shown on the waterfall graphs of Figures 8.A.3 through 8.D.3. Figures 8.A.4 through 8.D.4 show the resulting scattering function displays with Doppler shift vs. delay and magnitude. Mean and variance parameters are extracted to show a statistical profile of each environment as shown on Figures 8.A.5 through 8.D.5.

"Environment" Categorization

The test van was driven throughout Salt Lake City and the surrounding area collecting data along 21 paths ranging from 4.4 to 8.8 kilometers, zig-zaging across or encircling an area, for a total of approximately 140 kilometers of continuous data. The data was collected and numbered with a 5 digit identifier, two digits relating to the path number and three to the individual 1 M byte data files. Therefore, because each landmark was identified by the file being written at the time, the location along a path at which data was taken is known with the precision of each 1 M byte of data, 160 samples or 56 meters.

Each of the 21 paths was analyzed for the type of "environment" (buildings, vegetation, terrain, urbanization, etc.) that was predominant along the path or a significant portions of it. Preliminary review disclosed eight narrowly defined environments which were reduced to the four major environment categories listed below. Figures 8.A.1 through 8.D.5 correspond to environments A through D.

A. URBAN BUSINESS-INDUSTRIAL-COMMERCIAL:

This environment includes; the central city business district and surrounding area, approximately 6 blocks north-south by 8 blocks east-west; an industrial/warehouse area south west of the city center; and several moderately heavy commercial areas comprised of shopping centers and malls with multi story buildings.

The traffic paths are generally close to the buildings indicating that the expected reflections should be short in time delay and generally strong in level. The practical vehicle speed in the area ranges from stopped to approximately 40 KPH (25 MPH). Numerous stop lights and stop signs require stops from 0 to 45 seconds at frequent intervals. A simulation consisting of alternate stopped and moving data will be used.

B. SUBURBAN - RESIDENTIAL:

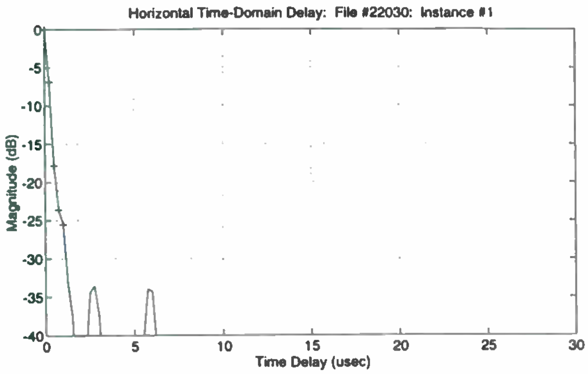
This environment includes many residential areas, scattered to the south of the city center. They range from single family large homes on large lots to duplex town houses and apartments. As a consequence, the neighborhood may include light commercial shopping areas, schools, libraries, college buildings, etc. Some of the typical older residential areas were heavily covered by mature trees of up to 25 meters in height.

The traffic paths are generally more open than for the urban environment but still relatively close to some structures. The expected reflections should be moderately short in time delay and moderately strong in level. In the more open areas in the environment, longer and weaker delays may predominate. The practical vehicle speeds still drop to zero on occasion but may increase to 70 KPH (45 MPH). A simulation with varying speed and few stops will be used.

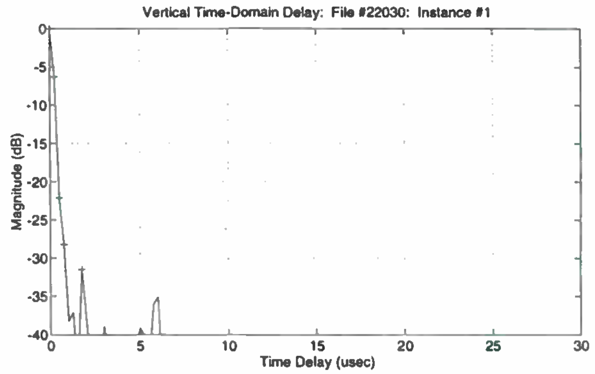
C. RURAL - PARKWAY - HIGHWAY:

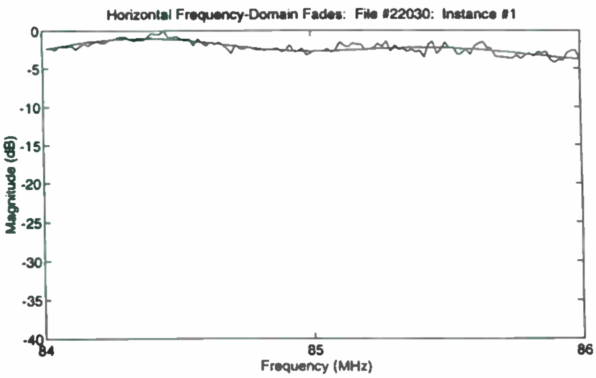
This environment includes generally open areas with a variety of roads ranging from; single lane country roads north west of the city, to divided

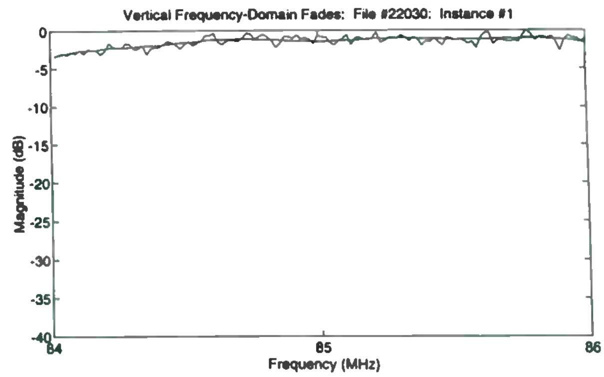
8.A.1



8.A.2







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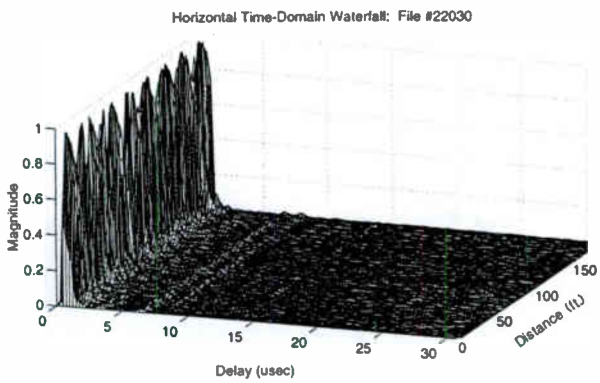

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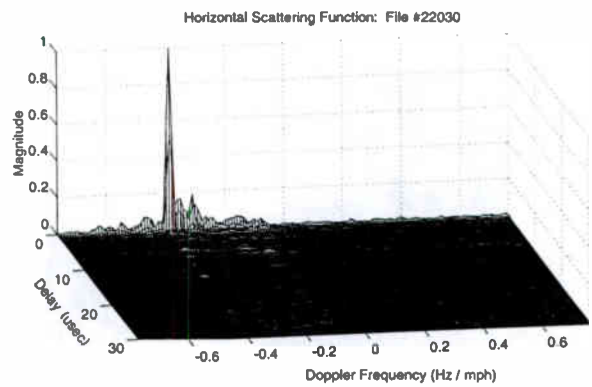
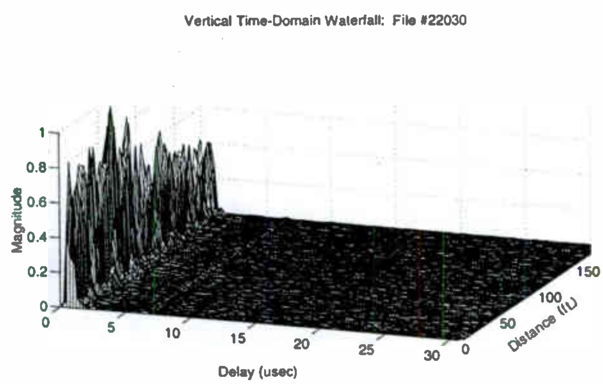
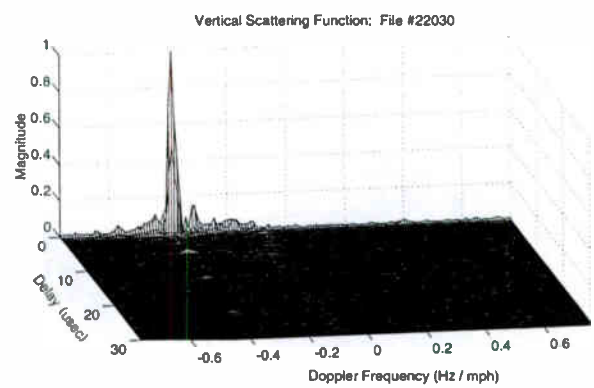
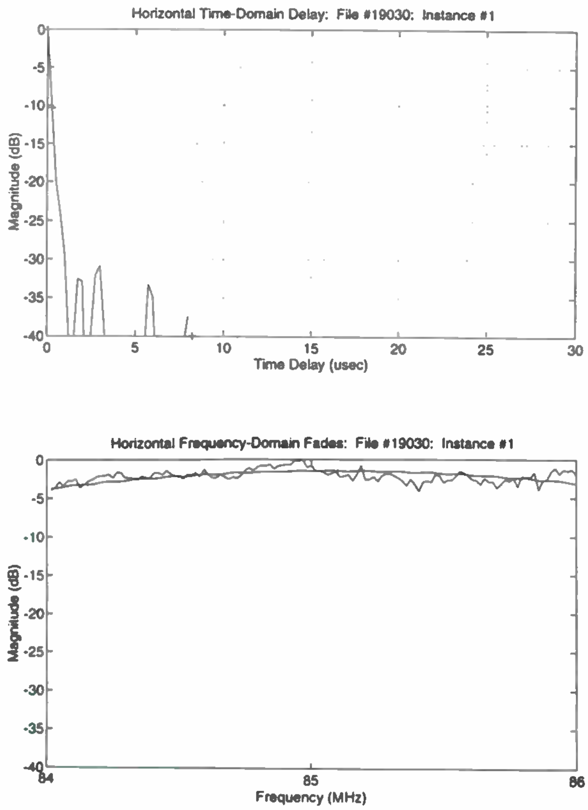






8.B.1.



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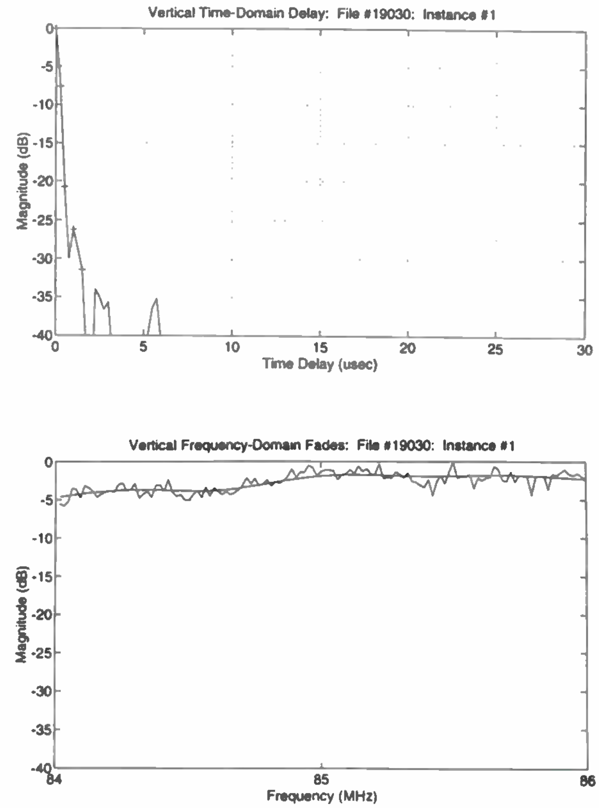

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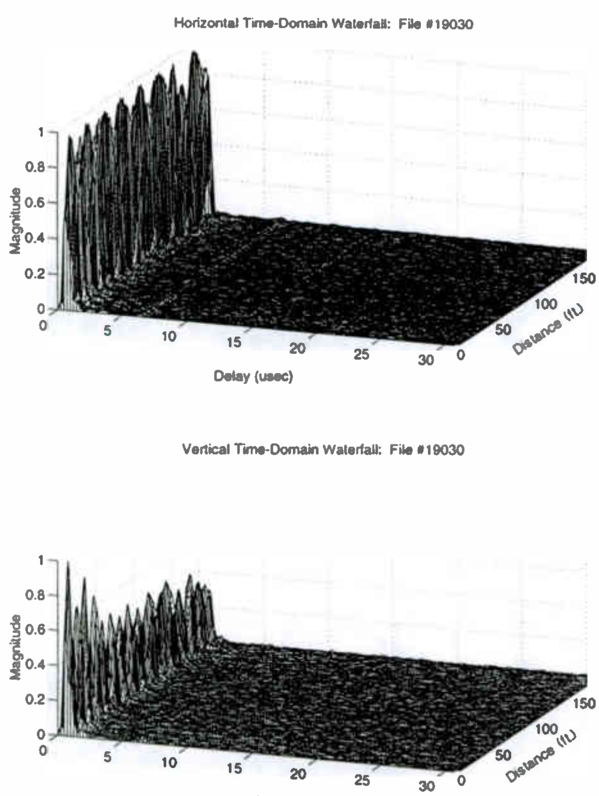

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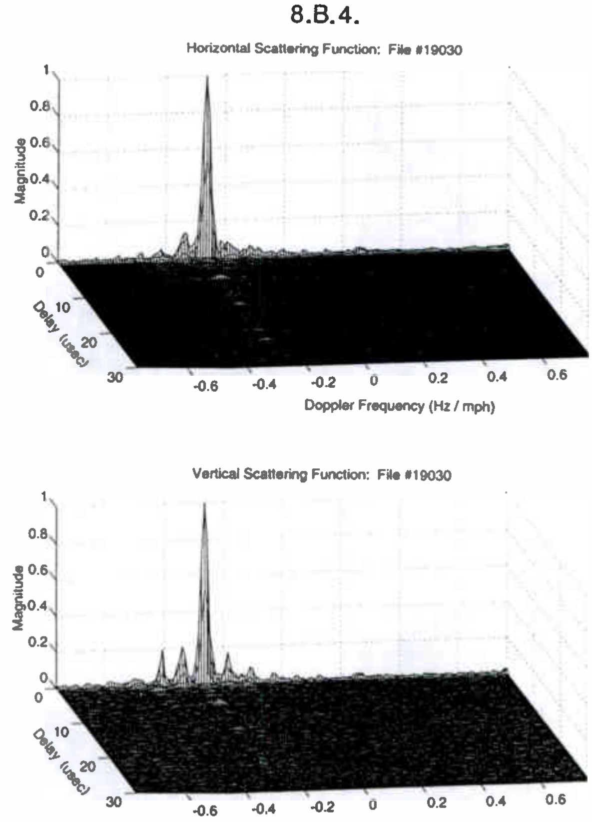

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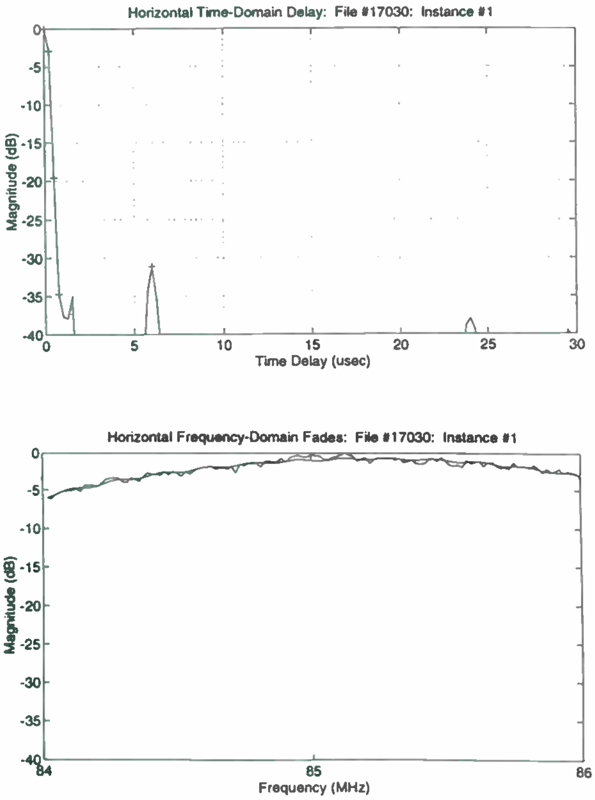

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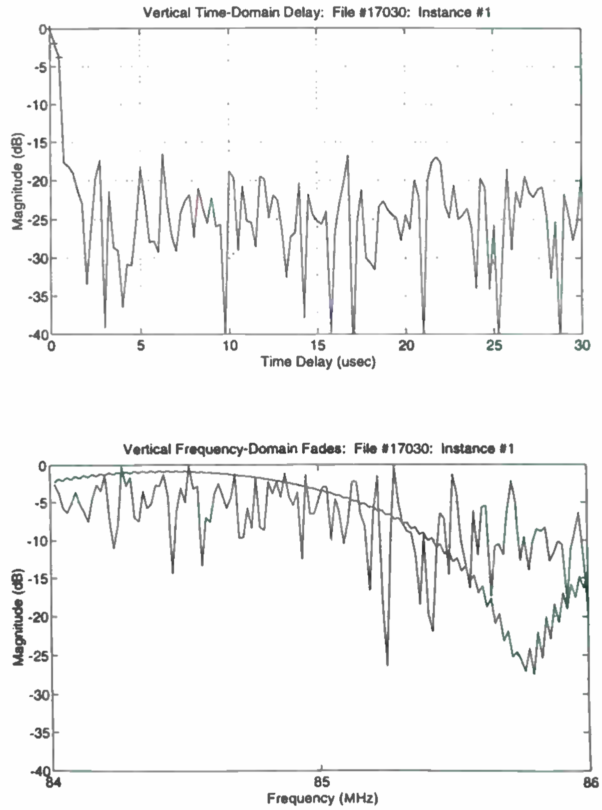

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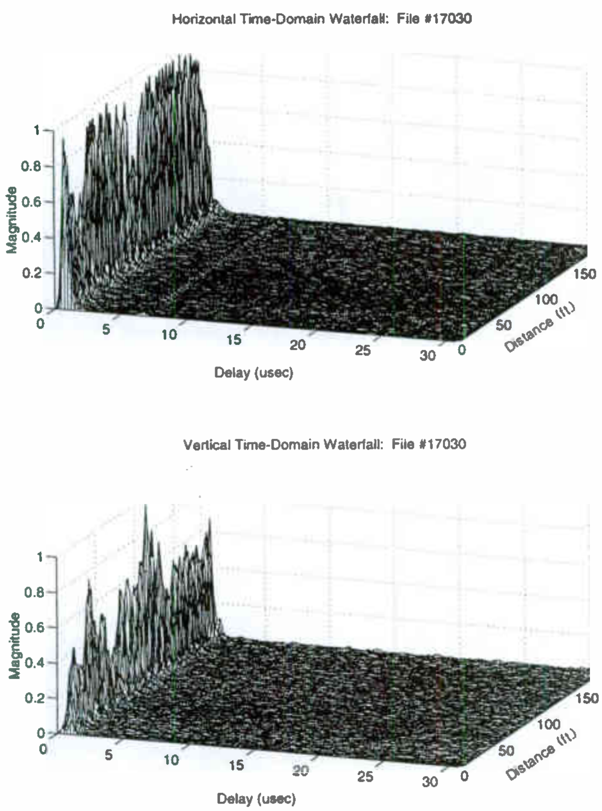

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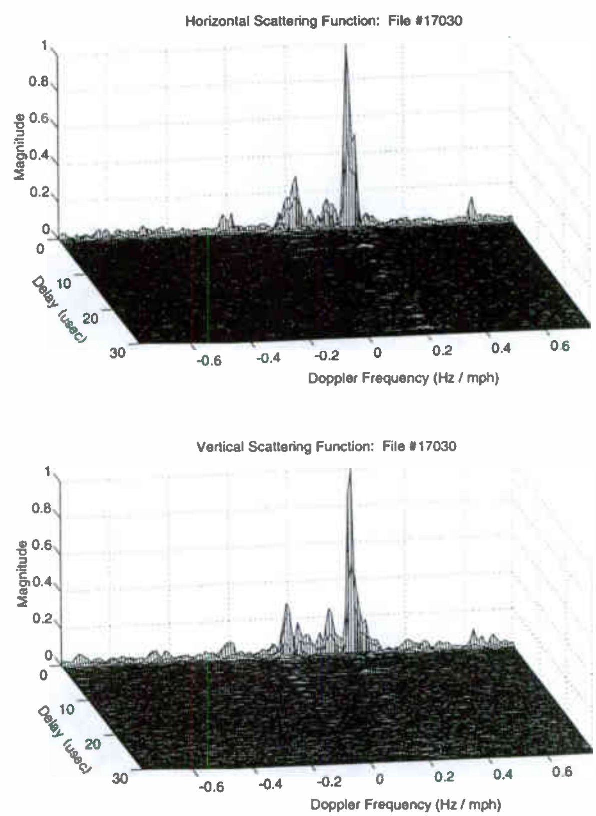

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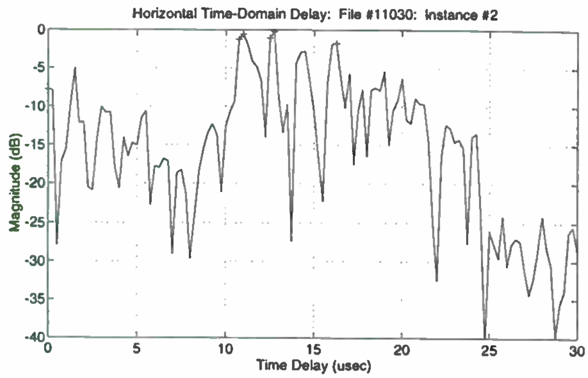

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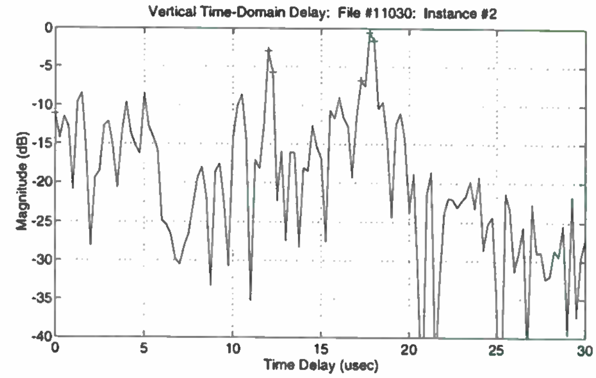

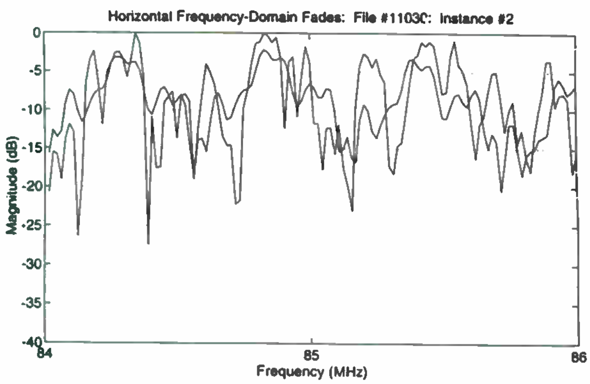

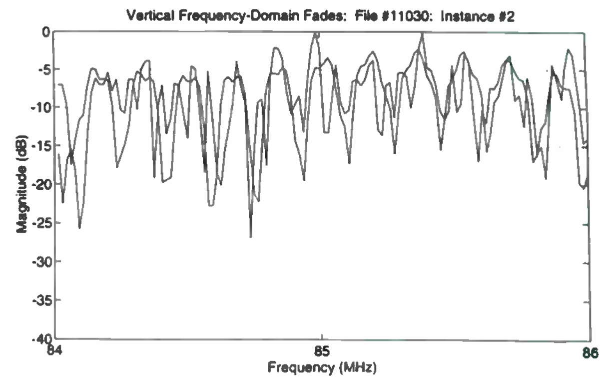

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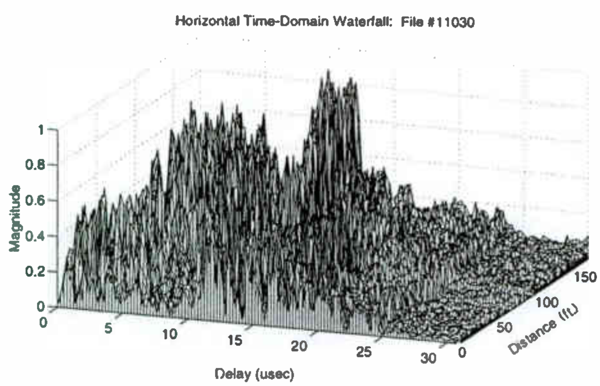

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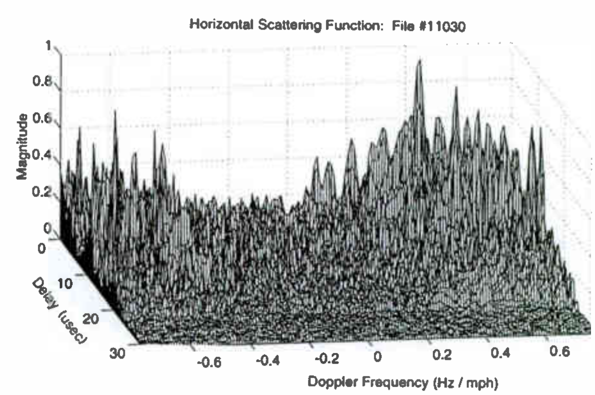

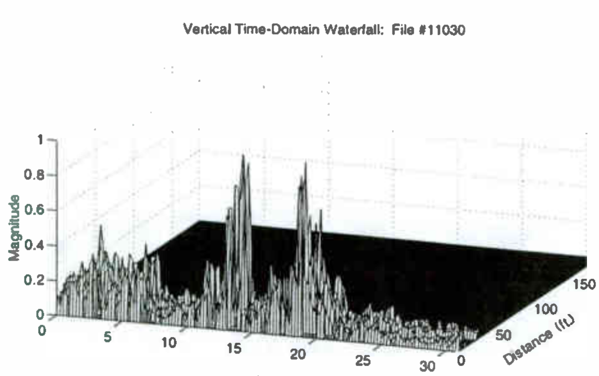

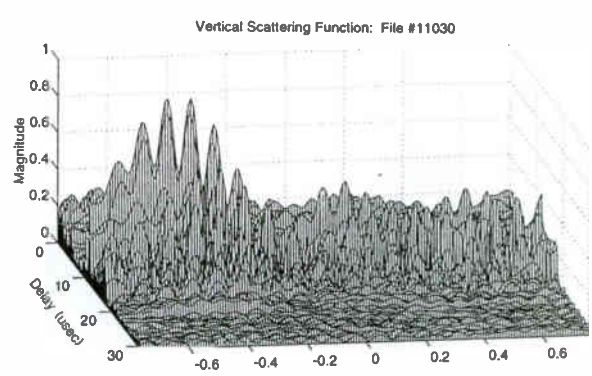

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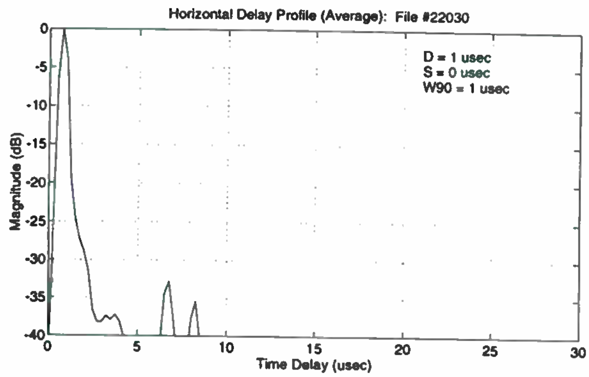
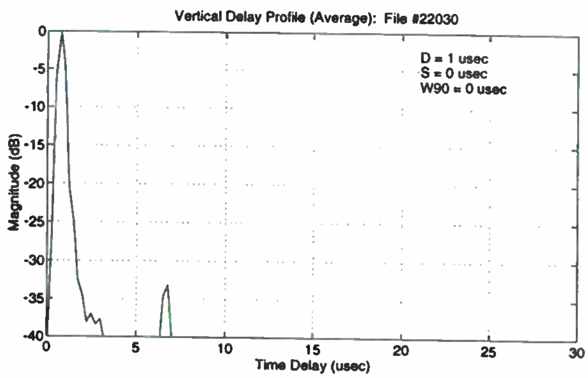
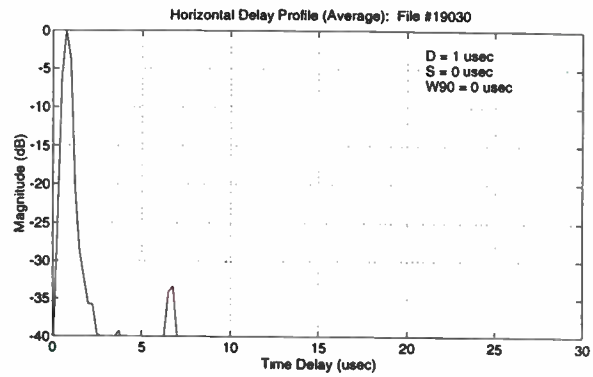
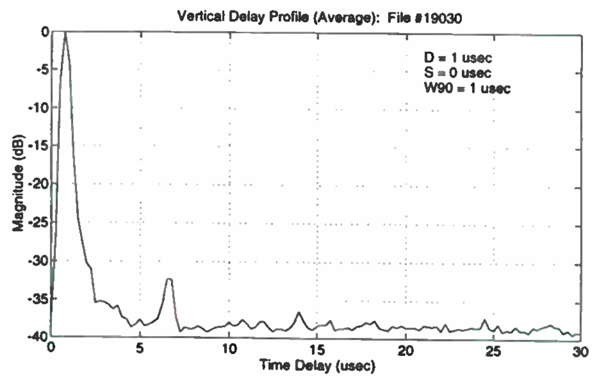


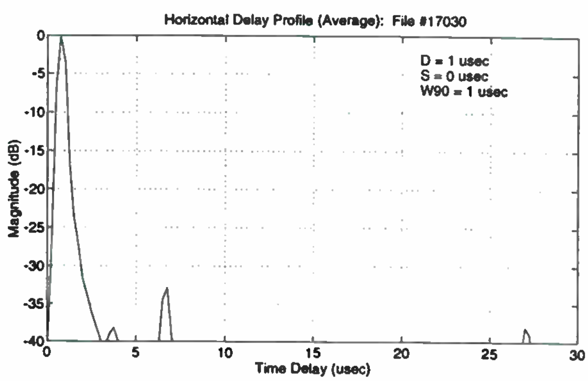


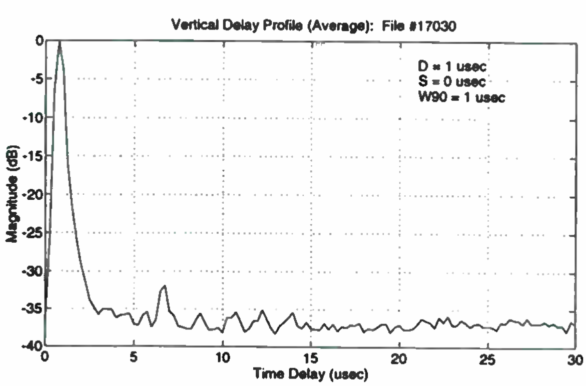
8.B.5.





8.C.5





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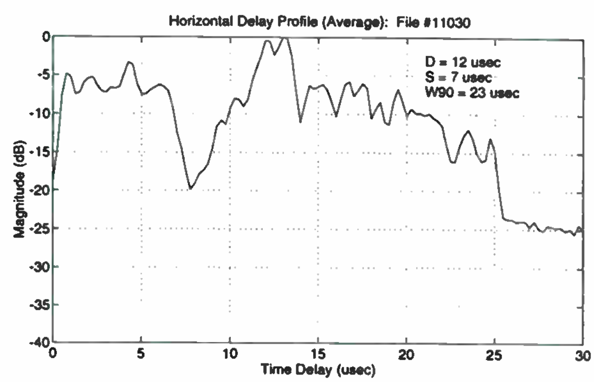

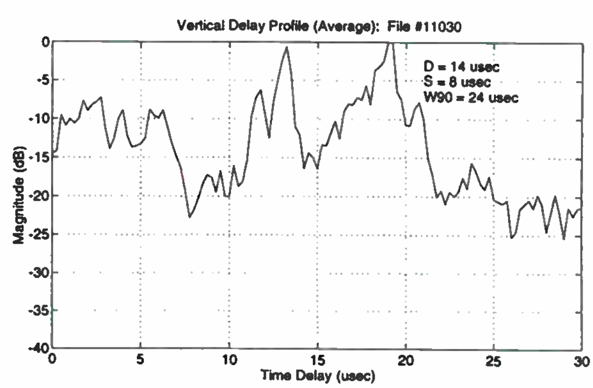

highway parkways south of the city, to a commercial boulevard east of the city at the foothills of the mountains.

The open nature of the areas generally keeps the closest structures moderately far from the measurement path, with the structures themselves including all types from multistory office buildings to small residential buildings. The expected reflections should be moderate but cover a relatively wide range of time delays and magnitudes. The practical vehicle speeds should very rarely drop to zero. The range to be used is from 40 to 100 KPH (25 to 60 MPH) with one stop.

D. TERRAIN OBSTRUCTED PATHS:

This environment will include those paths which have a significant terrain involvement, including the Big and Little Cottonwood canyons and the few paths that followed river valleys in the area south and east of the city.

The terrain along these paths is the significant factor but some, particularly the river valley path, have some structures along the path. The expected reflections should be numerous and rapidly changing with moderate to long reflections over a wide range of magnitudes. The major feature of the signal path should be the frequent loss of a direct path with the dominant reflection becoming the substitute direct path, one which is subject to rapid change as reflection characteristics change. The practical vehicle speeds should rarely drop to zero but should also not be too high. The range to be used is from 25 to 70 KPH (15 to 45 MPH) with one stop.

LABORATORY TESTING - Multipath Simulation

The general parameters and operations of the DAR systems to be tested are well known but some of the system specific parameters that will affect testing are as yet unknown. The characteristics of multipath propagation which will affect those systems are also generally well known with the precise values of those characteristics being determined by this Channel test. The characteristics and application of the Hewlett Packard multipath simulator are generally understood but will become better known upon actually using the instrument with the data from the test.

As a result of these unknowns and variables, the actual fine structure of the tests will not be known until just before the start of actual testing. Several parameters which may be subject to change include the length of the test samples in each environment, speed and number of stops. The range over which the testing will be conducted, however, can be determined at this time from the channel test data.

This choice of four environments reduces the time burden on the laboratory testing, primarily in the audio listening tests. CRC, the audio test segment contractor, has stated that the test segments should be relatively short, preferably 30 seconds, and no more than 60 seconds. Within this constraint, a multipath test simulation must be devised which presents the full range of multipath conditions of an environment to each system under test. Some system parameters will define the minimum time in which an environment can be simulated. For example, environments which require zero velocity testing will be examining the system performance under fixed fade conditions, performance highly dependent on the length of data time interleaving in the system. Each simulation stop must be long enough to test all systems and short enough to test a sufficient number of stops without exceeding the maximum audio sample time. It is anticipated that a minimum of five stops would be necessary and exceeding the 30 second test time is likely.

If a segment of actual measured data, or several shorter segments for a "stopped" test, can be found which adequately represent the full range of multipath conditions in an environment, then there is no apparent reason that actual data should not be used to program the simulator. If, however, this is not the case, then it may be necessary to define the parameters over the maximum range and generate an artificial but representative segment. This choice can only be made after all of the channel data can be analyzed and the systems under test and simulator limitations are known. This will only affect the actual method of simulated testing while the general range and limits of testing will remain unchanged, being indicated by the data measured in Salt Lake City.

CONCLUSIONS

This procedure for data gathering and analysis will deliver results directly applicable to the laboratory multipath simulation testing.

Measurements will have been made over the variety of terrain around the test site. Data will include central city areas, hills and rolling terrain in suburban and rural areas and mountainous areas. The typical range of time delay and magnitude of the reflections will be identified with the different areas. The mean and variance of the reflection delay time and complex magnitude will be determined and used to guide the laboratory testing with the multipath simulator. From this information other parameters such as the channel frequency domain characteristics and statistical data will be computed. The data that is collected will be of direct value to the DAR proponents and others interested in the multipath characteristics of the VHF channel. In its processed form the data is well suited for further analysis ranging from frequency domain characterization to point-by-point generation of a multipath model.

The VHF channel characterization test is a joint effort of many parties who have contributed their time and materials. EIA would like to acknowledge the assistance of all parties who have contributed to this ongoing program, particularly the individuals in the EIA WG-B Channel Test Sub-Group; Brian Warren chairman, and other industry volunteers. Corporate assistance has been provided by; PBS for the use of the Charlotte ATV test site, tower site manager Loadstar, Larcan transmitters, Alan Dick antennas, Bird, Bonneville International Corp., Cablewave, Delco Electronics, Dielectric, Harris, Herritage Media Corp., Passive Power Products, SEIKO Telecommunication Systems, Inc., and a host of others.

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DAB—II

Wednesday, March 23, 1994

Moderator:
Ken Springer, NAB, Washington, DC

***DAB ON TRIAL—EUREKA 147—THE SYSTEM WITH A FUTURE**
Paul Ratliff
BBC
Tadworth, Surrey

***NASA-JPL DAB SYSTEM UPDATE**
H. Donald Messer
Voice of America
Washington, DC

THE AT&T DAB SYSTEM UPDATE
Nikil Jayant
AT&T Bell Laboratories
Murray Hill, NJ

AT&T/AMATI DAR SYSTEM: AN UPDATE
John Bingham
Amati Communications
Palo Alto, CA

***USA DIGITAL RADIO FM DAB SYSTEM UPDATE**
Paul Donahue
Gannett Radio
Los Angeles, CA

***USA DIGITAL RADIO AM DAB SYSTEM UPDATE**
E. Glynn Walden
Group W
Philadelphia, PA

DAB SYSTEM BAKE-OFF
Judith Gross
Media Scan
New York, NY

*Papers not available at the time of publication

THE AT&T DAR SYSTEM UPDATE

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Murray Hill, New Jersey

Abstract

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

1. Introduction

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

2. In-band DAR Systems

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

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

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

3. The PAC Audio Codec

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

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

Masking in the *time-domain* (also called *non-simultaneous* masking) is illustrated in Figure 3, where a strong signal in a given time block can mask an adequately lower distortion in a previous or future block (*backward* or *forward* masking). Although less understood than simultaneous masking, temporal masking is implicitly used in block-transform coders such as PAC. In these coders, a long block length (for example, 512 samples) is used whenever possible to provide fine frequency analysis, and consequently better JND thresholding and coding gain. However, during occurrences of sharp transients in the audio, a long block length (with a possible audio onset late in the block) can cause a *pre-echo* (distortion lasting for a large or full fraction of the block length) that precedes the audio itself. In such cases, a shorter block length maximizes the possibility of temporal masking, in spite of the decaying skirts in Figure 3. An important feature of PAC is a *window-switching* algorithm (Figure 4) that can change the window length from 512 samples to, say, 128 samples, with the switching based on a proprietary perceptual algorithm.

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

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

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

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

4. Strategies for Robust Transmission

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

The AT&T DAR systems provides adaptive equalization to alleviate frequency-selective channel fading (Section 5), time-interleaving to randomize bursty errors due to slow fading, and three levels of protection against residual bit error effects:

(i) an *outer* code: protection of a small number of very critical bits in the transmitted bit stream (using an 8 kbps overhead out of a total of 160 kbps).

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

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

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

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

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

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

Table 1: AT&T DAR System: Channel Coding

CSNR (dB)	Delay (ms)	Mean Time between Failures (sec)				
		(32, 20)	(32, 18)	(32, 16)	(32, 14)	(32, 12)
17	425	–	.02	.07	0.21	0.75
17	725	.03	.08	.30	1.2	4.70
20	425	–	5	–	–	–
20	725	45	150	–	–	–

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

5. DAR Modem and Transceiver

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

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

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

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

6. The DAR System: Primary and Secondary Modes

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

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

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

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

7. Future Enhancements of the DAR System

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

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

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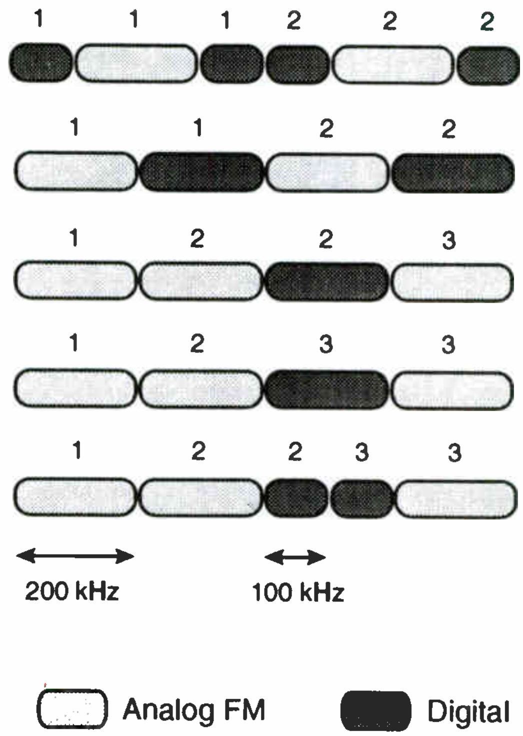
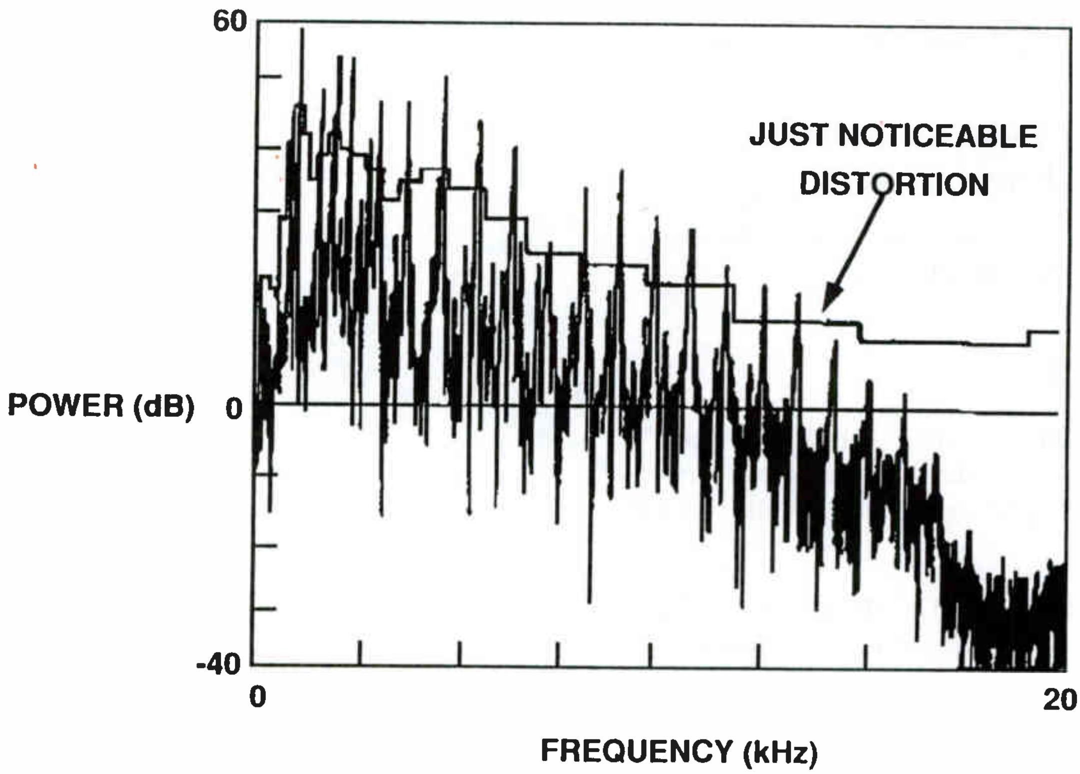
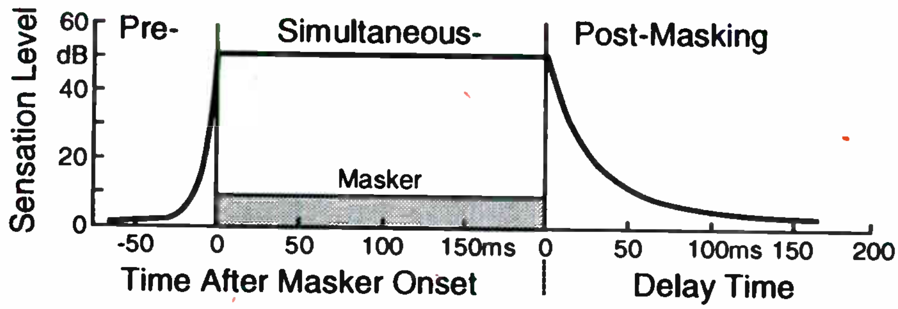


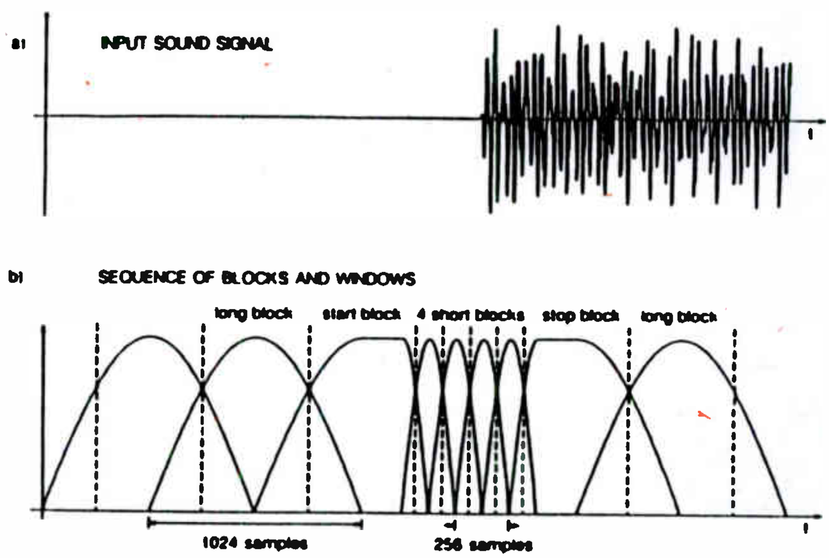
1. Alternative paradigms for analog and digital audio broadcasting.



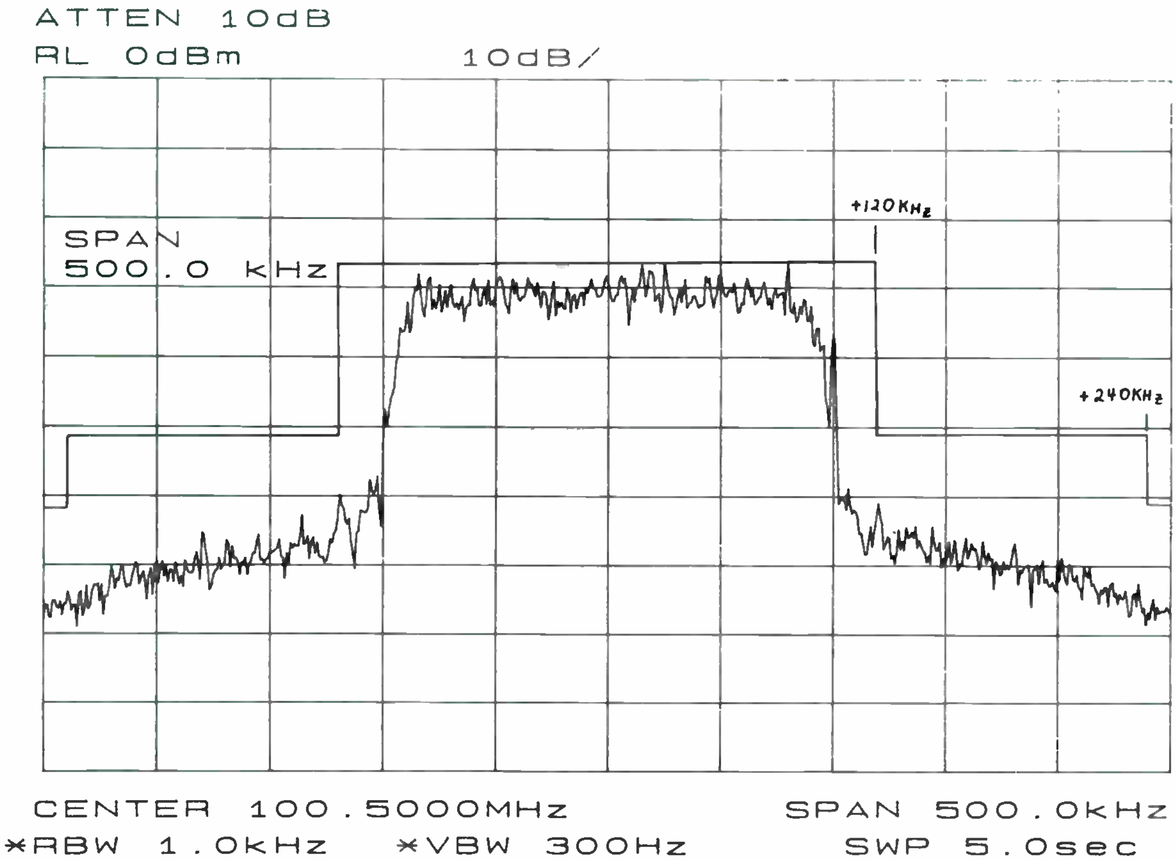
2. Masking of distortion by signal, in the frequency domain.



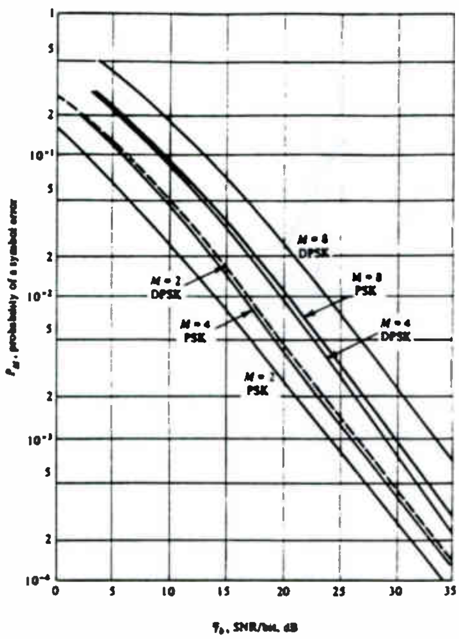
3. Masking of distortion by signal, in the time-domain.



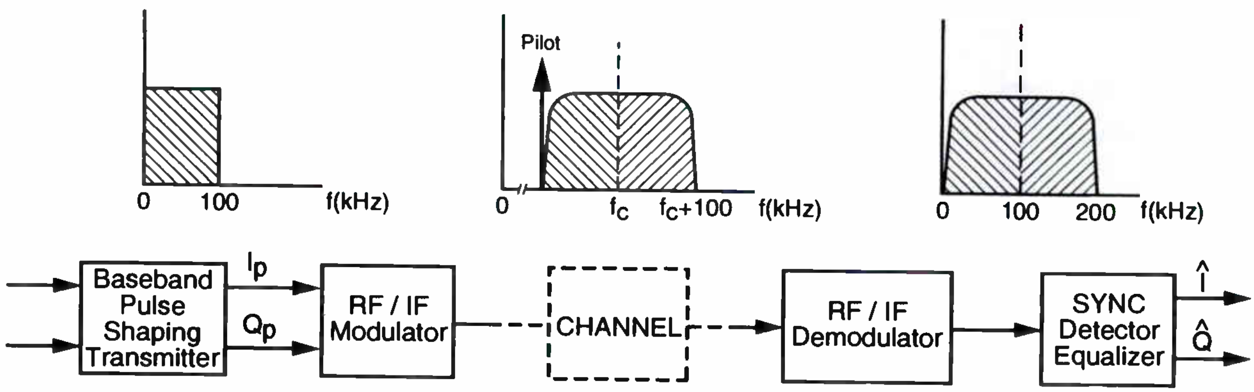
4. Adaptation of window length in transform coding.



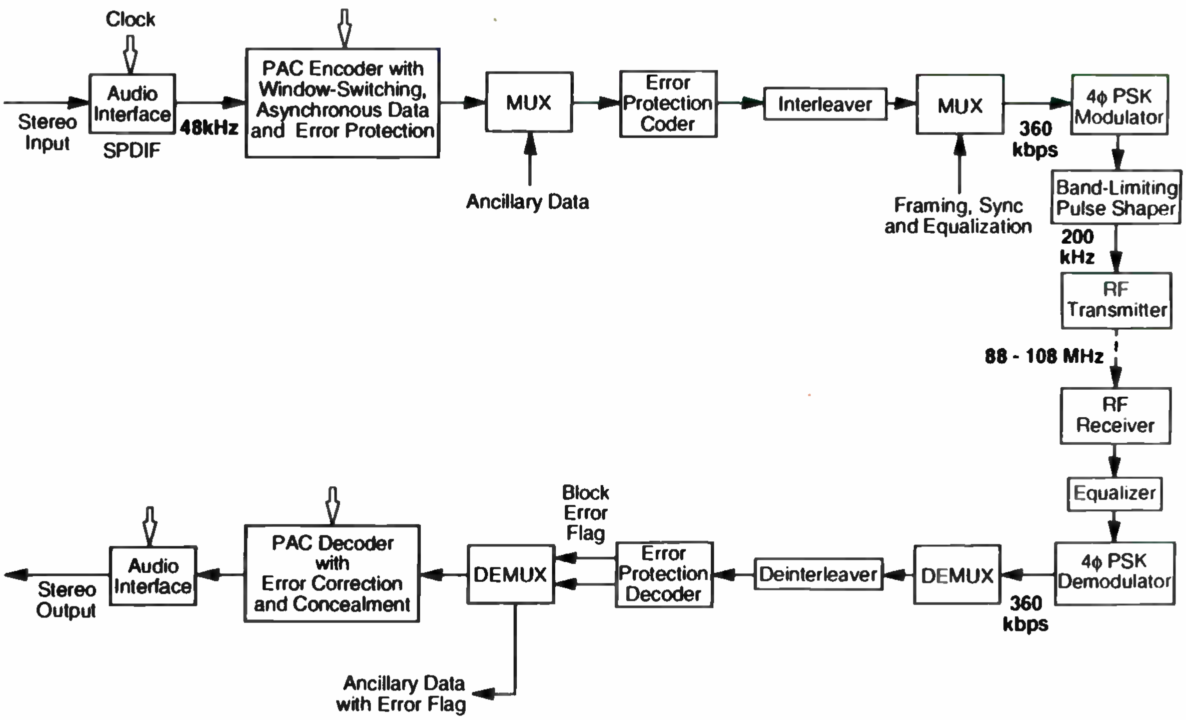
5. Transmitted power spectrum in the AT&T-DAR system, and the FCC-mask for broadcasting in the FM radio band.



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



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



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

AT&T/AMATI DAR SYSTEM: AN UPDATE

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ABSTRACT

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

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

INTRODUCTION

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

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

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

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

IN-BAND DAR

The Environment

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

The IBOC Problem

The four requirements for any IB system that dictate the design approach are:
1. Under conditions of multipath fading the DAR receiver must operate much better than a good FM receiver receiving a conventional FM signal. This is the basic reason for interest in DAR.
2. The DAR signal should not interfere in any way with its host FM, and, similarly, should not be affected by it.
3. Both DAR and FM receivers must operate under theconditions of (a) the more common second-adjacent-

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

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

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

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

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

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

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

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

MULTICARRIER MODULATION

The Choice of Multicarrier Modulation

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

The problem of ISI caused by large delay spreads could perhaps be solved with single-carrier modulation by equalization of the received signal, but the computation required to make the equalization adapt fast enough to track a moving receiver[2] is a very challenging one. Another solution is to use multicarrier modulation [4] with a guard period (cyclic prefix) whose length is greater than the largest delay spread. This is the solution adopted in the Eureka 147 system where multicarrier modulation is called Coded Orthogonal FDM (COFDM). Because of the large delay spreads encountered in receiving from the many transmitters in an SFN the guard period in the Eureka system is much longer than is needed in an IBOC system.

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

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

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

The Amati Discrete Multitone (DMT) Solution

The Transmitter

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

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

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

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

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

The Receiver

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

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

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

THE AUXILIARY OVERHEAD CHANNEL (AOC)

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

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

The present system was developed to be used with a single transmitter; the symbol and guard periods are not long enough to deal with the larger delay spreads encountered in a multi-transmitter system[5]. Amati is studying the

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

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

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

PRELIMINARY RESULTS

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

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

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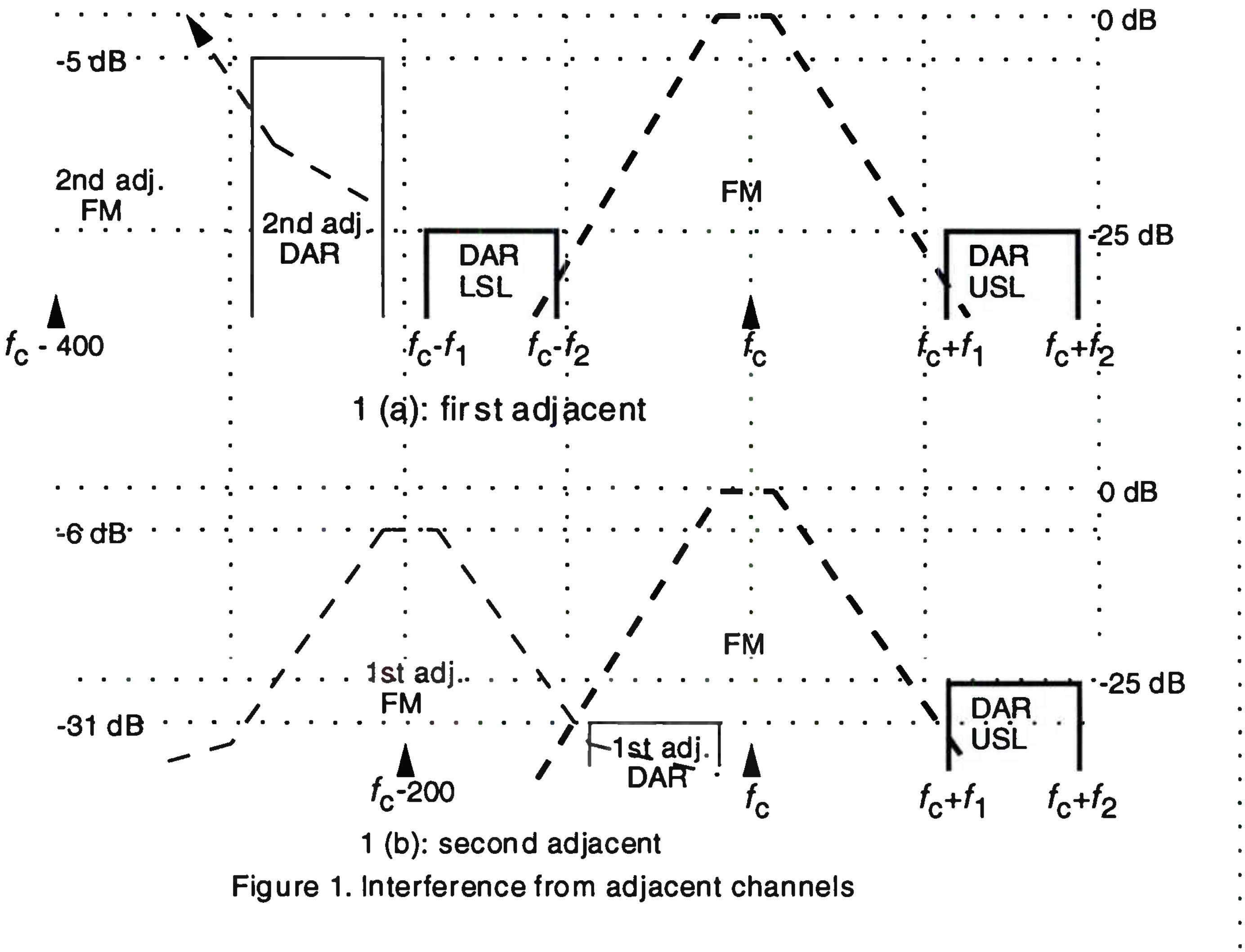

Figure 1. Interference from adjacent channels

SATELLITE AND AUXILIARY SERVICES

Wednesday, March 23, 1994

Moderator:
Carlos Girod, PBS, Alexandria, VA

***HOW THE FCC RULING FOR EARTH STATION ANTENNAS WILL IMPACT BROADCASTERS**
Anthony Campbell
Andrew Corporation
Orland Park, IL

***THE AFRISPACE INTERNATIONAL DIGITAL AUDIO BROADCASTING SYSTEM**
Noah Samara
WorldSpace
Washington, DC

***CONFIGURING REMOTE CONTROL FOR AN ENG SITE**
Bill Dumm
R.F. Technology, Inc.
Norwalk, CT

***THE PBS SATELLITE SYSTEM—AN UPDATE**
Carlos Girod
PBS
Alexandria, VA

*Papers not available at the time of publication

DATA BROADCASTING: TV

Wednesday, March 23, 1994

Moderator:
Bob Ogren, LIN Television Corp., Providence, RI

***NTSC DIGITAL BROADCAST: POSSIBLE APPLICATIONS OF THE TECHNOLOGY**
Douglas Yoon
Philips Interactive Media
Washington, DC

***WAVEPHORE: RELIABLE DIGITAL DATA OVER VIDEO**
Charles Jungo and Bruce Cross
WavePhore
Tempe, AZ

***INDEX PLUS+: AN APPLICATION OF EXTENDED DATA SERVICE IN VCRs**
Daniel Kwoh
Gemstar Development Corporation
Pasadena, CA

***NETWORK MANAGEMENT AND PERFORMANCE REQUIREMENTS OF A DATA BROADCASTING NETWORK**
David Boroughs
National Datacast (PBS)
Alexandria, VA

****BROADCASTING MULTIMEDIA—A COMPLEMENTARY SERVICE OR THE MAINSTREAM OF THE FUTURE?**
Anders Ahl
Sveriges Television
Stockholm,

***BROADCAST 2-WAY TV TECHNOLOGY**
Louis Martinez
Radio Telecom and Technology, Inc.
Riverside, CA

***EON-NET: A NOVEL INTERACTIVE VIDEO AND DATA SYSTEM**
Rafael Diaz
Eon Corporation
Reston, VA

THE SMART COMMERCIAL
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*Papers not available at the time of publication

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THE SMART COMMERCIAL

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Abstract

The rapidly approaching 500 channel world can seem rather frightening to an advertising world accustomed to targeting large network audiences with homogenous, universal campaigns. In this paper we show how the new 500 channel world brings with it the potential for advertisers to not only reach dispersed markets, but interact with, learn from, and strategically plan future product offerings. The SMART commercial will give advertisers the ability to create messages that will have the intelligence to navigate their way through the 500 channel environment and efficiently reach the new, segmented marketplace.

THE ADVENT OF THE DIGITAL HIGHWAY

With the introduction of the 500 channel world, the entire television industry will be turned upside-down. As the television industry changes, the advertising world will be forced to adapt. The methods in which advertisers reach their customers will shift to meet the demands of a more empowered audience. Although advertisers might be anxious about the encroaching, segmented 500 channel market place, we suggest that the new technology will provide an intelligent, efficient path to the consumer that has never before been possible.

The goal of every advertiser has always been to send a message about a product that will influence a consumer to purchase that product. In the network television scenario the process of reaching the target market has been to broadcast a commercial during a program that is viewed by those television households who would most likely be interested in the product, and would therefore be influenced by, the advertiser's message.

When the world was made up of three networks, the advertiser was pretty much guaranteed that the commercial would have a very good chance of being viewed by households that at least had their television sets on.

For most of television's history the three network system and the advertising community went virtually unchanged. Advertisers delivered their messages through mass market programs to mass audiences. Advertisers looked to certain programs with specific demographics to send their messages to viewers who might or might not be interested in the advertisers' products.

For the most part the advertisers' messages have reached the right audiences, but the process has been less than efficient. As a result of this strategy, often times advertisers have spent billions of dollars selling "Rice-A-Roni" to people who don't have pots or pans in their homes, and mini-vans to rich, affluent bachelors. This world has created wasted advertising dollars.

And to increase insult to injury, through the remote control, television viewers have "zapped" through channels, bypassing the billion dollar messages in search of the "perfect" program. This has resulted in many viewers missing commercials they are interested in, and viewing commercials for products of which they have no interest. With the 500 channel world approaching, the ability to find the target audience will become increasingly complicated. The introduction of the new, expanded world necessitates a more intelligent mechanism for advertisers to find their audience.

The digital infrastructure of 1's and 0's and fiber optics brings the intelligence of computers to the world of video. This infrastructure has inherent intelligence within it to help advertisers reach their intended market segment. With the digital information that will travel with the digital video and audio bits, commercials will be able to have an intelligence of their own. The Digital 90's brings will force advertisers to take advantage of digitization to implement a smarter way to distribute commercials.

THE SMART COMMERCIAL

The "SMART" commercial will help advertisers reach the new dispersed audience. It will revolutionize how advertisers interact with their audience, and how the audience will interact with advertisers. It will lead to a smart way for advertisers to find their target markets, to purchase advertising time, to design campaigns, and to facilitate the process of getting the products to the consumers.

The difference between today's commercial and the SMART commercial is one of approach. In today's commercial environment the advertiser delivers his message to his audience in an automobile without a steering wheel. The advertiser makes an educated guess as to who his audience is, and as to what his audience is interested in learning about and/or purchasing. The broadcasting advertiser has no way to tailor his message to subsets of his mass audience, nor to send subsets of his audience a different message. In essence, the advertiser pushes his products toward a market on a highway that has one off-ramp.

The proper approach to reach an audience is for the advertiser to respond intelligently to the interests of viewers who can communicate their desires, their interests, and perhaps their past, current, and future purchasing decisions. In order to be effective, the advertiser should respond to the pull of the viewer for viewer-specific information. The solution to match the advertiser to the customer is to develop an intelligent automobile with a map and a set of directions to exit many off-ramps on the digital highway. The solution is the SMART commercial.

Let us begin with the simplest example of the SMART commercial. The majority of households in the United States receive broadcast television programs through a cable. In many of these households resides an addressable analog set-top converter. The cable operator typically uses the addressability feature as a way to offer pay-per-view movies and events.

In the near future the addressable analog box will be replaced by an addressable digital set-top box. The digital set-top box will feature an electronic program guide that will contain many computer-like features. One of the most important features for advertisers is that the box will have the ability to take advantage of a memory module that will record the particular tastes of each household or of each member of the household.

This memory module will enable the set-top box to work like a computer. Just as a computer can be programmed to handle multiple tasks, so will the set-top box have the capability to record the characteristics, demographics, and traits of the household. These traits can be utilized to filter in those programs and commercials that would be of interest to the members of the household. In addition, the memory module will record advertising information that can be contained within the SMART commercial itself.

The digital set-top box will have the capability to learn the habits of the household and establish parameters to filter in programming preferences, and commercial preferences, when the television is engaged. For instance, if Household A frequently watches murder mysteries, the software inside the digital set-top box, the Electronic Program Guide (EPG), will suggest to the household that they turn to a particular program that is about to start which involves a murder mystery.

In the same light that the EPG can prompt households to a certain channel by way of remembering its preferences, this intelligence can be utilized to form the basis of directing SMART commercials to the proper households. Just as the EPG "learns" which programs Household A typically enjoys watching, the EPG will just as easily find out which commercials Household A typically enjoys viewing. If Household A switches channels or registers disinterest by pressing a "not interested" button every time a commercial about a sedan is "pushed" through the digital highway, the EPG will not "pull" commercials featuring sedans into Household A. If, on the other hand, Household A always watches commercials about light trucks or indicates interest in these commercials, the EPG will "pull" light truck commercials into Household A. This feature can be extremely profitable for companies like General Motors who offer both products.

Instead of spending money on a mass campaign about new GM sedans to all of America, without any hope of Household A's ever purchasing a GM sedan, GM could send a commercial about the new GM truck to Household A's, and send the commercial about the sedan to households who have expressed interest in looking at and purchasing sedans.

Although this infrastructure sounds complex and highly sophisticated, it is quite simple. Today's viewer turns on his television and receives one analog channel per 6 MHz of bandwidth. With digital compression, programmers digitize, compress and multiplex two to ten programs together and transmit the programs utilizing the same 6 MHz bandwidth. Digital compression provides the capability of placing several streams of programs and commercials within a 6 MHz multiplex. The beauty of the digitization process is that it allows digital information to become imbedded within the video to give the SMART commercial the "intelligence" it needs to efficiently navigate its way along the digital highway.

THE SMART COMMERCIAL CONCEPT

The SMART commercial is an entity whose goal is to maximize the efficiency of the digital architecture that will shortly be available. It will seek out those homes in which it can make the greatest impact in influencing the audience that will most likely make a purchase decision, and bypass those homes in which its influence is deemed to be ineffective. It utilizes the digitization process to store navigational data that will aid itself in seeking out consumers who will most likely be effected by its message.

Let us follow the production process of the SMART commercial.

(1) Video Generation

The advertising team creates a video, film or graphic representation of the advertisement for the product. The commercial is digitized. The digitization process includes the addition of blanks that will be filled with intelligence as the commercial moves from production to distribution.

(2) Market Segmentation

The advertising team logs the basic characteristics of the target audience for the product. Characteristics include traits such as sex, age, buying patterns, levels of affluence, etc.

(3) Creating Intelligence

Applying the intelligence layer involves placing the characteristics that were logged in the Market Segmentation process into the blanks that were created in the Video Generation process. The advertising team imbeds these digital navigational characteristics into the commercial. In addition, hidden data and/or video applications are merged with the primary video and audio bits. This additional information will allow consumers to store the commercial, interact with the commercial, or send back a request for more information.

Given the ability to imbed digital data, advertisers can create several different versions of the same commercial. The advertiser can imbed several language tracks that will play depending upon which household pulls in the commercial. Several layers of the commercial can be interchanged so that a commercial could display different phone numbers, features, incentives, etc. depending upon the particular audience. In fact, the video portion of the commercial might have many versions of different actors of various ages communicating the same message, yet the actors or message received will depend upon the characteristics of the household audience.

(4) Distribution

The commercial is sent via satellite, cable or microwave directly to homes that have the ability to decompress the digital signals or to cable operators that distribute the commercial to its customers. As it travels along its path to the home it has the ability to pick up additional navigational characteristics. If the commercial travels to a community that has particular segments that are interested in particular goods or have particular purchasing habits, that information can be imbedded into the commercial as it travels through the community. If the community members have the ability to request specific information, this request can also be "tagged" to the commercial as it makes its way through the community

SMART COMMERCIAL RECEPTION

Once the SMART commercial is created and sent through the digital highway, it will look for the reception sites for which it has intelligently been programmed to search. A digital set-top box will reside at the end points of the branches of the digital infrastructure on which the SMART commercial travels. The digital set-top box will identify information that is traveling on the digital highway, and will select the information that it requires. In its simplest form the digital set-top box will store the characteristics of the household or an individual in the household. When the commercial traveling through the highway identifies a household that fits its characteristics, it will enter the household and will live out its many stages of programmed lives.

THE MULTIFACETED LIVES OF THE SMART COMMERCIAL

(1) The simple life SMART commercial

The simplest form of the SMART commercial is one that mirrors today's commercials. Its life span lasts as long as it airs. However, it contains the basic customer characteristics that are used to direct the commercial to its destination point. During a particular avail during a particular program an advertiser might choose to render several versions of a commercial for a certain product, or several different commercials for different products.

Let us take the sedan/truck commercial. If the market can be segmented into two parts, a sedan household and a truck household, the advertiser will imbed vehicle segmentation information into the two commercials. The two commercials will be multiplexed together and will play at the same time during a particular avail. The "truck" home will receive information about a product that they will potentially purchase as opposed to viewing a sedan commercial that they will be disinterested in. As a result, the car manufacturer will have spent his advertising dollars more efficiently, since the avail price will now provide two distinct customers with product information that will more likely influence a purchase.

(2) The endless life SMART commercial

Given the digitization of video and a memory module within the digital set-top box, the SMART commercial has the ability to carry and download additional electronic information to the customer. When the SMART commercial enters the Creating Intelligence stage, the advertiser can imbed pages of electronic information. This electronic information can be thought of as an electronic brochure. As the customer views the commercial he can direct the SMART commercial to download the electronic information. As a result the customer will store product information that he or she can procure at a later time.

The advertiser truly benefits in this scenario, since the advertiser has not only provided customer-defined product information, but has provided an endless life commercial. The advertiser purchases a 15 or 30 second spot, but has created a commercial that can be accessed "endlessly" by the customer. The electronic information that the customer will access will include customer-defined data, detailed product specifications, retail locations that offer the product, phone numbers to call for more information, and ordering instructions.

(3) The interactive life SMART commercial

During the Creating Intelligence stage of the SMART commercial the advertiser can add the ability for the customer to respond to the commercial. In a fiber infrastructure the customer will be able to respond to the SMART commercial by signaling interest in the product, disinterest in the product, or with a more sophisticated digital set-top box, answer questions, ask for more information, or contact a service representative. The response data will be stored either in the digital set-top box to be downloaded at a later time, will be sent to a central location, or will be sent directly to the manufacturer or distributor.

This response data is invaluable to advertisers and manufacturers. The data provides immediate feedback for advertisers in how effective their commercial is, which type of audiences are responding to the message, and can provide the manufacturer with an estimate of his approaching sales. Such data can also help advertisers position future products and design future commercials.

An important feature for advertisers, given an interactive commercial, is that advertisers will be able to track the exact numbers of homes that had their television sets on during the showing of the commercial. Advertisers can imbed within the SMART commercial a mechanism to report back the households that received the commercial when it aired. Advertisers could more accurately monitor the effectiveness of the SMART commercial to more efficiently budget the next media campaign.

Additional interactive features will allow consumers to "trial" the product before they make a purchase decision. All of the interactive traits of the commercial can be added at the Creating Intelligence stage or added later. Some examples of "trial" capabilities include storing within the digital set-top box a few seconds of a CD, a few pages of a new book, a minute or two of a new film, interviews with customers who have purchased the product, a shortened version of a software product, pictures of the product in its many applications, etc. The interactive SMART commercial will not only reach the proper demographic segment, it will better inform the segment population about the product so that the customer will be more inclined to make the purchase.

IMPLEMENTATION OF THE SMART COMMERCIAL

The digital infrastructure that will make the SMART commercial possible is rapidly being deployed by large players in the cable and telephone industries. Fiber is quickly replacing copper wiring and twisted pair cable. Advances in digital compression have already proven that digitally compressed, transmitted pictures are many times preferable to analog transmissions.

As the infrastructure is built it is imperative that broadcasters and advertisers understand the changes and implement intelligent procedures to apply intelligence in the advertising process. As the computer, telephone, and television industries merge, it is crucial for advertisers to develop new ways to reach the new, segmented population. Utilizing the SMART commercial will be the only way to reach these scattered audiences.

Broadcast engineers must begin to learn about the approaching environment, understand the digitization process and plan for the future. In order for all advertisers to reach all markets, the SMART commercial must be designed so that it can travel without obstructions to all locations. Such a highway will demand standardization of the advertising creation and distribution process. Careful planning now will guarantee that advertisers of network programming will remain at the wheel in determining the destinations of their new, precious cargo - the SMART commercial.

CONCLUSION

This paper has touched upon the many applications that digital compression and the fiber infrastructure will bring to the advertising community. Advertisers need not fear the 500 channel world. Given an intelligent highway and a procedure to lock intelligence within a commercial, advertisers can look forward to a time when they can efficiently design, spend, and target SMART commercials that will better meet the demands of an empowered, segmented audience.

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DIGITAL VIDEOTAPE WORKSHOP

Thursday, March 24, 1994

Moderator:
Richard Streeter, CBS Inc., New York NY

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*Papers not available at the time of publication

RADIO TRANSMITTER MAINTENANCE WORKSHOP

Thursday, March 24, 1994

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Fairburn, GA

*Papers not available at the time of publication

CAMERA WORKSHOP

Thursday, March 24, 1994

Moderator:
Kelly Williams, NAB, Washington, DC

525-LINE PROGRESSIVE SCANNING TV CAMERA WITH 16:9 ASPECT
Akihiro Hori
Nippon Television Network
Tokyo,

***TIPS AND TECHNIQUES FOR OPTIMIZING CAMERA PERFORMANCE**
Fred Himelfarb
Woodmere, NY

*Papers not available at the time of publication

525-LINE PROGRESSIVE SCANNING TV CAMERA WITH 16:9 ASPECT

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Abstract

This Paper discusses a 16:9 aspect ratio, 525-line, progressive scanning television camera. This camera employs a newly developed Multiple Frame-Interline-Transfer (M-FIT) CCD and employs digital signal processing technologies. Although picture quality is high, the cost of this camera is not far different from conventional interlace cameras. This paper deals with the M-FIT CCD, the digital signal processing, the transmission of progressive signals, and applications that make good use of the high quality progressive signals.

1. Introduction

We have developed a 16:9 aspect ratio, 525-line, progressive scanning television camera. This is the world first CCD progressive scan studio camera with solid state image sensor. This camera will be used by the EDTV-II system in Japan.

The major challenges during a design of this camera were the design of the actual progressive scan image sensor, high speed digital signal processing, and high transmission rate between the camera head and a base station. The newly developed progressive scan 2/3" CCD with 520k pixels uses a Multiple Frame-Interline-Transfer (M-FIT) structure. Further this camera employs a "parallel signal processing" technology to achieve an overall lower processing speed and in addition to that it employs a 4:2:2:4 transmission technology to achieve a lower transmission rate as well.

This paper describes the new M-FIT CCD, the digital signal processing, and the transmission system. Applications that make good use of the high quality progressive signal are described as well.

2. Progressive Camera System Structure

Fig2.1 shows the progressive scanning television camera system. This system consists of two major components; a camera head and a base station. And a hybrid fiber optical / wire cable connects the camera head with the base station. The cable has two single mode fibers and four wires. The two fibers are used for digital signal transmission where one fiber is used for the transmission from the camera head to the base station and the other for the transmission of signals and control from the base station to the camera head. One pair of wires provides AC power supply from the base station to the camera head and the other lines is for emergency intercommunication.

The following paragraph outlines the camera head structure and the base station structure.

2.1 Camera Head

Fig2.2 shows the block diagram of the camera head. The M-FIT CCD will be discussed in the chapter 3.

The ANALOG PROCESS block is a critical element for digital processing in the camera to assure the highest digital accuracy and this processing consist of:
(1) Pre-amplification - of the signal in the front end of the camera
(2) Signal addition process - which provides black shading correction, flare correction, pedestal
(3) Gain control - consisting of white balance, manual gain, white shading correction
(4) Non-linear-processing/compression - sometimes called pre-gamma
The signal is converted from an analog domain into digital in the A/Ds which are quantizing the signal with 10-bits accuracy and 36 MHz sampling frequency.

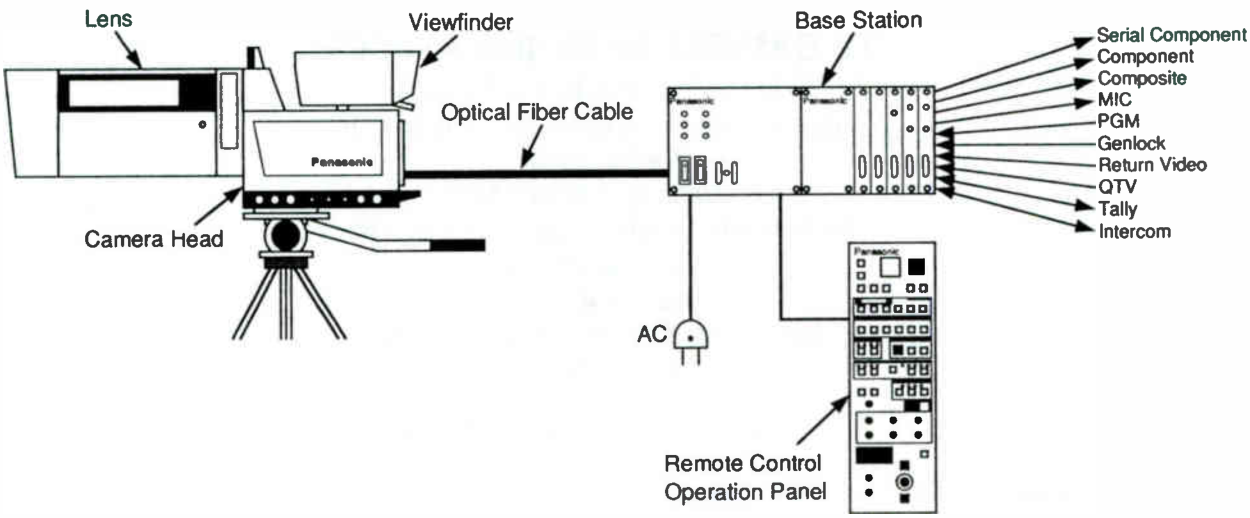

Fig 2.1 Progressive Scanning Television Camera System

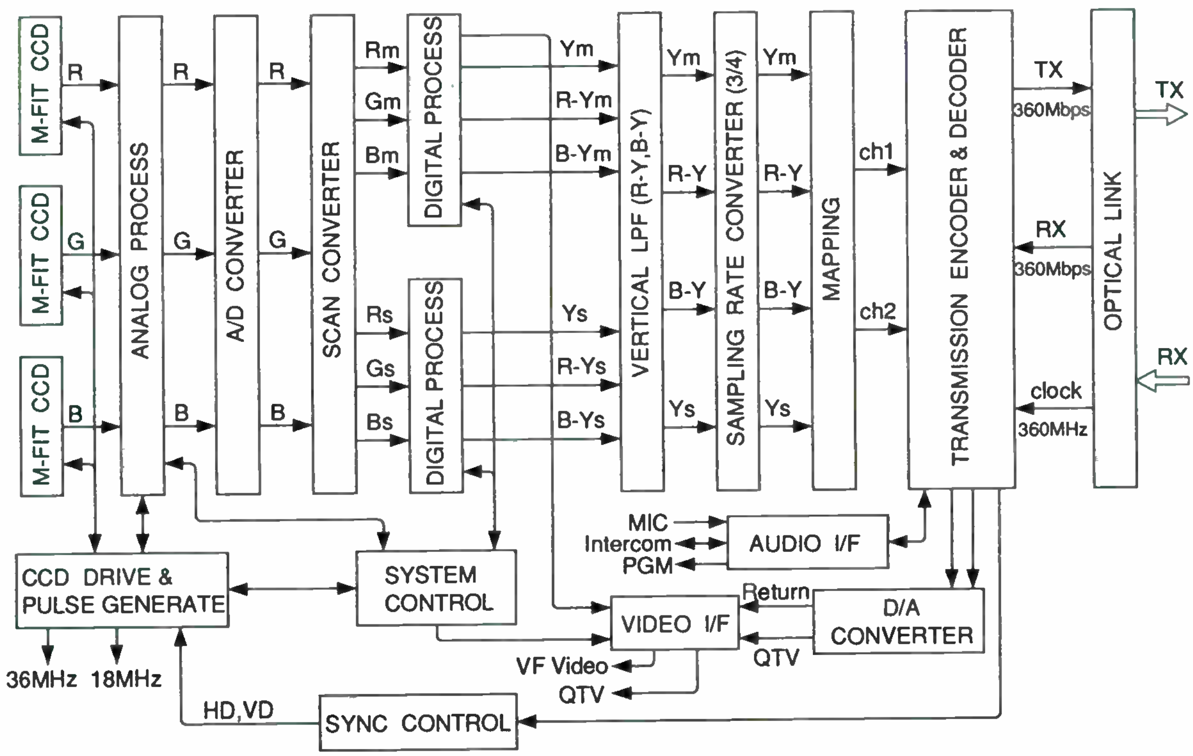

Fig 2.2 Block Diagram of Camera Head

In the SCAN CONVERT block, six channels of an interlaced signal are created. Each channel of the R, G, B signal is converted into two channels of an interlaced signal Xm and Xs. The camera head further contains two DIGITAL PROCESS blocks which both have an identical digital processing. These digital processes are as follows:

(1) Gamma auto-knee, calculating the average value of the small area's image signal
(2) Pixel shifting, masking, matrix, enhancer, high chroma saturation aperture

Concerning the DIGITAL PROCESS outputs, the sampling rate of the luminance signals (Ym & Ys) is 18 MHz and the sampling rate of the color difference signals (R-Ym, B-Ym, R-Ys, B-Ys) is 9 MHz.

The VERTICAL Low Pass Filter block is utilized by the color difference signals. In this block four channels of color difference interlaced signals (R-Ym, B-Ym, R-Ys, B-Ys) are converted into two channels (R-Y, B-Y) with a help of vertical low pass filters. This means that the progressive color difference signals are converted into interlaced signals.

In the SAMPLING RATE CONVERT block the luminance signals (Ym, Ys) are converted into 13.5 MHz sampling rate and the color difference signals (R-Y, B-Y) are converted into 6.75 MHz sampling rate. These converted signals are of 4:2:2:4 sampling structure and they are in a parallel format.

Further these 4:2:2:4 parallel signals are encoded for serial transmission from the camera head to the base station.

2.2 Base Station

Fig2.3 shows the block diagram of the base station. In the base station the received serial data signal from the camera head is converted back into 4:2:2:4 parallel signal format. Additionally these signals are further converted into 8:4:4 parallel signal format. The luminance signal (Y) of the 8:4:4 stream is 27 MHz sampling rate and the color difference (R-Y, B-Y) signals of the 8:4:4 data stream have a 13.5 MHz sampling format. These progressive signals are later converted into analog signals in a D/A converter for analog outputs. For serial outputs the signal is appropriately encoded.

3. M-FIT CCD

3.1 CCD Structure

In a schematic diagram Fig3.1 it is shown the structure of the new M-FIT CCD. It consists of an image area, a storage area, a single channel H-CCD, and an output buffer amplifier. The aspect ratio of the CCD is 16:9 and the active video pixel numbers are 948 (H) x 485 (V), and the actual pixel size is 11 (H) x 10 (V) μm^2.

The image area consists of a number of unit cells. Each cell is composed of a single photo-diode coupled with two gates of V-CCD, similarly as in a conventional FIT CCD configuration. The V-CCD in the image area and in the storage area consist of 243 transfer stages and 485 transfer stages, respectively. Due to that fact, the storage area is capable of storing all of signal charges from the image area and this is the basic difference as compared to conventional interlaced scan CCD design.

3.2 Flow of Image Sensor

The basic functional operation is achieved by the multiple transfer of a charge from photo-sensitive diodes by an effective use of time sharing during blanking interval. The multiple transfer operation is schematically shown in Fig3.2.

Signal charge from the "odd" photo-diodes is first shifted into the V-CCD register located at the upper section of the storage area. The signal charge from the "even": photo-diodes is transferred to the storage area by "pushing" forward the previous data. After that processing step, the signal charges in the storage area are shifted into the H-CCD and sequentially transferred to the output stage. This operating sequence is shown on the timing diagram of Fig3.3.

This multiple transfer makes it possible to increase the dynamic range almost by factor two as compared to the conventional single transfer CCD. Reason for that lies in the area of the V-CCD which in case of the multiple transfer increases with the number of transfer time. The charge handling capability of the M-FIT CCD is schematically shown in Fig3.4 as compared to the single transfer case. From the picture is obvious that the increase of the charge handling capability of the V-CCD results in remarkable increase of the dynamic range for the M-FIT type CCD.

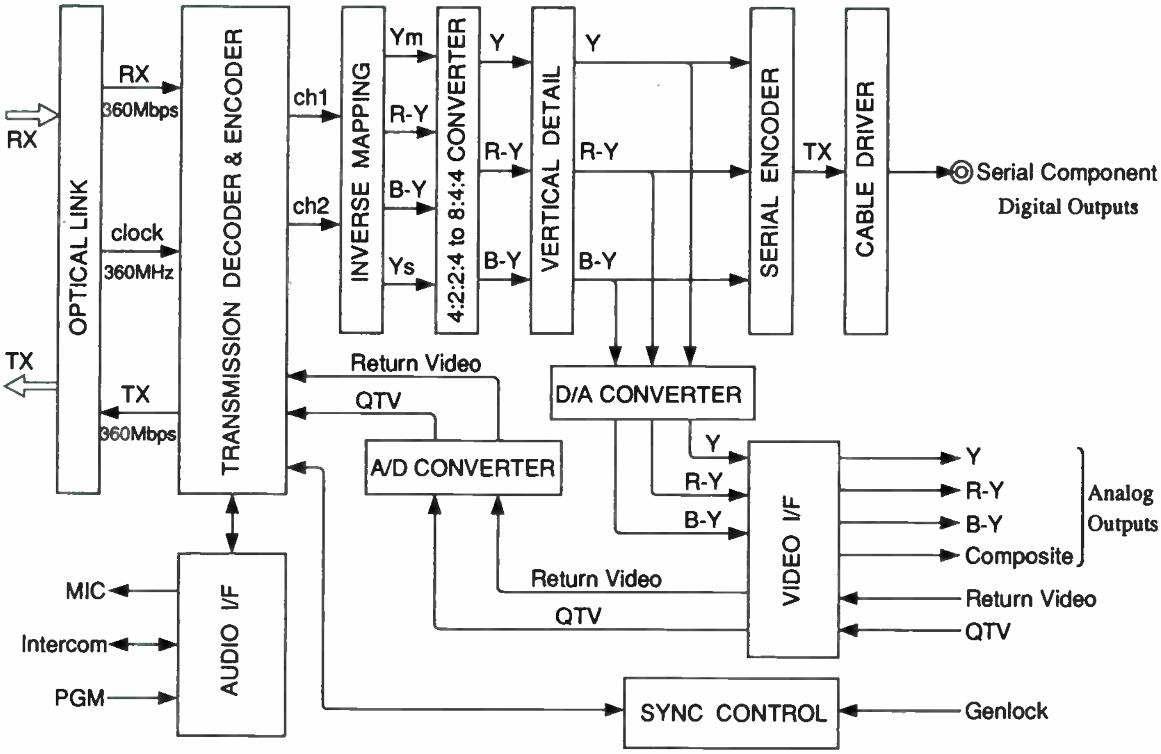

Fig 2.3 Block Diagram of Base Station

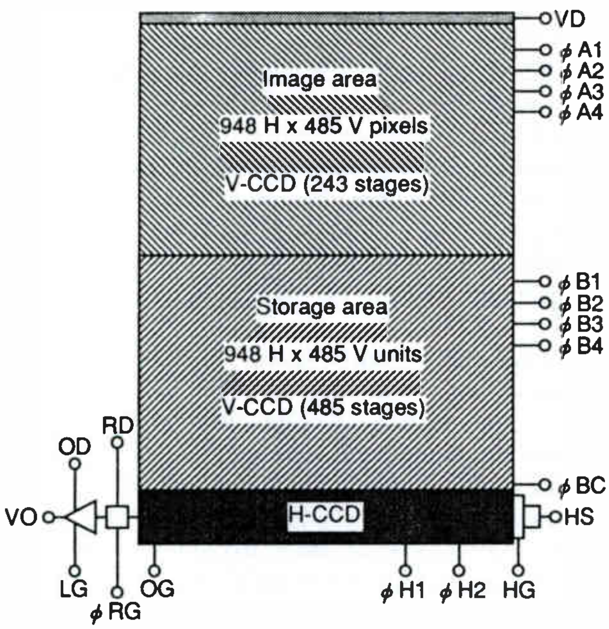

Fig3.1 CCD Schematic Diagraam

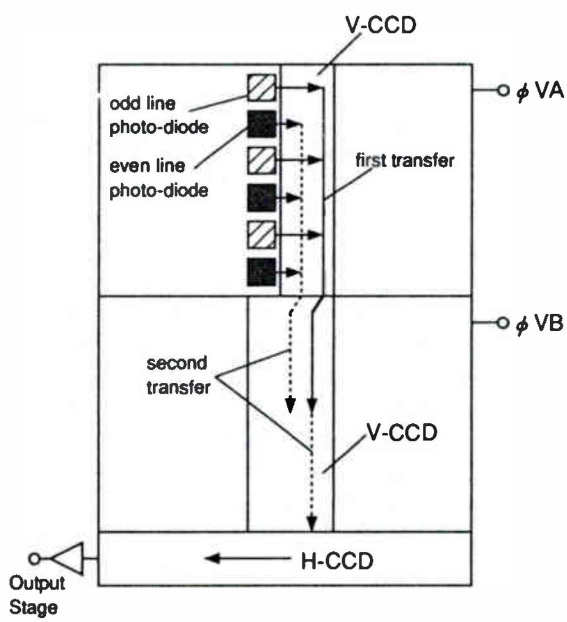

Fig3.2 Operation of M-FIT

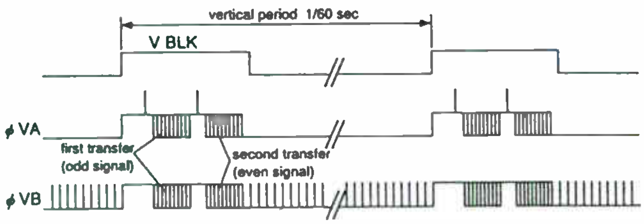

Fig3.3 Timing Diagram

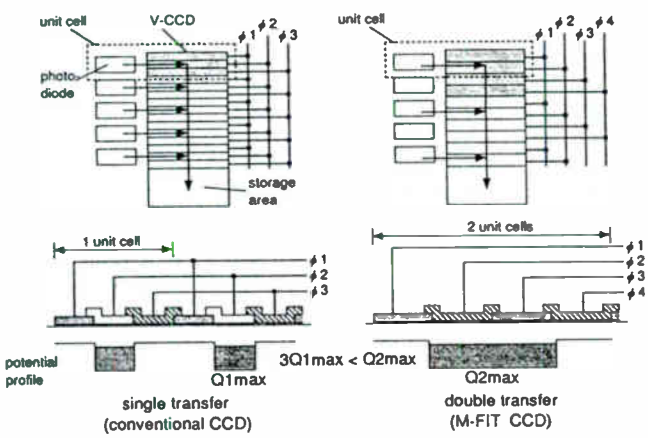

Fig3.4 Charge Handling Capability

It is also noted as well that the multiple transfer configuration can be implemented utilizing the conventional 2-poly-Si process without increasing the process complexity.

3.3 CCD Characteristics

The photo-conversion characteristics measured are shown in Fig3.5 together with those for a conventional progressive scan CCD imager. The graph indicates that the sensitivity reaches as high as 25nA/lx and the dynamic range is as high as 80dB. It is also noted that an on-chip micro lens fabrication process is used as well, to increase further the light sensitivity. The sensitivity of the complete device with micro lenses is 2.5 times higher than that of a device without micro lenses. Therefore the dynamic range is almost twice as large as that of a conventional device owing it to the M-FIT operation. The vertical resolution of 480 TV line is obtained. The typical characteristics of the device are summarized in the Table 3.1.

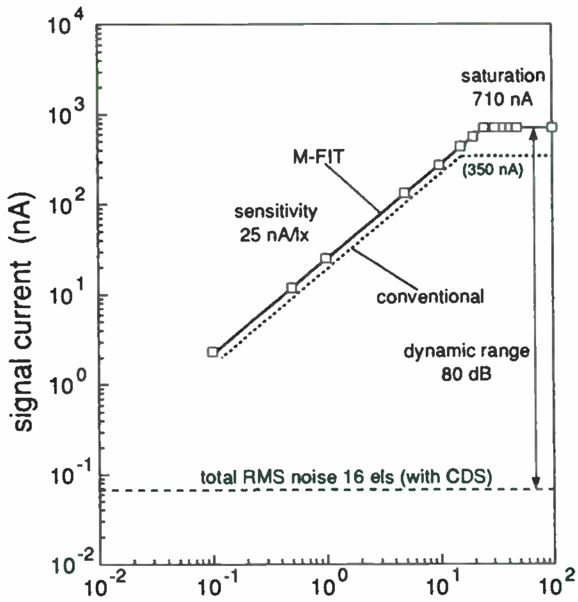

Fig3.5 Photoconversion Characteristics

Table 3.1 CCD Characterisitics

number of pixels	948 (H) x 485 (V)	
image region	9.48 (H) x 5.335 (V)	mm
pixel size	10 (H) x 11 (V)	μm
saturation current	710	nA
sensitivity	25	nA/lx
dynamic range	80	dB
smear level	-125	dB
image lag	undetectable	

4. Video Signal Processing

4.1 Signal Arranging with Memory

An M-FIT CCD output signal of one of the field images is shown in Fig4.1 (center figure). According to chapter 3, the output signal of an M-FIT CCD is not of a progressive sequence. At first M-FIT CCD outputs the odd lines, and then outputs the even lines of the image. It is possible to arrange these lines in a progressive sequence by using field memories. Due to fact that this processing is in a "parallel digital domain" two channels of interlaced signal are created. During an even field, the odd lines will from a main channel and the even lines will become a sub channel. During an odd field, the odd lines will be sub channel and the even lines will be the main channel. This process arrangement is achieved in the SCAN CONVERT block of the camera head.

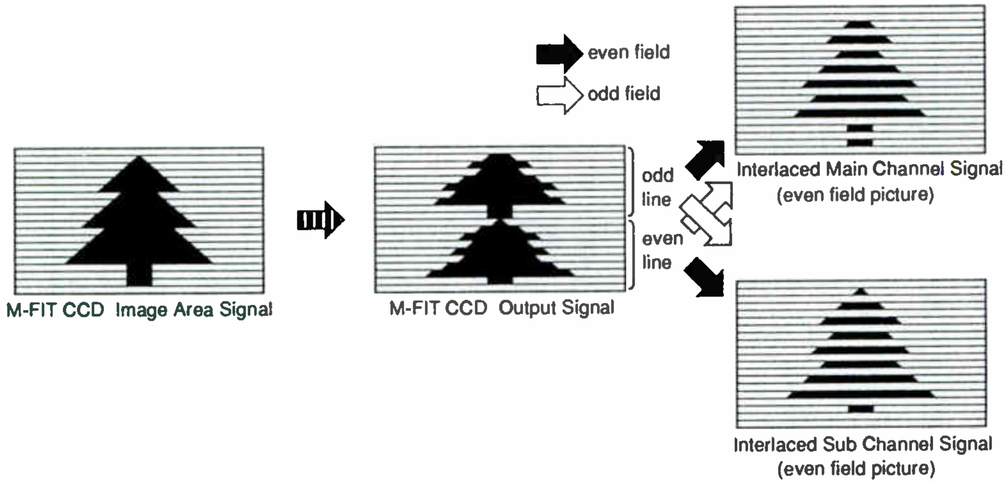

Fig4.1 Making Two Channels of an Interlaced Signal

4.2 "Parallel Digital Signal Processing"

The main channel interlaced signal is processed the same way as the sub channel in the DIGITAL PROCESS block in the camera head. There is no difference between the two channel processing on account of the digital processing.

"Parallel digital processing" has a very good benefits because it lowers the processing clock rate. The master clock frequency of the DIGITAL PROCESS block is 36 MHz. If the progressive camera would employ a single digital processing method, a 72 MHz clock would be needed. Therefore this "parallel digital processing" capability allows the use of LSIs which were developed for conventional interlace cameras.

5. Transmission within the Progressive Camera System

5.1 Digital Interface of a 525 Line Progressive Scan System

The 525 line progressive scan signal contains twice as much information as a conventional 525 line interlaced system. Therefore (4:2:2) x 2 = 8:4:4 would be the choice for an interface. Subjective assessment of the 8:4:4 and 4:2:2:4 system were made using a 5 grade Double-Stimulus Quality-Scale Method. The result of the evaluation is shown in Fig5.1. It indicates that there is very slight difference in image quality between the 8:4:4 and 4:2:2:4 structure. Therefore the 4:2:2:4 signal structure will be more appropriate for our application because the data rate is lower than for an 8:4:4 type signal.

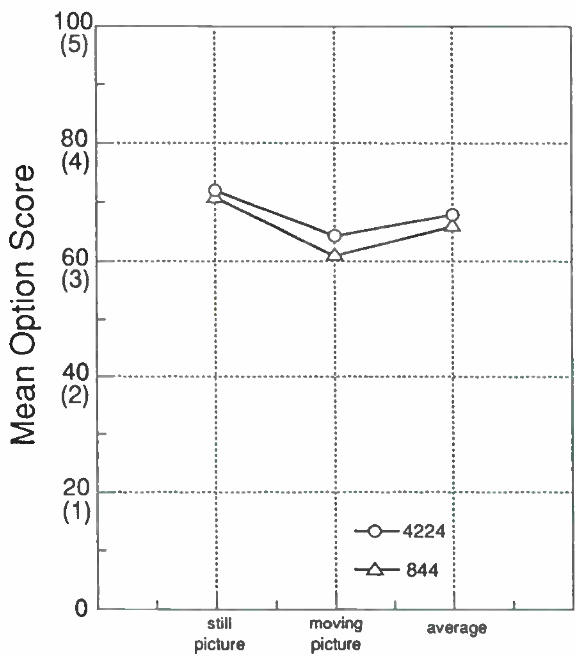

Fig5.1 Subjective Assessment Result

5.2 Encoding 4:2:2:4 Signals

The 4:2:2:4 structure consists of four channels. These are main channel luminance (Ym; 10 bits, 13.5 MHz sampling), sub channel luminance (Ys; 10 bits, 13.5 MHz sampling), and two channels of color difference signal (R-Y, B-Y; 10 bits, 6.75 MHz sampling). From that it is clear the 4:2:2:4 data rate is equal to 405 Mbps. These signals are shown in Fig5.2 and they are converted into two channel data streams in the MAPPING block. Timing-Reference-Signal and

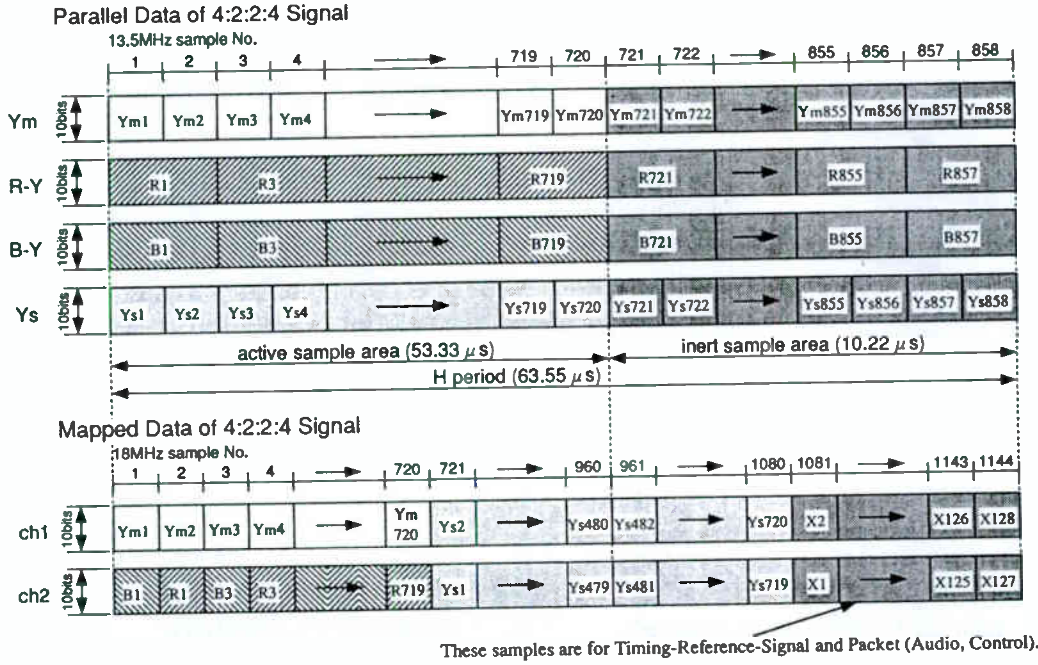

Fig5.2 Mapped Data of 4:2:2:4 Signal

other packets (audio and control) are inserted into the resulting MAPPING output signals. The actual serial data is generated in the TRANSMISSION ENCODER & DECODER block. The resulting serial data rate is 360 Mbps. The data rate conversion is achieved by extensive use of mapping techniques.

5.3 Optical Transmission

Camera system uses two optical links for bi-directional serial data transmission. These optical links provides high quality and transmission over a long distance. The characteristics of the optical fiber link are summarized in the table5.1. The main advantages of the optical transmission system are here briefly outlined:

(1) High quality signal transmission:
Use of the fiber optical cable results in very low signal degradation and is free from any interference be it from RFI or other sources. The fiber optical serial transmission helps to avoid a signal loss during the transmission and conversion that is normally inherent in conventional analog systems. Pictures that are transmitted over the fiber optic link are received at the base station with full quality and free of signal distortions.

(2) Long distance transmission:
The optical fiber cable can be used up to 20 km (12.4 miles) if a local power is supplied to the camera head, or up to 2.4 km (1.5 miles) with power supplied through the connecting cable. The long cable capability means greater operating flexibility both in the studio and in the field.

Table 5.1 Specifications of the Transmission

	camera head → base station	base station → camera head
Transmission Bit Rate	360 Mbps	
Transmission Mode Coding	Scrambled NRZI	
Transmission Line	Single Mode Optical Fiber	
Light Source	FP-LD InGaAsP 1310 nm	
Photo Detector	Pin-PD InGaAsP	
Max. Transmission Distance	20 km (12.4 miles)	
Video Signals		
Number of Channels	4 (4:2:2:4 component)	2 (composite)
Signal Band Width	Ym,Ys : 6MHz, R-Y,B-Y : 3MHz	6 MHz
Sampling Frequency	Ym,Ys : 13.5MHz, R-Y,B-Y : 6.75MHz	18 MHz
Quantization	10 bits/sample	10 bits/sample
Audio Signals		
Number of Channels	4	3
Signal Band Width	20 kHz	20 kHz
Sampling Frequency	48 kHz	48 kHz
Quantization	16 bits/sample	16 bits/sample

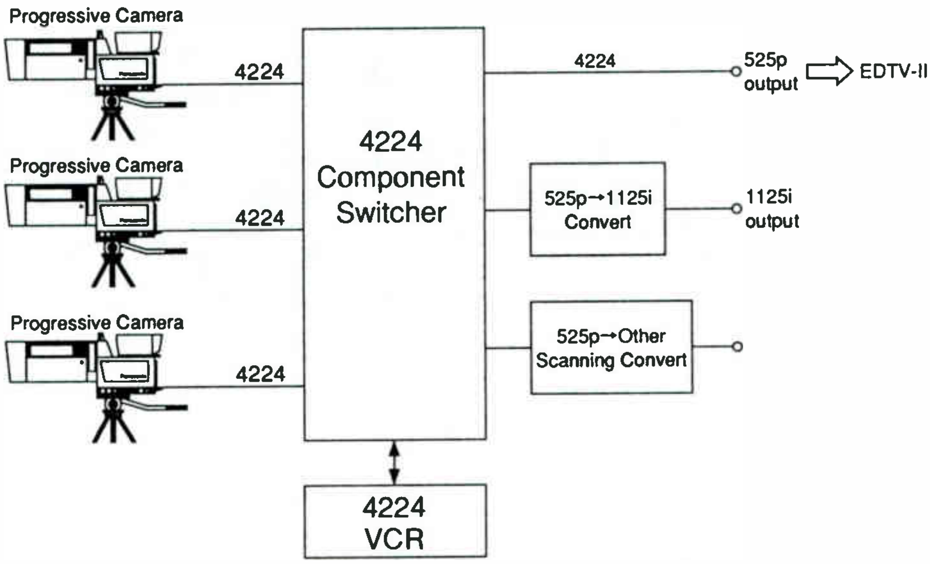

Fig 6.1 525 Progressive Scan Production System

6. Applications

6.1 Progressive Scan Production System

An example of a 525 line progressive scan production system is shown in Fig6.1. A 525 line progressive scan switcher is achievable by connecting two conventional interlace component switchers with some additional modifications. As for VCR, the Lsystem which is available today in the market consist of parallel operation of D1 or D5 VCRs (D1/D5 x 2) or a use of compression system and one component VCR. It is expected that 4:2:2:4 VCR will be available soon. Scan conversion process of a 525 progressive system to the other scanning formats is discussed in a latter section.

6.2 The Advantages in Use a 525 Line Progressive Signal System

The 525 progressive scan system will be used by the EDTV-II system in Japan. Other advantages of use for a 525 progressive signal are as follows:

(1) Generating a high quality interlaced signal converted from a progressive signal:
Two neighboring line CCD output signals are added to obtain a 525 interlaced signal in a conventional CCD device. This is equivalent to a application of a 2-tap vertical filter. Reason for this process is to optimize vertical aliasing and vertical resolution. A higher quality interlaced signal can be obtained by applying an ideal filtering for a progressive type signal. This would provide better and higher vertical resolution and much less vertical aliasing than a conventional interlaced CCD camera.

(2) Scanning format conversion from 525 progressive signal to other scanning formats:
Most of the TV systems today employ an interlaced scanning format. Considering the scanning format conversion such as 525 interlace to 1125 interlace scan, the original picture is processed differently depending on the content of the picture. The processing differs in the part of the picture with motion as compared to the picture part without motion. Usually, not only one field of information is used in the conversion process, but previous field information is used as well in the case of still picture. In the case of moving picture, however, only that field information is used. Therefore motion detection is necessary for such a conversion process. This often a cause of picture quality degradation due to imperfection of the motion detection. Similarly a sudden change from a still picture with high resolution to a picture with motion and lower resolution is problematic as well. In a case of a scan format conversion from progressive signal, only the intrafield conversion is necessary. Although the conversion hardware is simpler, the converted picture still has a higher quality and natural look.

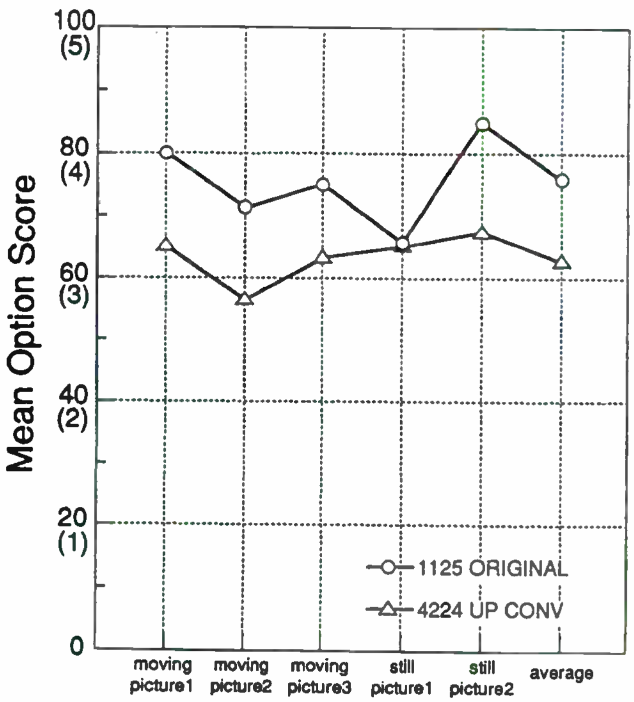

Fig6.2 Subjective Assessment Result

We have experimented with a scan conversion from 525 line progressive to 1125 interlace. Subjective assessment by a 5 grade Double-Stimulus Continuous Quality-Scale Method was again used in these tests. Fig6.2 shows the result. Despite the data rate difference between 4:2:2:4 (405 Mbps) and 1125 HD, (1485 Mbps) which is approximately 4 times as high, only grade 13 (0.65 on 5 grade scale) degradation was obtained. The selected pictures for these tests contained a high vertical frequency component. Usually a much smaller difference would be expected in ordinary pictures. A cost of the 525 line progressive system would be lower than HD system because the data rate is lower, therefore an HD production system using up-conversion from progressive is not only cost effective, but data rate effective as well.

7. Conclusion

A progressive scanning television camera was developed. The characteristics of this camera are summarized in Table7.1. It is the world first broadcasting camera with a CCD imager progressive scan output and very high picture quality. The cost of this camera, however is not far different from conventional interlace cameras. The reason for that are:

(1) Inexpensive M-FIT CCD
The M-FIT CCD is as inexpensive as a CCD which was developed for conventional interlace cameras. The multiple transfer configuration can be implemented by utilization of conventional 2-poly-Si process without an increase in the process complexity.

(2) Low speed digital processing
"Parallel Digital Signal Processing" makes use of existing C-MOS LSIs which were developed for interlace cameras.

(3) Low rate of transmission
4:2:2:4 transmission format and mapping technique lowers transmission rate.

References

1. K. Itakura et al., "A multiple Frame-Interline-Transfer CCD," ISSCC DIGEST OF TECHNICAL PAPER, pp 190-191, Feb. 1993

2. A. Hori et al., "Examination of a Progressive Component TV System with an Eye to Media Conversion," 18-th International Television Symposium, Switzerland, Symposium Record, Broadcast sessions, pp 792-802, June 1993

3. S. Nishikawa et al., "Broadcast-Quality TV Camera with Digital Signal Processor," SMPTE Journal, Vol. 99, pp 727-733, Sep. 1990

Table 7.1 Camera Specifications

method	16:9 aspect ratio, 1:1 progressive scan, 525 line / 59.94 Hz
pick-up devise	M-FIT 2/3" CCD image sensor x 3
optical system	F1.4 prism with quartz filter
sensitivity (89.9% reflection)	2,000 lx (at F8.0 electronic shutter off)
highlight compression	apporox. 600%
registration	less than 0.05% over entire screen (excluding lens)
analog outputs	Y, R-Y, B-Y / R, G, B
digital outputs	SMPTE 259M (270Mbps) x 2 ch

Grand Alliance HDTV System Specification

Draft Document

Submitted to the ACATS Technical Subgroup

February 22, 1994

Please note that this document, "Grand Alliance HDTV System Specification", is still in draft form. This document has been submitted to the Technical Sub-group of the FCC's Advisory Committee on Advanced Television Systems (ACATS) on February 22, 1994 and has not yet received final approval. This document will be subject to review by the Advisory Committee and the Grand Alliance, and will evolve over time. Any modifications to this specification document will be reviewed with the Advisory committee and reflected in the final document.

Grand Alliance
HDTV System Specification

Draft Document

Submitted to the ACATS
Technical Subgroup

February 22, 1994

Table of Contents

SCOPE OF DOCUMENT

This document details the System Specification of the Grand Alliance HDTV System, and is intended to form the basis for the documentation of the proposed standard. It is comprised of the documents that have evolved from the work of the Grand Alliance Specialist Groups with the guidance and cooperation of the Advisory Committee on Advanced Television Services (ACATS) Experts Groups. This system specification document also details the prototype hardware, currently under construction. This prototype implementation will be delivered to the Advanced Television Test Center (ATTC) for verification of the performance of the proposed GA HDTV system standard.

EXECUTIVE SUMMARY

LAYERED SYSTEM ARCHITECTURE WITH HEADER/DESCRIPTORS

The Grand Alliance HDTV System is a layered digital system architecture that uses headers/descriptors to provide flexible operating characteristics. The layers of the GA HDTV System, and some of their most important capabilities, are:

- the Picture layer provides multiple picture formats and frame rates
- the Compression layer uses MPEG-2 video compression and Dolby AC-3 audio compression
- the Transport layer is a packet format based on MPEG-2 transport, that provides the flexibility to deliver a wide variety of picture, sound and data services
- the Transmission layer is a Vestigial Sideband signal that delivers a data rate of over 18 Mbps in the 6 MHz simulcast channel

While all of the GA HDTV system's layers operate in unison as an effective simulcast system, each layer has also been designed to have outstanding interoperability. The GA HDTV system's layered digital system approach with header/descriptors will create interoperability among a wide variety of consumer electronics, telecommunications, and computing equipment.

PICTURE LAYER

The picture layer consists of raw pixel data, organized as pixels, scan lines and frames. The GA HDTV system provides for multiple formats and frame rates, *all of which will be decoded by any GA HDTV receiver*. This approach allows program producers and application developers to make their own tradeoffs among resolution, frame rate, compression artifacts and interlace artifacts, and to choose the format that is best for their particular use. The formats are:

Spatial Format (X x Y active pixels)	Temporal rate
1280 x 720 (square pixels)	23.976/24 Hz progressive scan
	29.97/30 Hz progressive scan
	59.94/60 Hz progressive scan
1920 x 1080 (square pixels)	23.976/24 Hz progressive scan
	29.97/30 Hz progressive scan
	59.94/60 Hz interlaced scan

COMPRESSION LAYER

The compression layer transforms the raw video and audio samples into a coded bit stream -- essentially a set of computer instructions and data that are executed by the receiver to recreate the pictures and sound. The compression layer of the Grand Alliance HDTV system:

- uses video compression syntax that conforms to the ISO-MPEG (International Standards Organization — Moving Picture Experts Group) MPEG-2 video data compression draft standard, at a nominal data rate of approximately 18.4 Mbps.
- uses Dolby AC-3 audio compression, at a nominal data rate of 384 kbps.

TRANSPORT LAYER

The transport layer separately packetizes video, audio and auxiliary data and allows their mix to vary dynamically, providing the flexibility needed to innovate new services and new kinds of programming. The transport layer of the Grand Alliance HDTV system:

- uses a packet format that is a subset of the MPEG-2 transport protocol.
- provides basic service capabilities that include video, five-channel surround-sound audio and an auxiliary data capacity.
- offers great flexibility in the mix of video, audio and data services that can be provided to appropriately-featured receivers. It separately packages each type of data (e.g., video, audio, etc.) in its own set of transmission packets. Each packet has a *Packet ID* header that identifies the content of the data stream. This capability enables the creation of new services, ranging from many stereo channels of audio, to broadcast distribution of computer software, to the transmission of very high resolution still images to computers.
- allows the mix of services to be dynamically allocated. This capability will allow rapid burst-mode addressing of receivers. It will also enable broadcasters to send multiple "streams" of video, audio and data programming to their audience, all complementing or enhancing the basic program content. This capability can fundamentally change the nature of television programming, since it enables software to be broadcast to "smart receivers" that can operate in conjunction with the HDTV picture and sound. With this capability, HDTV will likely become a more interactive medium than today's television, enabling new forms of educational and entertainment programming and games.
- provides important extensibility, since a Grand Alliance HDTV receiver will disregard any packet with a PID header that it does not recognize or cannot process. This will eliminate future "backward-compatibility" problems in the installed base of receivers, removing a crucial constraint from the introduction of new services.

TRANSMISSION LAYER

The transmission layer modulates a serial bit stream into a signal that can be transmitted over a 6 MHz analog channel. The transmission layer of the Grand Alliance HDTV system:

- uses a trellis-coded 8-VSB modulation technique to deliver approximately 18.8 Mbps in the 6 MHz terrestrial simulcast channel
- provides a pilot tone that facilitates rapid signal acquisition and increases pull-in range.
- provides a training signal that facilitates channel equalization for removing multipath distortion.
- provides a related 16-VSB modulation technique to deliver two 18.8 Mbps data streams in a 6 MHz cable television channel

SUMMARY

The GA HDTV system has the flexible operating characteristics that allow it to provide broad interoperability needed to form the basis for new and innovative services and applications of HDTV in many industries.

Chapter I

SYSTEM BACKGROUND

A successful HDTV standard will be used for HDTV delivery to the public by terrestrial broadcasting and a variety of alternate delivery media, and it will also form the basis for new services and new applications of HDTV across a wide range of industries. In order for the enormous potential impact of digital HDTV to be realized, a high degree of interoperability is an essential attribute of an American standard.

1.1 INTRODUCTION

The formation of the Grand Alliance moves the HDTV standardization process from a competitive phase into a new collaborative phase. The previous competitive phase stimulated the rapid development of technology, as well as broad industry debate over various attributes in which the proposed systems differed. In the new collaborative phase, the Grand Alliance and the Advisory Committee must rapidly establish a national consensus and move forward into laboratory and field performance verification, and the documentation of a standard for FCC approval. Failure to achieve these objectives will harm every industry that is potentially advantaged by the deployment of HDTV systems and technology.

In some cases where the Grand Alliance confronted the dilemma of conflicting goals, it was technically feasible and reasonably cost-effective to allow a multiplicity of modes within the system. In these cases, the presence of modes favored by a particular industry do not preclude the inclusion of modes that are more favorably viewed by a different industry. The result is a system that is maximally useful to all industries, rather than one that burdens one industry at the expense of another.

In other cases where the Grand Alliance confronted the dilemma of conflicting goals, a choice had to be made among the conflicting alternatives. In these cases, there are good technical reasons for the choice that was made, and careful consideration has been given to interface requirements that bridge the differences between the approaches of different industries and make the system interoperable.

1.2 SCOPE OF HDTV STANDARDS AND THEIR IMPACT

HDTV technology will not only revolutionize the television industry, but it will have a profound impact across a wide range of industries. The proliferation of high speed digital communications needed to deliver HDTV to the home will make the associated industries a key part of our National Information Infrastructure. Further, the core capability of cost-effectively delivering full-motion high resolution pictures and high-quality surround-sound will become a cornerstone of image communications that may be adopted by a variety of industries and applications.

However, it is important to understand the scope of the GA HDTV System, which is a proposed transmission standard that is under consideration by the FCC through its Advisory Committee on Advanced Television Service (ACATS). The Grand Alliance HDTV system (and presumably the FCC standard) is NOT a production standard, or a display standard, or a consumer equipment interface standard. In fact, a digital compression and transmission system effectively DECOUPLES these various elements of the end-to-end imaging chain.

A digital HDTV transmission standard must specify sufficient information that allows any manufacturer to build a receiver that can receive a radio frequency (RF) transmission, and from it, produce pictures and sound. For a digital system, this means specifying the meaning of a bit stream and the signal format for transmitting those bits in a 6 MHz television channel. Thus, the picture formats, the compression syntax, the packet format and the RF signal format are examples of items that require standardization. While these standardized elements establish a minimum functionality for HDTV encoders and receivers, they do not specify encoder or receiver implementation. In the opinion of the GA, this traditional approach to transmission standards is highly desirable, because it enables natural competitive and economic forces to influence the implementation of encoders and receivers, which can evolve over time as underlying technology as cost changes take place.

1.3 GRAND ALLIANCE GOALS

Since we are considering a proposed HDTV transmission standard that contains many complex tradeoffs, it is important to state the design goals that the GA established for itself. These goals essentially summarize the selection criteria that were established by ACATS during the competitive phase of the standards process. The GA design goals are to provide:

- High quality HDTV pictures and sound
- An effective simulcast system that:
  - Provides wide HDTV service area for terrestrial broadcast

- Avoids unacceptable interference to existing NTSC service during its lifespan
- A cost effective solution for consumers, producers and all users at the time of introduction and over the life of the HDTV standard
- Interoperability with other media (e.g., cable, satellite, computers, etc.) and applications
- The potential for a worldwide standard

In this simple list of design goals, there are many conflicting desires. For example, there is a three-way tradeoff among HDTV picture quality, HDTV service area and interference to existing NTSC service. In order to have outstanding HDTV picture quality, a high data rate is desired. But achieving a high data rate over a large HDTV service area requires high power transmission that results in too much interference to existing NTSC service. Reducing the HDTV service area is unacceptable, because there would be insufficient audience to make the service economically viable. Fortunately, with some balancing of objectives and associated engineering tradeoffs, the combination of video compression technology and digital transmission technology is sufficient to enable the creation of an HDTV simulcast service.

As another example of conflicting goals, even when considered solely as an entertainment medium, the HDTV transmission standard must have a high degree of interoperability with its common production sources, in order to ensure economical operation from production through delivery to the home. Even in this limited context, interoperability of one kind may be in conflict with interoperability of another kind. The production sources for HDTV transmission will include: the HDTV production standard, film, computer graphics, and NTSC video. The conflicting interoperability desires are that while the HDTV production standard will likely be interlaced with square pixels at 59.94 Hz, the desire for NTSC interoperability argues for preserving non-square pixels and the 59.94 Hz temporal rate, while the desire for film interoperability argues for a 24 Hz temporal rate, while the desire for computer interoperability argues for square pixels, progressive scan and possibly a frame rate higher than 70 Hz. Clearly, these conflicting interoperability desires are mutually exclusive, yet the interoperability of an HDTV transmission standard with all of these sources is important. A solution can be found that takes into account existing and widely accepted interoperability practices, such as the conversion of 24 Hz film to 59.94 Hz through a combination of a 1000/1001 slowdown to 23.976 Hz and a 3:2 frame repeat sequence to speed up the temporal rate to 59.94 Hz. In addition, the conversion between square pixels and non-square pixels must take place at some interface boundary; the question is simply where. Understanding the technical difficulty and cost of one conversion compared to another is an important element of the complex tradeoffs that must be made in the design of an HDTV system.

To further complicate our decision space, we observe that HDTV is on the vanguard of an explosion in digital technology. HDTV products will exist in a world where digital delivery of television by cable, satellite and pre-recorded media are proliferating; where the availability of low cost computational power and digital data networks are enabling the creation of a National Information Infrastructure; and where the development of new interactive services and applications will drive the creation of information appliances and products that may be hybrids of today's products, or entirely new products based on technologies that rapidly cross the computing, communications and consumer electronics industries.

2. SYSTEM OVERVIEW

The Grand Alliance HDTV system employs two fundamental system principles that make it a highly interoperable system. First, it is designed with a layered digital system architecture, that results in the ability to interface with other systems at any layer. This means that many different applications can make use of various layers of the HDTV architecture. In addition, the GA has designed each individual layer of the system to be highly interoperable with other systems at corresponding layers. Secondly, the Grand Alliance HDTV system takes full advantage of the potential flexibility offered by a digital approach by using header/descriptor design principles that allow maximum flexibility to be achieved. As a truly flexible digital system, the Grand Alliance HDTV system provides the basis for an HDTV standard that will have a revolutionary impact on many industries and applications that are well beyond the scope of television today.

2.1 LAYERED DIGITAL SYSTEM ARCHITECTURE

The GA HDTV system is a layered digital system that consists of four primary layers: the Picture layer, the Compression (video and audio) layer, the Transport (packetization) layer and the Transmission layer. These layers are conceptually illustrated in Figure 2.1.

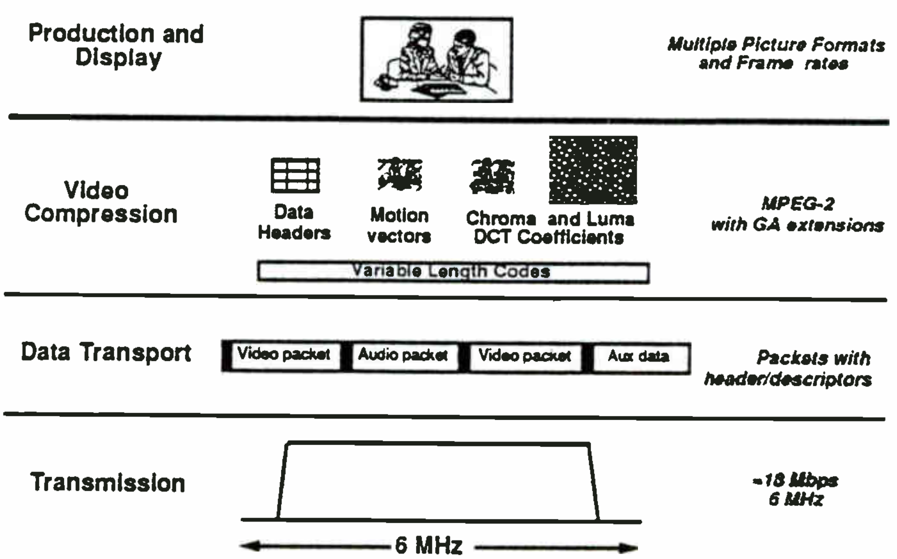

Figure 2.1 - The GA HDTV system's layered architecture and the PS-WP4 reference model.

While Picture layer Production and Display standards are not within the scope of a transmission standard, header/descriptors in the compression layer of the GA system support the transmission of multiple picture formats and frame rates that are related to both high-definition video and film production, and that are also appropriate for computer-based applications. At the Video Compression Layer, the GA HDTV system is based on MPEG-2 compression. At the transport layer, the GA HDTV system is also based on MPEG-2, which uses a flexible ATM-like packet protocol with headers/descriptors. At the Transmission Layer, the VSB modulation system will provide a net user data rate of over 18 Mbps delivered in the 6 MHz terrestrial simulcast channel.

The GA HDTV system has been designed to provide interoperability at every layer of its architecture, and the PS-WP4 reference model provides an excellent framework to explain its interoperability features. Figure 2.2 shows the protocol stack that consists of the layers of the GA HDTV system (shown on the left) expanded in a form that corresponds to the reference model established by PS-WP4 (shown in the middle) as a basis for evaluating interoperability. Also shown is the ISO Open Systems Interconnect (OSI) reference model for data communications (on the right).

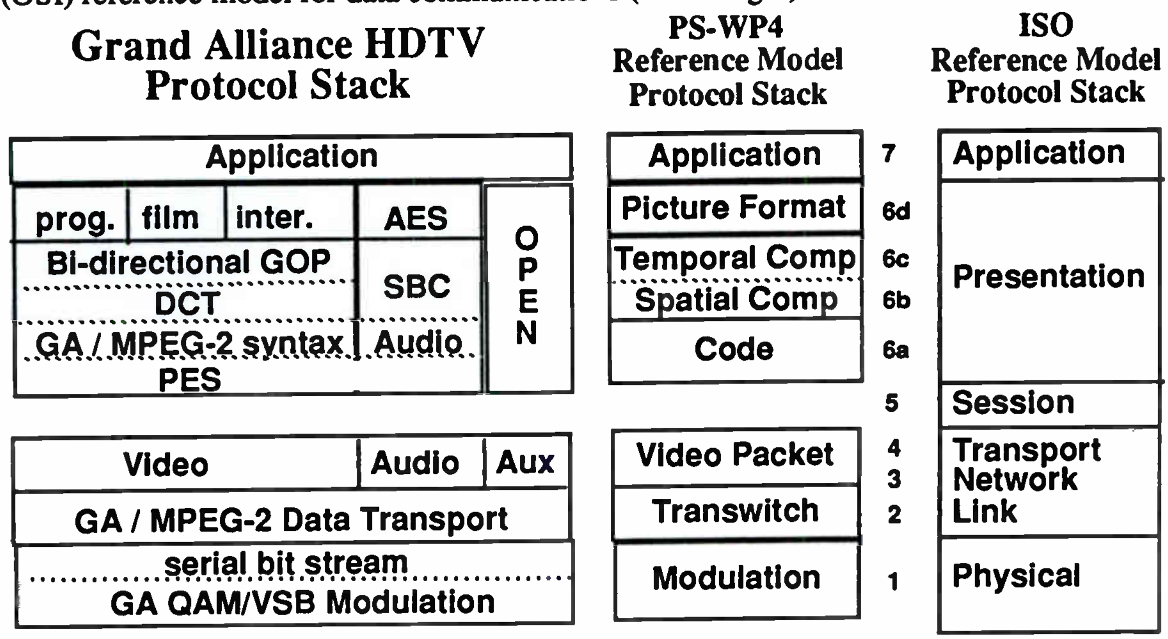

Figure 2.2 - The GA HDTV system's layered architecture and the PS-WP4 reference model

At the picture layer, progressive scan formats are provided at both video and film frame rates, in addition to an interlaced format. The compression layer conceptually consists of temporal compression, spatial compression and code (syntax) sublayers. For temporal compression, the MPEG Group of Pictures (GOP) approach is used to allow bi-directional motion-compensation across a block of frames. Spatial compression is achieved through the Discrete Cosine Transform (DCT). For the compressed code representation, the GA compression syntax is based on MPEG-2, but has additional syntax required to support AC coefficient leaky prediction. The Program Elementary Stream (PES) layer is the fundamental data stream that comprises a compressed video bit stream. At the video packet/transwitch layer, the GA system uses the MPEG-2 packet format, which provides the capability to synchronize the delivery of video, audio and auxiliary data packets in an ATM-like multiplexing environment. Finally, at the transmission layer, the serial bit stream is converted to a 6 MHz signal by Vestigal Sideband Modulation. Subsequent chapters of this document will describe each layer of the architecture.

Chapter II

VIDEO PICTURE FORMAT

1. Introduction

This document provides more detailed specifications of the scanning formats proposed on June 30, 1993 by the Grand Alliance, and includes modifications made since June 30 to the initial proposals. The document reflects extensive and valuable interactions between the Grand Alliance Specialist Group on Formats and the ACATS Technical Subgroup's Expert Group on Scanning Formats and Compression.

The parameters needed for specification of the HDTV systems are in two categories: those related to the interface to the encoder (on the transmission side), and those related to the encoding within the encoder in preparing the compressed data for transmission.

For the interface specification, the total line and pixel counts and the pixel clock rate are important. We have chosen to restrict the interface variations for the prototype to two formats, with 720 active lines and 1080 active lines. This limitation does not preclude other natural extensions for frame rates or for progressive scanning when such extensions become useful and equipment for new format variations is available.

For specifying the transmission format, which deals with the compressed representation of the active picture information, the total line and pixel counts are not relevant, but the frame rate variations can be distinguished and coded according to their frame rates are significant.

Parameters to be specified

The parameters that need to be specified for the interface and transmission formats include the following:

A. Number of frames per second.

B. Number of fields per frame (i.e., whether interlaced or progressive scanning is specified).

C. Number of active lines per frame.

D. Total number of lines per frame.

E. Number of active video samples per line.

F. Total number of video clock cycles per line.

G. Sampling frequency for video samples (derivative from the above specifications).

H. Allowable combinations of parameters.

Note that the HDTV system proposed by the Grand Alliance intrinsically includes more than one format, so that for some parameters there is more than one allowable value. Also note that 59.94 Hz and 60.0Hz frame/field rates are considered implementation variations, not defining separate formats.

2. Format Specifications

The Grand Alliance proposal includes two main format variations, with different numbers of lines per frame. These formats have 720 active lines and 1080 active lines per frame.

For interfacing to the encoder, the 720-line format uses 1280 active samples per line, while the 1080-line format uses 1920 active samples per line. These choices yield square pixels for all formats. Also for interfacing to the encoder, the 720-line format is progressively scanned, with 60 Hz (nominal) frame rate, while the 1080-line format is interlaced 2:1 with a 60 Hz nominal field rate. The prototype encoder input accepts 787.5 total lines per frame and 1125 total lines per frame for the two interface formats.

For compression and transmission, the frame rate for the 720-line format and for the 1080-line format can be 60.0 Hz, 24 Hz, or 30 Hz. The same formats are also supported with the NTSC-friendly numbers, i.e., 59.94 Hz, 23.98 Hz, and 29.97 Hz.

The 60.0 Hz and 59.94 Hz variations for the 1080-line format are encoded as interlaced scanned images for the initial HDTV systems. Other formats are encoded as progressively scanned.

2.1. Number of frames per second

The frame rates for the Grand Alliance HDTV system will include both 59.94 Hz and 60.0 Hz, since the 59.94 Hz rate may simplify interworking with NTSC material during the simulcast period, and because the 60.0 Hz rate may have interoperability benefits in the future.

The proposal includes 24 Hz and 30 Hz (including the corresponding adjustments for the NTSC-friendly frequencies) as film modes. This means that the encoder will be designed to encode the reduced pixel rate from image sequences that originated at 24 Hz or 30 Hz. It also means that for both the 720-line and 1080-line formats, the encoder will be designed to identify and take advantage of the lesser pixel rate in the film modes if the film mode material is presented to the encoder at 59.94 Hz or 60.0 Hz rate, using a pull-down technique.

Initial prototypes of the HDTV equipment will not contain interfaces for direct connection to 24 Hz or 30 Hz material, although the encoder architecture can support such interfaces when they become useful.

2.2. Number of fields per frame

The 1080-line format with 59.94 Hz or 60.0 Hz rate will be interlaced 2:1, yielding a frame rate of 29.97 Hz or 30.0 Hz. All the other format variations will use progressive scanning.

2.3 Number of active lines per frame

The number of active lines per frame will be either 720 active lines or 1080 active lines.

2.4. Total number of lines per frame

The total line count in the prototype for the 720-line format will be 787.5 lines per frame.

The total line count in the prototype for the 1080-line format will be 1125 lines per frame.

2.5 Number of active video samples per line

The number of active samples per line for the 720-line format will be 1280 samples of pixels.

For the 1080-line format, the progressive-scan variations with 24 Hz or 30 Hz frame rates will include 1920 active video samples per line. For the 59.94 Hz and 60.0 Hz interlaced variations of the 1080-line format, the number of active samples per line will be 1920 for the input interface.

The prototype encoder will be designed to support both 1440 and 1920 active samples per line for encoding and transmission, for the interlaced format only. The encoded data stream will include a signal that indicates the number of samples per line.

2.6 Total number of video samples per line

The total number of video samples per line for the 720-line format used in the prototype will be 1600, for all variations.

For the 1800-line format in the prototype, each line will represent 2200 cycles of the pixel rate clock, or 2200 sample times.

The Grand Alliance prototype also incorporates a variation of the 1080-line format with 1440 active samples per line.

2.7 Sampling frequency for video samples

For the 720-line format, the pixel rate clock used in the prototype will be 75,600,000 Hz for the 60.0 Hz frame rate, and approximately 75,524,476 Hz for the 59.94 Hz frame rate (which is 1000/1001 times the 60.0 Hz rate).

The clock rates used in the prototype for the 1080-line format will be 74,250,000 Hz for the 60.0 Hz interlaced variation, and approximately 74,175,824 Hz for the 59.94 Hz interlaced variation, at the input interface to the encoder.

Note that the ratio of clock rates for the specified 60 Hz inputs is 56:55 (75.6 MHz: 74.25 MHz).

2.8 Allowable combinations of parameters

Internally, the encoder for the 720-line format uses three frame rates, to allow the encoder to take advantage of film sequence redundancies. For the input to the encoder, a single format at either 59.94 Hz or 60.0 Hz will be interfaced to the encoder for the initial system.

The 1080-line format will have an input interface with 1920 active samples per line. As previously indicated, the Grand Alliance may reduce the number of samples per line for compression, for the interlaced format only, if necessary to preserve image quality. In that case, the number of samples encoded would be 1440 samples per line, for the interlaced variation at 59.94 or 60.0 Hz field rates. The progressive scan 24 Hz and 30 Hz variations with 1080 active lines will use 1920 active samples per line.

3 Summary Table for formats

The table attached shows the allowable variations. The table indicates entries for 59.94 Hz and 60.0 Hz field/frame rates, although these should not be considered distinct formats, merely frame rate implementation options.

GRAND ALLIANCE VIDEO SCANNING FORMATS
Encoder Format for Transmission

Active Lines	Active Pixels/Lines	Rate (Hz)	Pro/Int
720	1280	59.94/60.0	1:1
	1280	23.98/24.0	1:1
	1280	29.97/30.0	1:1
1080	1920/1440	59.94/60.0	2:1
	1920	23.98/24.0	1:1
	1920	29.97/30.0	1:1

Chapter III

VIDEO COMPRESSION SYSTEM

1 OVERVIEW

The Grand Alliance (GA) HDTV system is an all-digital system designed to transmit high quality video and audio over a single 6 MHz terrestrial channel. A major challenge in the system design is video compression. This report summarizes the current status of the video compression subsystem of the GA HDTV system. The Grand Alliance compression specialist group is in the process of further evaluating the video compression subsystem and additional improvements maybe added in the future.

Modern digital transmission technologies deliver approximately 17 Mbps to about 20 Mbps to encode video data within a single 6 MHz terrestrial channel. This means that encoding the HDTV video source whose resolution is typically six times that of the NTSC resolution requires a bit rate reduction by a factor of 50 or higher. To achieve this bit rate reduction, the GA HDTV system is designed to be very efficient in utilizing available channel capacity by exploiting modern, sophisticated video compression technology. The methods utilized, for example, include source adaptive processing, motion estimation/compensation, transform representation, and statistical coding. The GA system has been designed to incorporate most key video compression technology utilized in previously tested systems. The GA system has also been designed to perform well with progressive as well as interlaced pictures. Due to efficient channel utilization, the GA video compression system is an excellent choice, not only for terrestrial broadcasting, but for cable and satellite broadcasting.

In designing the GA video compression system, we have emphasized compatibility with MPEG (Moving Picture Experts Group) syntax. For example, video compression techniques which are not compatible with MPEG syntax have not been considered unless significant benefit was clearly demonstrated. There are several reasons behind the emphasis on MPEG compatibility. The MPEG standard was developed utilizing modern sophisticated video compression methods and is very efficient in video compression. In addition, the MPEG standard was established by cooperative efforts of a number of organizations throughout the world. By placing special emphasis on compatibility with MPEG syntax, we have designed a video compression system which can serve as the HDTV standard not only for North America, but also for world-wide use.

We believe that the video compression subsystem of the GA HDTV system can deliver excellent HDTV quality video within a single 6 MHz terrestrial broadcast channel in North America. Furthermore, we believe that the system is an excellent choice for world-wide HDTV delivery for cable and satellite channels.

2 BASICS OF VIDEO COMPRESSION

A major objective of video compression is to represent a video source with as few bits as possible while preserving the level of quality required for the given application. Video compression has two major applications. One is efficient utilization of channel bandwidth for video transmission systems. The other is reduction of storage requirements. Both of these applications apply to the HDTV system.

Video compression for HDTV application is shown in Figure 2-1. The video source is encoded by the video encoder. The output of the video encoder is a string of bits that represents the video source. The channel coder transforms the string of bits to a form suitable for transmission over a channel through some form of modulation. The modulated signal is then transmitted over a communication channel. The communication channel typically introduces some noise, and provision for error correction is made in the channel coder to compensate for this channel noise. At the receiver, the received signal is demodulated and transformed back into a string of bits by a channel decoder. The video decoder reconstructs the video from the string of bits for human viewing. This report focuses on video compression, namely video encoding and decoding.

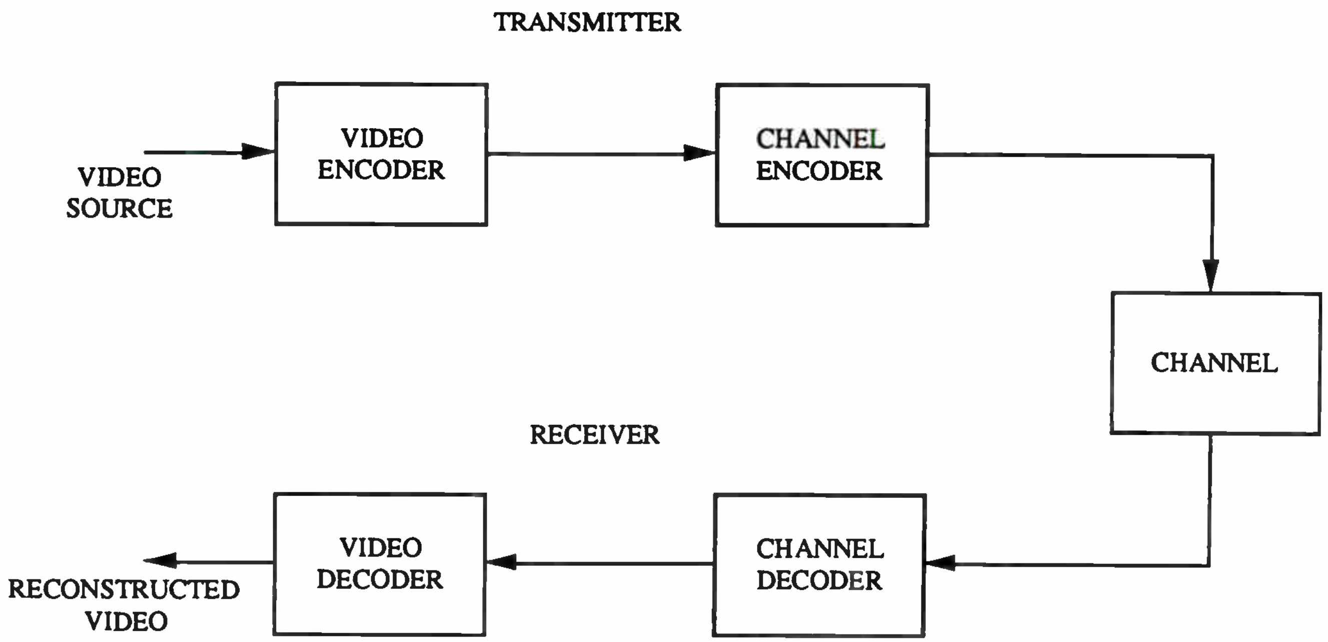

Figure 2-1. Video Compression for HDTV Application

The video compression system consists of three basic operations, as shown in Figure 2-2. In the first stage, the video signal is expressed in a more efficient representation which facilitates the process of compression. In essence, the representation determines what specifically is coded. The representation may contain more pieces of information to describe the signal than the signal itself, but most of the important information will be concentrated in only a small fraction of this description. In an efficient representation, only this small fraction of the data needs to be transmitted for an appropriate reconstruction of the video signal. The second operation, quantization, performs the discretization of the representation information. To transmit video over a digital channel, the representation information is quantized to a finite number of levels. The third operation is assignment of codewords, which are the strings of bits used to represent the quantization levels.

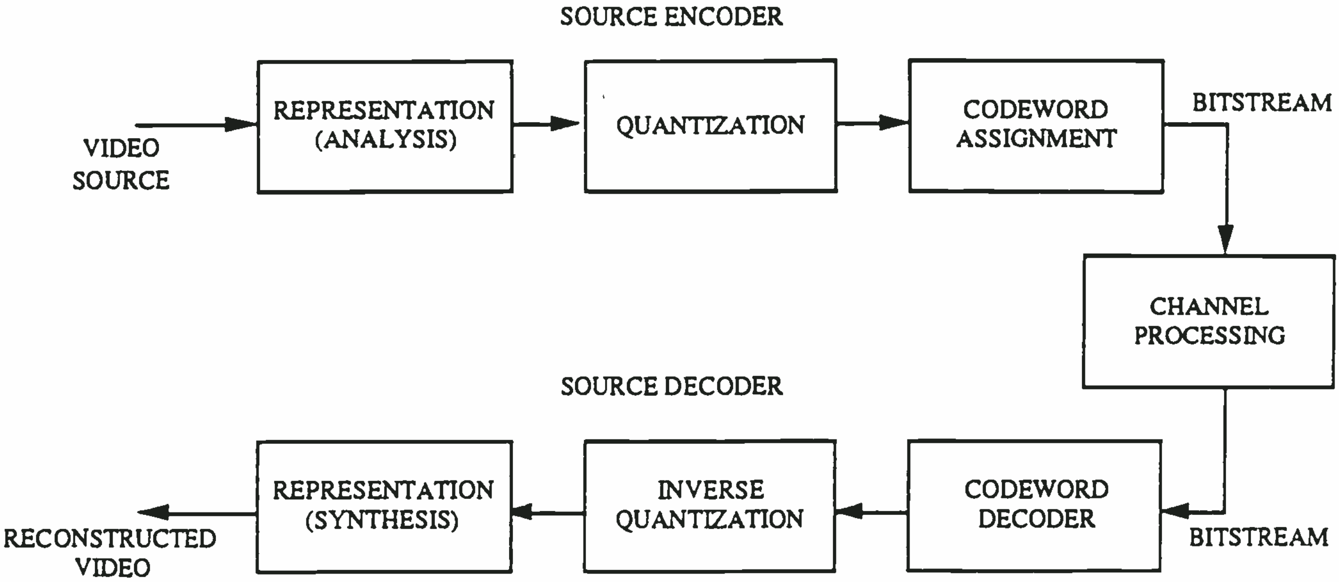

Figure 2-2. Overview of Video Compression System

Each of the three operations attempts to exploit the redundancy present in the video source and the limitations of the human visual system. By removing the redundancy present in the video source, the same or related information does not have to be transmitted repeatedly. By exploiting the limitations of the human visual system, the information in the video source that is not utilized by the human visual system does not need to be transmitted. In the following sections, we discuss the methods used in the GA video compression system.

2.1 Representation

To remove the redundancy in the signal and to eliminate the information irrelevant to the human viewer, the GA system utilizes many techniques discussed in this section.

2.1.1 Source-Adaptive Processing

A given television system must provide an interface to many kinds of video source formats. These include video from a video camera, the various film standards, magnetic and optical media, and synthetic imagery. In the GA system, we exploit the differences that may exist among the various sources to improve the performance of the video compression system.

2.1.2 Color-Space Processing

The input video source to the GA video compression system is in the form of R (red), G (green), and B (blue) components. The RGB components are highly correlated with each other. Furthermore, the human visual system responds differently to the luminance and chrominance components. To reduce the correlation and exploit this difference in the human visual system, the RGB components are converted to the YC1C2 color space through a linear transformation. Y corresponds to the luminance (intensity or black-and-white picture) while C1 and C2 correspond to the chrominance.

In the YC1C2 color space, most of the high frequency components are concentrated in the Y component. Furthermore, the human visual system is less sensitive to high frequency components of the chrominance components than of the luminance component. To exploit these characteristics, in the GA video compression system the chrominance components are low-pass filtered and sub-sampled by a factor of two along both the horizontal and vertical dimensions, producing chrominance components that are one-fourth the spatial resolution of the luminance component.

2.1.3 Motion Estimation/Compensation

A video sequence is a series of still images shown in rapid succession to give the impression of continuous motion. Even though each of the frames is distinct, the high frame rate necessary to achieve proper motion rendition usually results in much temporal redundancy among the adjacent frames. Motion compensation attempts to remove this temporal redundancy.

Much of the variation in intensity from one frame to the next is due to object motion. In motion-compensated coding, the current frame is predicted from the previously encoded frame by estimating the motion between the two frames and compensating for the motion. The difference between the current frame and the prediction of the current frame is called the motion- compensated residual and this residual is encoded. For a typical video sequence, the energy in the residual is much less than that in the original video due to the removal of the temporal redundancy. Encoding the residual rather than the video itself ensures that the temporally redundant information is not encoded repeatedly. In motion estimation, the same imagery is assumed to appear in consecutive video frames, although possibly at different locations. The motion may be global or local within the frame. To optimize the performance, an estimate of the motion is computed for each local region within a frame. The most common model for the local motion is simple translational motion. This model is highly restrictive, and cannot represent the large number of possible motions, such as rotations, scale changes, and other complex motions. Nevertheless, by assuming these motions only locally and by identifying and processing those regions separately where the model fails, excellent performance can be achieved.

One approach for motion estimation, which is very popular and is utilized in the GA video compression system, is based on block matching. In this approach, the current frame is partitioned into rectangular regions or blocks, and a search is performed for the displacement which produces the "best match" among possible blocks in an adjacent frame. Block matching is popular, because it achieves high performance and exhibits a simple, periodic structure which allows straightforward VLSI implementation. The process of motion-compensated prediction used in the GA system is illustrated in Figure 2-3. The frame to be encoded is partitioned into blocks and each block is predicted by displacing a block from the reference frame. The difference between the current frame and its prediction is the motion-compensated residual.

2.1.4 Intra-frame and Inter-frame Encoding

Motion-compensated coding is an example of inter-frame encoding. In some cases, it is useful to encode the video itself (intra-frame coding) without motion compensation. The GA video compression system utilizes both intra-frame coding and inter-frame coding.

Motion compensation is a form of predictive coding applied along the temporal dimension. For typical video frames, prediction by motion compensation is quite good. For some frames such as those at scene changes, however, a prediction is not as good. Even within a given frame, some regions may be predicted well with motion compensation while other regions may not be predicted very well using motion compensation based on translational motion. In some cases, therefore, the system may perform worse with predictive coding than by simply coding the frame itself. The GA video compression system utilizes inter-frame encoding when the motion- compensated prediction is quite good, while utilizing intra-frame encoding otherwise.

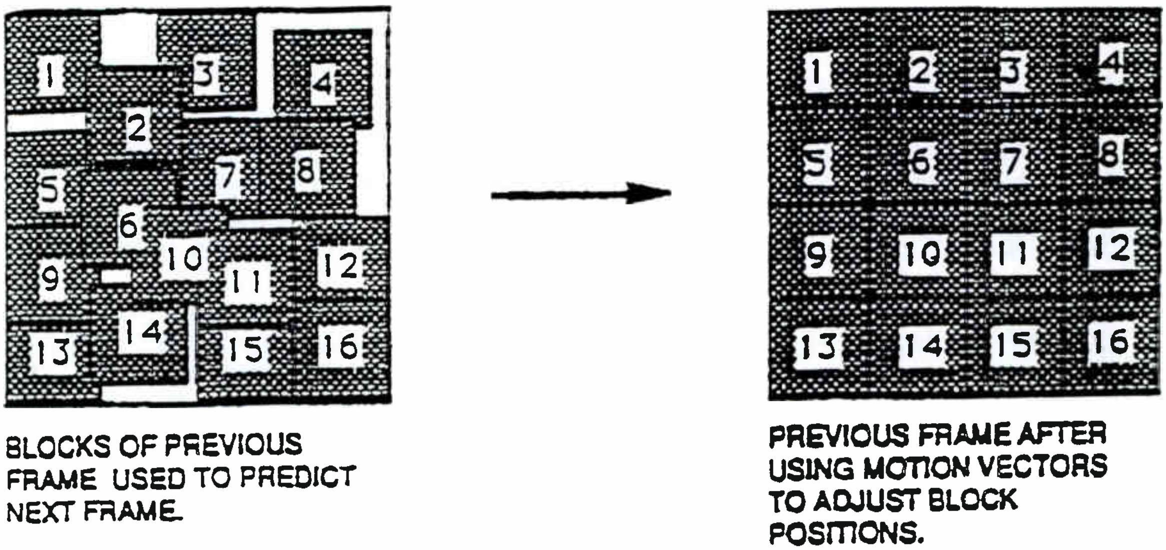

Figure 2-3. Motion-Compensated Prediction

The intra-frame coding mode is also utilized in the GA system for receiver initializations and channel acquisition (when the receiver is turned on or the channel is changed). It is also used when non-correctable channel errors occur. With the motion-compensated prediction, an initial frame must be available at the decoder to start the prediction loop. Therefore, a mechanism must be built into the system so that if the decoder loses synchronization for any reason, it can rapidly reacquire tracking. One approach used is periodic intra-frame encoding (refresh). In this approach, the entire frame is encoded using the intra-frame mode (I frames). Alternatively, a part of the frame is encoded using the intra-frame mode, and by encoding different parts of the frame in different frames, the entire frame is refreshed (progressive refresh). This refresh approach produces a periodic re-initialization of the temporal prediction which allows the decoder to re-acquire tracking. The GA system utilizes both progressive refresh and complete-frame refresh through the use of I-frames.

2.1.5 Discrete Cosine Transform

Motion compensation reduces the temporal redundancy of the video signal, but there still remains spatial redundancy in the motion-compensated residual (MC-residual). This is especially true if no MC processing is performed and the original frame is to be encoded. For simplicity, the term residual will be applied to whatever frame of data is to be spatially processed, irrespective of whether MC has been applied or not. To reduce the spatial redundancy of the residual, the Discrete Cosine Transform is used.

The Discrete Cosine Transform (DCT) compacts most of the energy of the residual into only a small fraction of the transform coefficients. The coding and transmission of only these energetic coefficients can result in the reconstruction of high quality video. The DCT was chosen in the GA system because it has good energy compaction properties, results in real coefficients, and there exist numerous fast computational algorithms for its implementation. The DCT of a two-dimensional signal may be computed by applying the one-dimensional DCT separably first to the rows and then to the columns of the signal.

The size of the DCT might be chosen as the entire frame, but much better performance can be achieved by subdividing the residual into many smaller regions each of which is individually processed. The motivation for this is very easy to see and it is one of the most important aspects of a high quality video encoder. If we compute the DCT of the entire residual frame, we treat the whole residual frame equally. For a typical image, some regions contain a large amount of detail and other regions contain very little detail. By exploiting the changing characteristics of different images and of different portions of the same image, significant improvements in performance can be realized. In order to take advantage of the varying characteristics of the residual over its spatial extent, the residual is partitioned into 8x8 blocks. The blocks are then independently transformed and adaptively processed based on their local characteristics. The partitioning of the residual into small blocks before taking the transform not only allows spatially adaptive processing, but also reduces the computational and memory requirements. Partitioning the signal into small blocks before computing the DCT is referred to as the Block DCT.

The DCT tends to reduce the spatial correlation of the residual. This makes the DCT domain an appropriate representation since the DCT coefficients tend to have less redundancy. In addition, the DCT coefficients contain information about the spatial frequency content of the residual. By exploiting the spatial frequency properties of the human visual system, the DCT coefficients can be encoded to match the human visual system so that only the perceptually important DCT coefficients are encoded and transmitted.

2.2 Quantization

Through the processing discussed up to this point, an elegant representation in the form of the motion field, spatial frequency coefficients, and luminance/chrominance components has been created; however, no compression has been achieved. In fact, an expansion of data has resulted, since there are currently more pieces of information used to describe the video than before. However, most of the perceptually important information has been compressed into only a fraction of these "pieces of information", and this data can be selected and encoded for transmission.

The goal of video compression in the HDTV application is to maximize the video quality at a given bit rate. This requires a wise distribution of the limited number of available bits. By exploiting the statistical redundancy and perceptual irrelevancy within the new representation, an appropriate bit allocation can yield significant improvements in performance. Quantization is performed to discretize the values, and through quantization and codeword assignment, the actual bit rate compression is achieved. The quantization process can be made the only loss step in the compression algorithm. This is very important, as it simplifies the design process and facilitates fine tuning of the system. Quantization may be applied to elements individually (scalar quantization) or to a group of elements simultaneously (vector quantization).

In scalar quantization (SQ), each element may be quantized with a uniform (linear) or nonuniform (nonlinear) quantizer. The quantizer may also include a dead zone (enlarged interval around zero) to quantize or core to zero small, noise-like perturbations of the element value. The close relationship between quantization and codeword assignment suggests that separate optimization of each may not necessarily yield the optimum performance. On the other hand, joint optimization of quantization and codeword assignment is a highly nonlinear and complex process. Experiments have shown that a linear quantizer with an appropriate step-size individually chosen for each element to be quantized, followed by proper entropy coding, may yield close to optimum performance. This is the approach taken in the GA system.

When quantizing transform coefficients, the differing perceptual importance of the various coefficients can be exploited by "allocating the bits" to shape the quantization noise into the perceptually less important areas. This can be accomplished by varying the relative step-sizes of the quantizers for the different coefficients. The perceptually important coefficients may be quantized with a finer step-size than the others. For example, low spatial frequency coefficients may be quantized finely, while the less important high frequency coefficients may be quantized more coarsely. A simple method to achieve different step-sizes is to normalize or weight each coefficient based on its visual importance. All of the normalized coefficients may then be quantized in the same manner, such as rounding to the nearest integer (uniform quantization). Normalization or weighting effectively scales the quantizer from one coefficient to another. The GA video compression system utilizes perceptual weighting, where the different DCT coefficients are weighted according to a perceptual criterion prior to uniform quantization. The perceptual weighting is determined by quantization matrices and the GA video compression system uses adaptive quantization matrices.

In video compression, most of the transform coefficients are quantized to zero. There may be a few non-zero low-frequency coefficients and a sparse scattering of non-zero high-frequency coefficients, but the great majority of coefficients will have been quantized to zero. To exploit this phenomenon in the GA system, the two-dimensional array of transform coefficients is reformatted and prioritized into a one-dimensional sequence through a zigzag scanning. This results in most of the important non-zero coefficients (in terms of energy and visual perception) being grouped together early in the sequence. They will be followed by long runs of coefficients that are quantized to zero. These zero-valued coefficients can be efficiently represented through run length encoding. In run length encoding, the number (run) of consecutive zero coefficients before a non-zero coefficient is encoded, followed by the non-zero coefficient value. The run length and the coefficient value can be entropy coded, either separately or jointly. The scanning separates most of the zero and the non-zero coefficients into groups, thereby enhancing the efficiency of the run length encoding process. Also, a special End Of Block (EOB) marker is used to signify when all of the remaining coefficients in the sequence are equal to zero. This approach is extremely efficient, yielding a significant degree of compression.

2.3 Codeword Assignment

Quantization creates an efficient discrete representation for the data to be transmitted. Codeword assignment takes the quantized values and produces a digital bit stream for transmission. The quantized values can be simply represented using uniform or fixed length codewords. Using this approach, every quantized value will be represented with the same number of bits. Greater efficiency, in terms of bit rate, can be achieved by employing entropy coding. Entropy coding attempts to exploit the statistical properties of the signal to be encoded. A signal, whether it is a pixel value or a transform coefficient, has a certain amount of information, or entropy, based on the probability of the different possible values or events occurring. For example, an event that occurs infrequently conveys much more new information than one that occurs often. By realizing that some events occur more frequently than others, the average bit rate may be reduced.

There are two important issues that arise when considering the application of entropy coding. First, entropy coding involves increased complexity and memory requirements over fixed length codes. Second, entropy coding coupled with the non-stationarity of the video signal results in a time-varying bit rate. (Other aspects of the source coding may also raise this issue). A buffer control mechanism is necessary when the variable bit rate source coder is to be coupled with a constant bit rate channel.

In the GA system, an entropy coder is used to reduce the statistical redundancy inherent in the elements encoded for transmission. The primary redundancy is the nonuniform probability distribution over the possible range of each element. The more the probability distribution deviates from a uniform distribution, the greater improvement can be achieved via entropy coding. Other sources of statistical redundancy which may exist include the statisticaldependence among the encoded elements.

Huffman coding which is utilized in the GA system is one of the most common entropy coding schemes. In Huffman coding, a code book is generated which minimizes the entropy subject to the codeword constraints of integer lengths and unique decodability. Events which are more likely to occur will be assigned shorter length codewords while those which are less likely to occur will be assigned longer length codewords. Huffman coding results in a variable bit rate per event; more importantly, the average bit rate is reduced. The training or generation of the code book is achieved by using a representative set of data to estimate the probability of each event. Optimal performance can be achieved by designing an individual code book for each element to be encoded. However, this results in a large number of code books. Close to optimal performance is achieved by using a new code books where elements with similar statistics are grouped and encoded together. Similarly, the size of each code book can be reduced by grouping together very unlikely events into a single entry within the code book. When any event belonging to this group occurs, the codeword for this group is transmitted followed by an exact description of the event.

Whenever entropy coding is employed, the bit rate produced by the encoder is variable and is a function of the video statistics. If the application requires a constant bit rate output, a buffer is necessary to couple the two. The buffering must be carefully designed. Random spikes in the bit rate can overflow the buffer while dips in the bit rate can produce an underflow. What is needed is some form of buffer control that would allow efficient allocation of bits to encode the video while ensuring that no overflow or underflow occurs.

The buffer control typically involves a feedback mechanism to the compression algorithm whereby the amplitude resolution (quantization) and/or spatial, temporal and color resolution may be varied in accordance with the instantaneous bit rate requirements. The goal is to keep the average bit rate constant and equal to the available channel rate. If the bit rate decreases significantly, a finer quantization can be performed to increase it. The buffer control mechanism is an essential part of any high-performance constant-output bit rate HDTV system. The GA system utilizes a highly sophisticated buffer control mechanism to deliver the highest video quality within a reasonable buffer size.

In this section, some of the basic video compression elements utilized on the GA video compression system have been discussed. Further details can be found in references 1 and 2.

3. VIDEO COMPRESSION APPROACH

Figure 3-1 shows the functional block diagram of the GA video encoder. The components are described in the following sections.

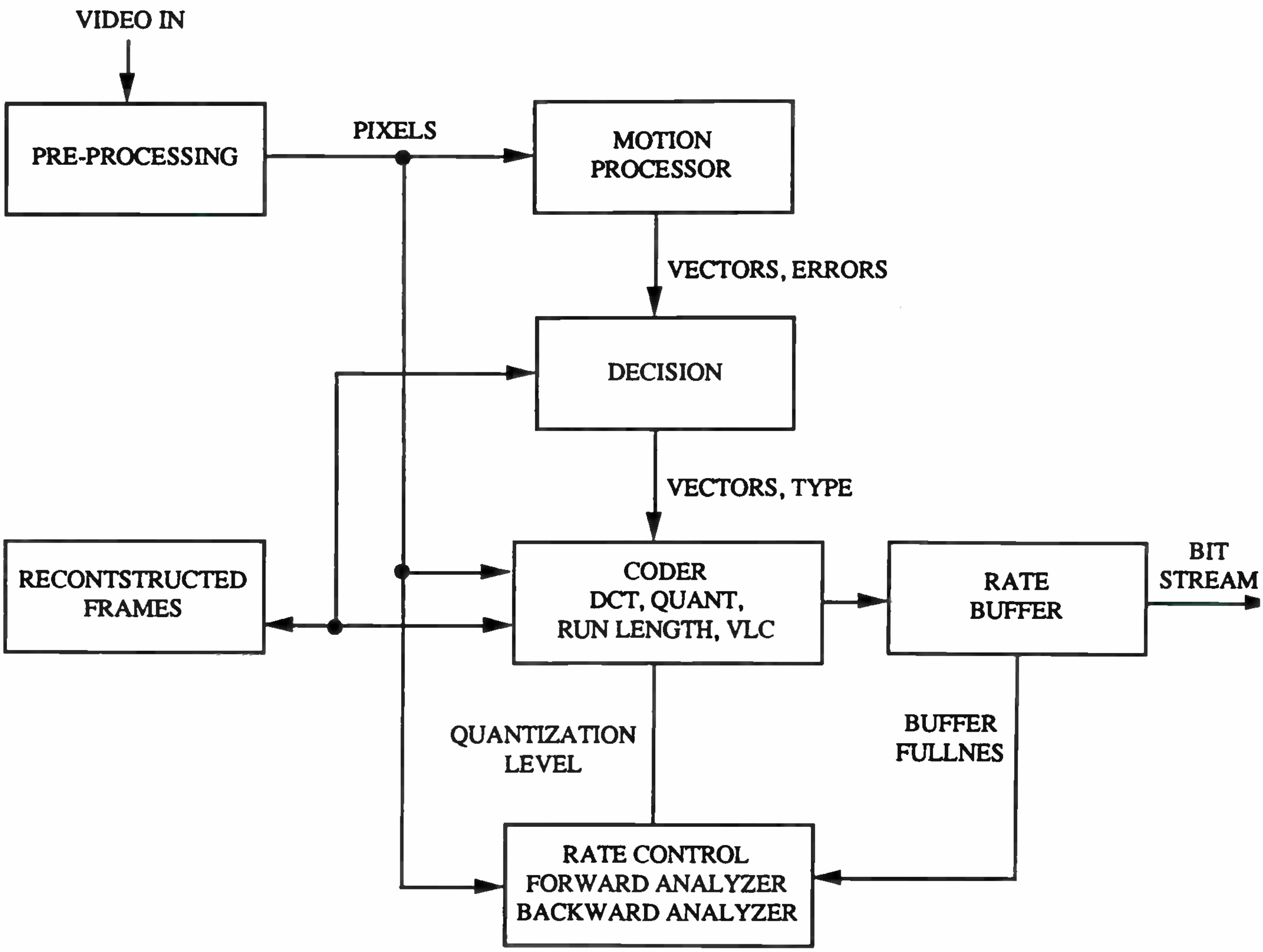

Figure 3-1. GA Video Encoder

3.1 Video Preprocessor

As shown in Figure 3-2, analog video is received in RGB format and digitized using 10-bit A/D converters. Gamma correction is then applied to each color component in order to compensate for the non-linear response of the camera. This helps to reduce the visibility of quantization noise, particularly in the dark regions of the image.

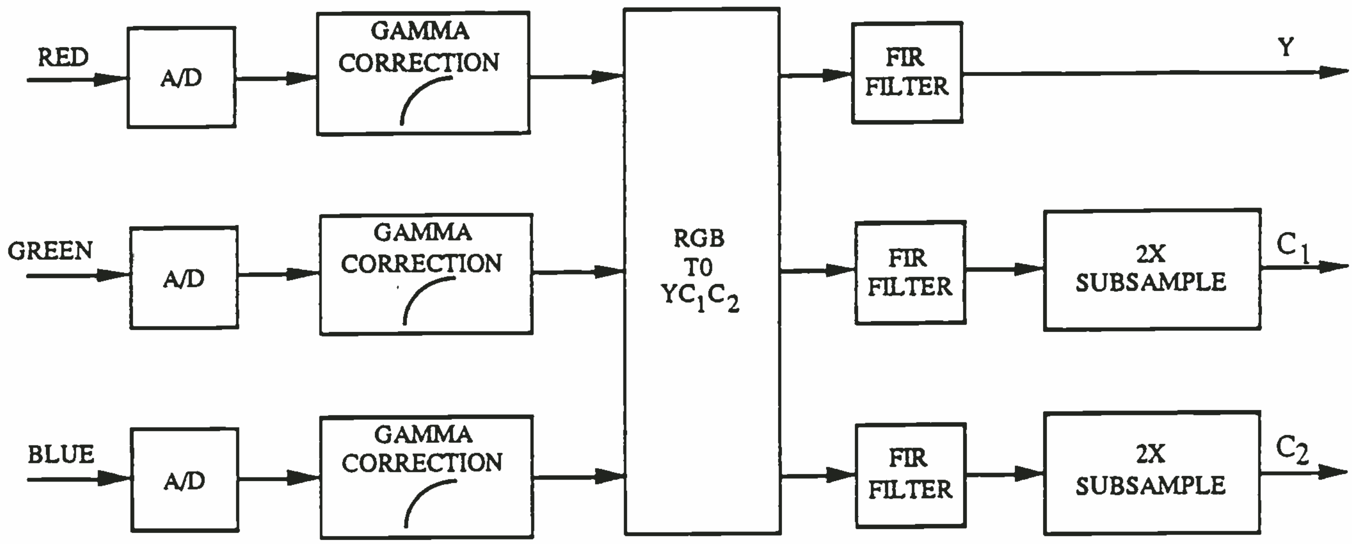

Figure 3-2. Video Preprocessor

The digitized and gamma-corrected RGB samples are then converted to the SMPTE 240M YC1C2 color space using a linear matrix transformation. Programmable FIR filters are used to shape the frequency response of each color component. The two chrominance components use half-band filters in order to prevent aliasing during the subsequent sub-sampling process. The sub-sampling is performed by a factor of 2, both horizontally and vertically.

3.2 Motion Estimation and Compensation

The GA compression algorithm utilizes multiple prediction methods to effectively provide motion compensation for progressive and interlaced pictures. The prediction methods include frame prediction, adaptive field prediction, dual prime (used for forward prediction only), and bi-directional prediction. Among these, the dual prime and the bi-directional prediction modes deserve further discussion. In the dual prime case the prediction is formed from pixels in previously displayed frames, while in the B-frames case the prediction is based on both previous frames and future frames. The dual prime and bi-directional methods are complementary since the dual prime mode is for interlaced only and is precluded when a sequence uses B-frames.

3.2.1 The "P-Frame" Motion Vector Prediction Mode

P-frames are frames where the prediction is in the forward direction only (i.e., predictions are formed only from pixels in previously displayed frames). These forward-predicted frames allow the exploitation of inter-frame coding techniques to improve the overall compression efficiency and picture quality.

Figure 3-3 illustrates a time sequence of video frames consisting of intra-coded pictures, predictive coded pictures, and bi-directionally predictive coded pictures. P-frames are predicted using the most recently encoded P-frame of I-frame in the sequence. In Figure 3-3, P-frames occur every second frame except when an I-frame is used. Each macroblock within a P-frame can be either forward-predicted or intra-frame coded. If a macroblock is forward-predicted, then either frame-based or field-based prediction may be used. The decision is made by the encoder based on the smallest prediction error using each method. Dual prime prediction may also be chosen at this stage if the prediction error using dual prime is smallest. Regardless of whether field-based, frame-based or dual prime prediction is used, the predicted macroblock is subtracted from the original macroblock, and only the "difference" values are encoded and transmitted.

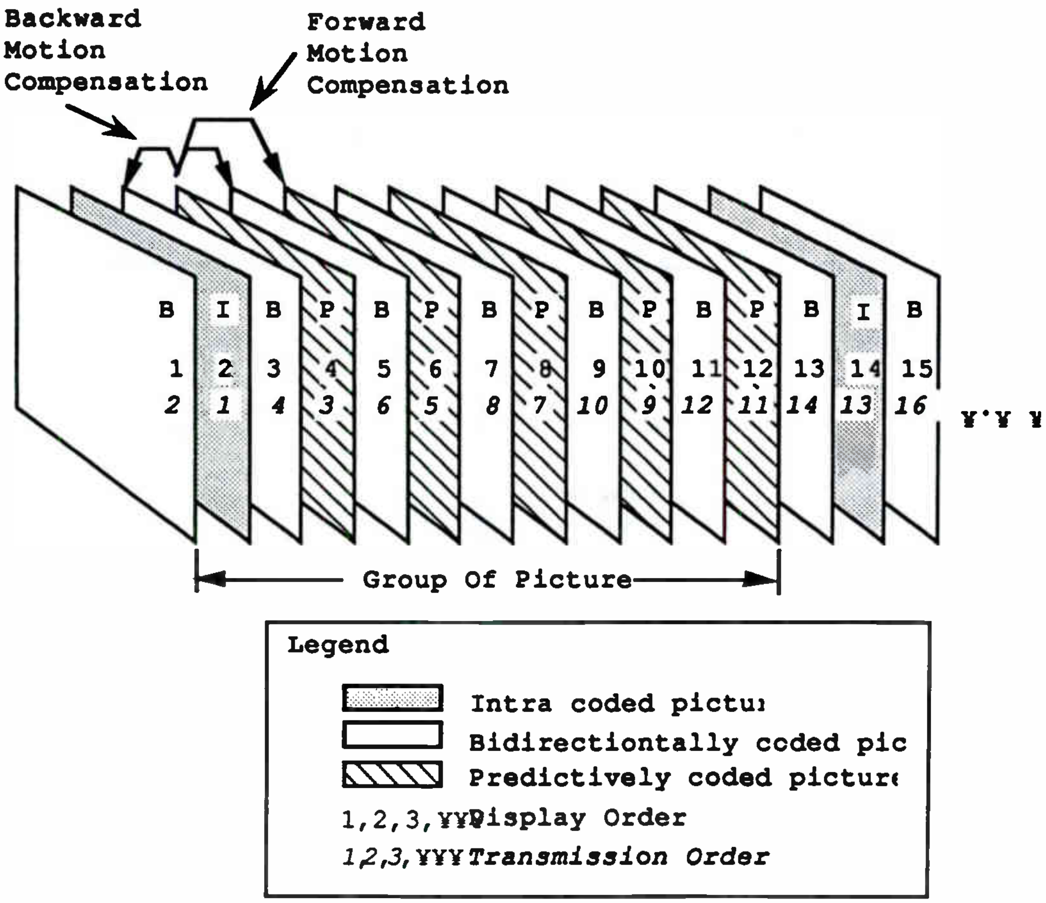

Figure 3-3 Example of a Coded Video Sequence Using P-Frames and B-Frames

3.2.1.1 Dual Prime

The dual prime prediction mode is an alternative "special" prediction mode which is based on field-based motion prediction but requires fewer transmitted motion vectors than conventional field-based prediction. This mode of prediction is available only for interlaced material and only when the encoder configuration does not use B-frames (M=1). For these reasons, this mode of prediction is particularly useful for improving encoder efficiency for low delay applications.

The basis of dual prime prediction is that field-based predictions of both fields in a macroblock (MB) are obtained by averaging two separate predictions which are predicted from the two nearest decoded fields (in time). Each of the MB fields is predicted separately, although the four vectors (one pair per field) used for prediction are all derived from a single transmitted field- based motion vector. In addition to the single field-based motion vector, a small "differential" vector (limited to vertical and horizontal component values of +1, 0 and -1, and represented by two 1-2 bit codes) is also transmitted for each MB. Together, these vectors are used to calculate the pairs of motion vectors for each MB field. the first prediction vector in the pair is simply the transmitted field-based motion vector. The second prediction vector is obtained by combining the differential vector with a scaled version of the first vector. Once both predictions are obtained, a single prediction for each macroblock field is calculated simply by averaging each pel from the two original predictions. The final averaged prediction is then subtracted from the macroblock field being encoded.

Figure 3-4 illustrates the relationship between the transmitted vectors (one field-based vector and one differential vector) and the prediction vectors for each of the fields in a macroblock. The two separate sets of vectors shown in Figure 3-4 correspond to the predictions for the two fields which make up the macroblock. The transmitted field-based motion vector and the transmitted differential vector (identical for each set) are represented by solid lines. The differential motion vector is the smaller vertical vector. The scaled vectors are represented by dotted lines. The second (calculated) field-based motion vectors are represented by the dashed lines.

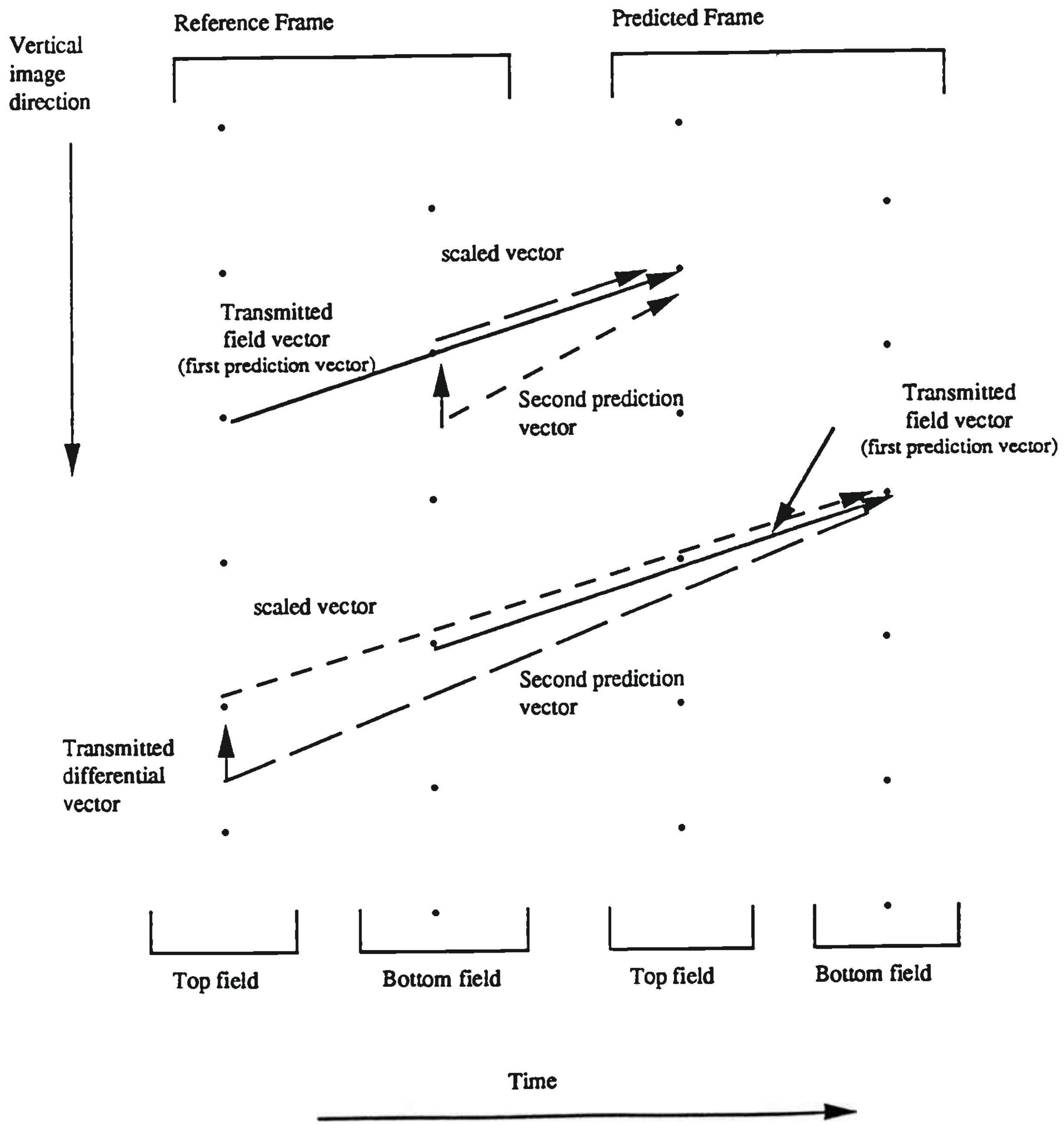

Figure 3-4. Dual Prime Prediction

3.2.2 The "B-Frame" Motion Vector Prediction Mode

The B-frame is a picture type within the coded video sequence which has some special prediction modes available to the encoder. It is useful for increasing the compression efficiency and perceived picture quality when encoding latency is not an important factor. It is also an attractive tool since it works equally well with both interlaced and progressive material. However, it has impact on the receiver cost since it requires additional memory. B-frames are included in the GA system because the increase in compression efficiency is noticeable especially with progressive scanning where techniques such as dual prime are not available. The improvement in performance was found to be sufficient to justify the cost of the additional memory required by the ATV receiver.

The basis of the B-frame prediction is that a video frame is correlated both with frames which occur in the past and the frames in the future. Consequently, if a future frame is available to the decoder, a superior prediction can be formed, thus saving bits and improving performance. Some of the consequences of using future frames in the prediction are: the B-frame cannot be used for predicting future frames, the transmission order of frames is different than the displayed order of frames, and the encoder and decoder must reorder the video frames, thereby increasing the total latency. In the example illustrated in Figure 3-3, there is one B-frame between each pair of I/P frames. Each frame is labeled with both its display order and transmission order. Although a B-frame is displayed between its adjacent I/P frames, it is transmitted out of sequence. This is so that the video decoder has the adjacent frames decoded and available for prediction. For a given macroblock within the B-frame, the encoder has four options for its prediction. They are: forward prediction, backward prediction, bi-directional prediction, and intra-frame coding. When bi-directional prediction is used, the forward and backward predictors are averaged and then subtracted from the target macroblock to form the prediction error. The prediction error is then transformed, quantized and transmitted in the usual manner.

3.3 Refreshing Options

A motion compensated prediction loop is not practical without some form of refreshing. This refreshing may be constant or variable and periodic. If a sequence of input images were perfectly predictable, the decoder would not receive any coefficient data and therefore could not reconstruct the picture after initialization. Decoder initialization occurs after the channel is changed, or the signal is lost and then recovered.

Consider the coding of a still image. The motion vectors and the prediction error would be zero. If the decoder were started with a sequence of zero prediction frames, then a blank or zero reconstructed frame would result. If the viewer changed to a channel transmitting a coded still image, the still image would never appear. A portion of the original picture must be spatially or temporally mixed with the prediction in the encoder to allow the decoder to synchronize to the encoder after decoder initialization.

The GA system satisfies the need for refreshing with two different mechanisms. I-frame refreshing uses periodic intra-coded frames. The advantages are that it provides clean insertion points in the compressed bit stream and nominal picture quality after acquisition (typically .5 seconds). The disadvantages are that it requires a larger bit buffer, it increases latency, and it complicates rate control.

Progressive refreshing uses periodic intra-coded (16x16) macroblocks. The advantages over I-frame refreshing are that it reduces the required buffer size, it simplifies rate control, and it reduces latency. One disadvantage is that motion vectors must be restricted in order to guarantee complete picture buildup after channel acquisition or channel errors.

3.4 Adaptive Field/Frame Processing

There are two options when processing interlaced video signals. The first option is to separate each frame into its two field components and then process the two fields independently (Figure 3-5). The second option is to process the two fields as a single frame by interleaving the lines of field 1 and field 2 (Figure 3-6).

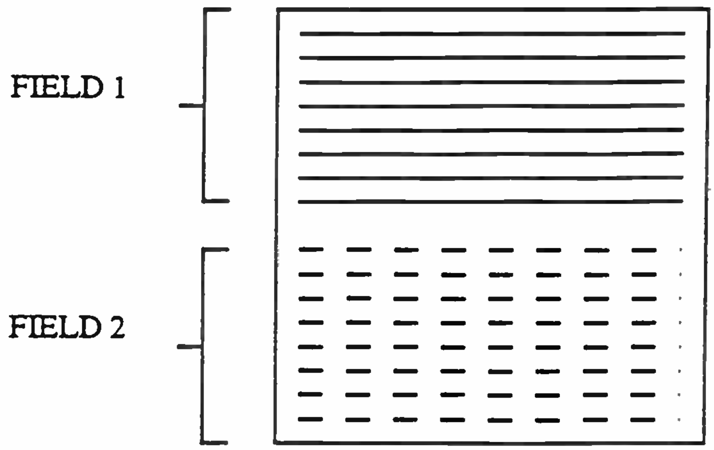

Figure 3-5. Field Processing

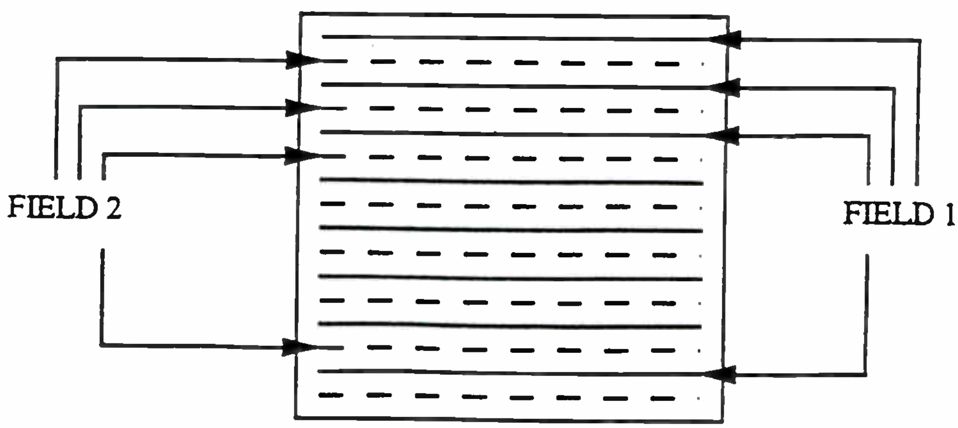

Figure 3-6. Frame Processing

Frame processing works better than field processing when there is little or no motion. Since each frame has twice as many lines or samples for a given picture height, there will be more correlation between samples and hence compressibility will be increased. Therefore, to achieve the same accuracy, field processing will require a higher bit rate, or alternatively, for equal bit rates, frame processing will achieve greater accuracy.

Similar advantages over field processing will be realized if horizontally moving features have little horizontal detail or if vertically moving features have little vertical detail. In other regions, where there is little detail of any sort, frame processing may still work better than field processing, no matter how rapidly changes occur.

Field processing generally works better than frame processing in detailed moving areas. In such cases, the interleaving of the first and second fields would introduce spurious high vertical frequencies if frame processing were used. This would reduce the correlation between lines and therefore the effectiveness of the compression process.

The GA system combines the advantages of both frame processing and field processing by adaptively selecting one of the two modes on a block-by-block basis. The mode which is selected is the one which maximizes correlation between adjacent lines within the block. Since the decision is made on a local basis, the system can adjust to scenes containing both moving and non-moving features, and therefore accurately reproduce vertical details in non-moving areas, and deliver good motion rendition in others.

The adaptive field/frame feature is only enabled when processing interlaced source material. Non-interlaced sources are always processed using frame mode.

3.5 Forward Estimation for Rate Prediction

Motion compensation, adaptive quantization and variable length coding produce highly variable amounts of compressed video data as a function of time. For example, the compressed bit-rate after a scene change can be several times greater than the bit-rate of the channel. Therefore, compressed video data buffering in the encoder and decoder is required for efficient channel utilization.

Buffer size is constrained by the maximum tolerable delay through the system and by cost. The fullness of the buffer is controlled by adjusting the amount of distortion or quantization error in each image. In the encoder, the buffer will fill more quickly if the distortion is low. A feedback control system is required to regulate the distortion level which controls the buffer fullness to prevent overflow. The design of this control system is complicated by:

- Delay between a change in the distortion level and the subsequent change in buffer fullness.
- Perceptual constraints on the instantaneous and averag distortion levels.
- Difficulties in modeling the bit-rate as a function of distortion level.

The visibility of distortion due to an increase in scene complexity is minimized by smoothly increasing the distortion level. This is facilitated by accurately modeling the rate versus distortion level since an accurate model allows the desired buffer fullness to be achieved for each frame. The model allows an estimate of complexity to be translated into a bit-rate for a given frame.

In the GA rate control system, the complexity estimate is the variance of the ideal (unquantized) displaced frame difference (DFD). Since the whole-pel motion vectors are available one frame ahead of the loop, the ideal displaced frame difference is calculated using whole-pel accuracy (i.e., no sub-pel estimation). The pixel values of the ideal DFD are squared and summed over the picture.

3.6 Adaptive Intra and Non-Intra Quantization Matrices

The MPEG-2 syntax allows the quantization matrices to be specified for every picture for improved coding efficiency. A certain probability distribution is associated with the VLC codes for quantized coefficients in the MPEG standard. Although one cannot change the VLC distribution to match the actual distribution of the data, the quantization matrices can be adjusted to help match the distribution of the data to the distribution of the VLC. Over the course of encoding the frame data, the variance of each spatial frequency band is calculated for both Intra and Non-Intra data. Applying upper and lower bounds per band to ensure reasonable operation in all cases, the desired quantization matrix can be derived.

Transmitting the quantizer matrices costs bits in the compressed data stream; if sent every picture in the 60 Hz progressive mode, the matrices consume 0.1% of the channel bandwidth. This modest amount of overhead is reduced by updating the quantization matrix at the start of each GOP, at least every 400 ms (for start-up), or when the difference between the desired quantizer matrix and the prevailing quantizer matrix becomes significant.

Sufficient compression cannot be achieved unless a large fraction of the DCT coefficients are dropped and therefore not selected for transmission. The coefficients which are not selected are assumed to have zero value in the decoder. The GA system encodes the selections and runs of zeros following a zigzag pattern through the array of frequency ordered coefficients. The DC coefficients are coded differentially to take advantage of high spatial correlation.

3.7 Perceptual Weighting and Coefficient Selection by Perceptual Sensitivity

The human visual system is not uniformly sensitive to coefficient quantization error. Perceptual weighting of each source of coefficient quantization error is used to increase quantization error in order to lower the bit-rate. The amount of visible distortion resulting from quantization error for a given coefficient depends on the coefficient number or frequency, the local brightness in the original image and the duration or the temporal characteristic of the error. DC coefficient error results in mean value distortion for the corresponding block of pixels which can expose block boundaries. This is more visible than higher frequency coefficient error which appears as noise or texture.

Displays and human visual systems exhibit non-uniform sensitivity to detail as a function of local average brightness. Loss of detail in dark areas of the picture is not as visible as it is in brighter areas. Another opportunity for bit savings is presented in textured areas of the picture where high frequency coefficient error is much less visible than in relatively flat areas. Brightness and texture weighting require analysis of the original image since these areas may be well-predicted in the DFD. Finally, distortion is easily masked by limiting its duration to one or two frames. This effect is most profitably used after scene changes where the first frame or two can be greatly distorted without perceptible distortion at normal speed.

Traditionally, a given coefficient is transmitted whenever its quantized level is non-zero. By selecting some non-zero coefficients for elimination, a fine level of quantization can be achieved on the remaining coefficients, potentially improving the overall picture. Perceptual selection is such a method.

Perceptual Selection is used in the GA system using the properties of the human visual system to code pictures using fewer bits within a perceptually consistent level of quality. For each transform block, the perceptual selection method determines the acceptable amount of distortion per each frequency band; if the magnitude of the coefficient in the loop is smaller than the acceptable distortion level, then the coefficient is set to zero, regardless of the quantization step size level.

3.8 Adapting M and N

The motion estimator range of the GA prototype system is ±32 V x ±128 H between two successive frames. The use of B-frames reduces the effective motion tracking range by a factor of M where M-1 is the number of B-frames between a given pair of I- or P-frames.

Scenes with rapid large area motion may exceed the range of the motion estimator if M is too high. This condition is detected in the forward analyzer which then signals the frame reorder section and motion estimator to use a lower M value for subsequent frames. The increased bit- rate caused by the temporary motion estimator overrun is absorbed by the rate buffer to maintain picture quality.

3.9 Film Mode

The GA system is able to detect source material originating from 24 fps film that has been converted to 60 fps using the 3:2 pulldown process. As shown in Figure 3-7, some fields (or some frames) are known to be identical due to the 3:2 pull down method. The GA system detects this redundancy through utilization of a film detection method that looks for the 3:2 pull down pattern. In the future, when the film mode is detected, the GA system removes the redundancy and converts it back to the original 24 fps source format prior to video encoding. The removal of redundancy ensures that the same information is not transmitted repeatedly, thus significantly improving the efficiency of the video compression system.

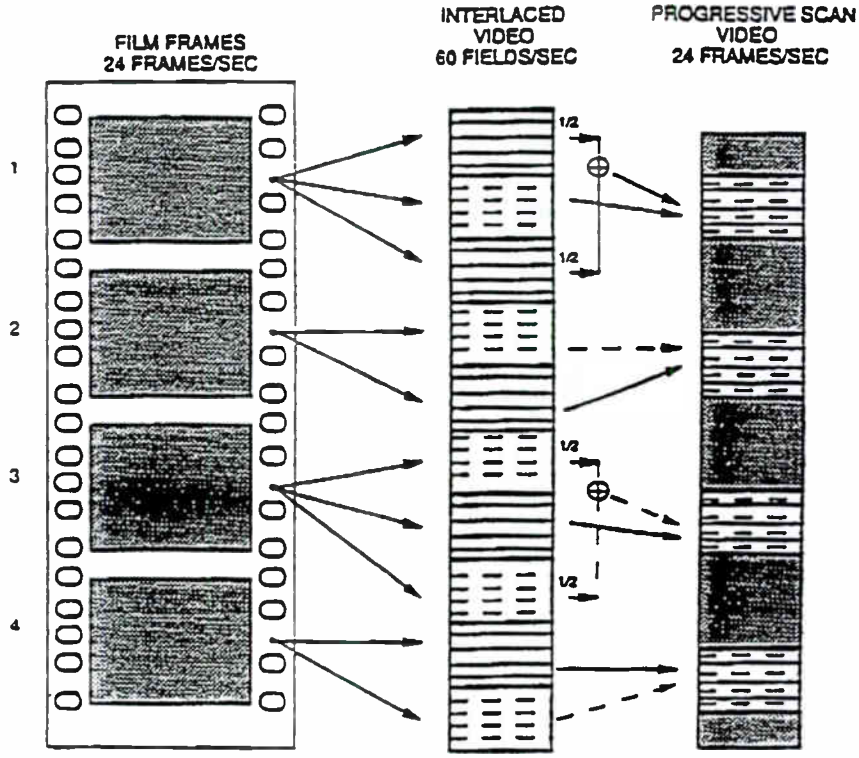

Figure 3-7. Conversion of Film to Interlaced Video and Restoration of 24 fps Progressive Scan

The GA system is also able to detect 30 fps material that has been converted to a 60 fpsprogressive scan format by simple frame repetition. When this conversion is detected, theencoder will discard the redundant frame and indicate to the decoder that each frame is to bedisplayed twice.

4. INTER-OPERABILITY

The GA compression algorithm is based on the emerging MPEG-2 compression standard. Consequently, the HDTV decoders will have resident decoder functionality to implement a device with a high degree of inter-operability. This inter-operability is a manufacturers option, and is not mandated.

4.1 Inter-operability with MPEG

The opportunities for inter-working are with MPEG-2 at High and High 1440 Level, CATV, DBS, and Teleco Standard Definition MPEG-2 and MPEG-1 services, and inter-working with MPEG-1 services such as those expected to be implemented for computer multimedia terminals. In each case, the HDTV decoder has the required decode engine and frame memory to decode all of these services. The implementation of image scaling for display, or multi-scan displays is left to the manufacturer.

The inter-operability choices with MPEG-2 are shown in Figure 4-1. Each line illustrates a potential inter-working scenario. The lines originate at the encoding side, with the arrow pointing to the decoder. A solid line indicates that inter-operability is achieved, while a dashed line indicates that the system is not inter-operable unless the manufacturer adds functionality to the decoding device. The lines labeled A and E show a US HDTV decoder inter-working with a MPEG-2 bit stream. Line A is dashed, while line B is solid, since the GA decoder is a super set of an MPEG-2 decoder. In case additional syntax elements are added to the GA system in the future to improve the performance, then line B becomes dashed and an MPEG-2 decoder will not decode the GA bit streams unless the manufacturer adds the additional functionality to the decoding device. Line C is also dashed, since there is not yet a requirement that a GA television be able to decode and display a compressed digital standard definition bit stream. This level of inter-operability does not require any change to the decoding engine, since MPEG-2 is a subset of the GA algorithm. It will require that the display interpolation functionality be resident in the GA decoder. There is no line from the GA system to the MPEG-2 main because MPEG-2 main level decoder will not be able to decode the GA bit stream due to memory and speed constraints.

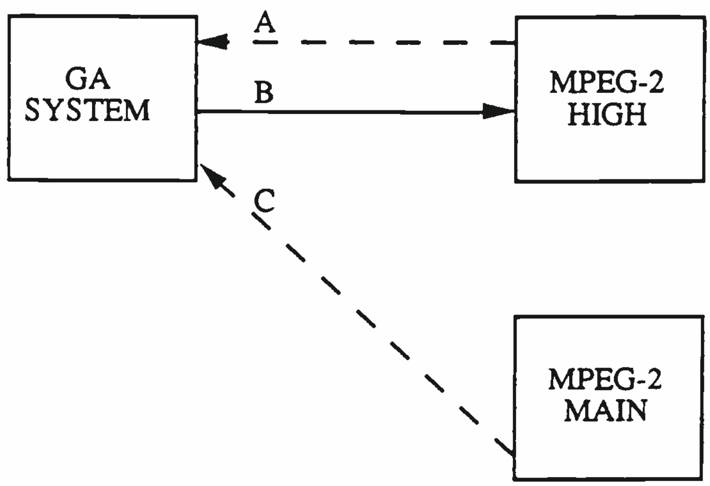

Figure 4-1. Inter-operability between GA and MPEG-2

4.2 Inter-operability with Computers

The GA system will inter-operate with computer displays in a similar manner, as shown in Figure 4-2. Here, the bit stream inter-operates at either the Transport Layer or the Compression Layer. The compression layer will syntactically inter-operate, however, the computer will require some display processing to convert the video frame-rate assuming a computer display refresh rate of * 66 Hz as is commonly the case. The windowing system will also manage the format conversion between GA video formats and the display format native to the computer. The format conversions required will be: frame-rate conversion, picture aspect ratio (16:9 versus 4:3), interlaced to progressive (when required), and pel aspect ratio (when required).

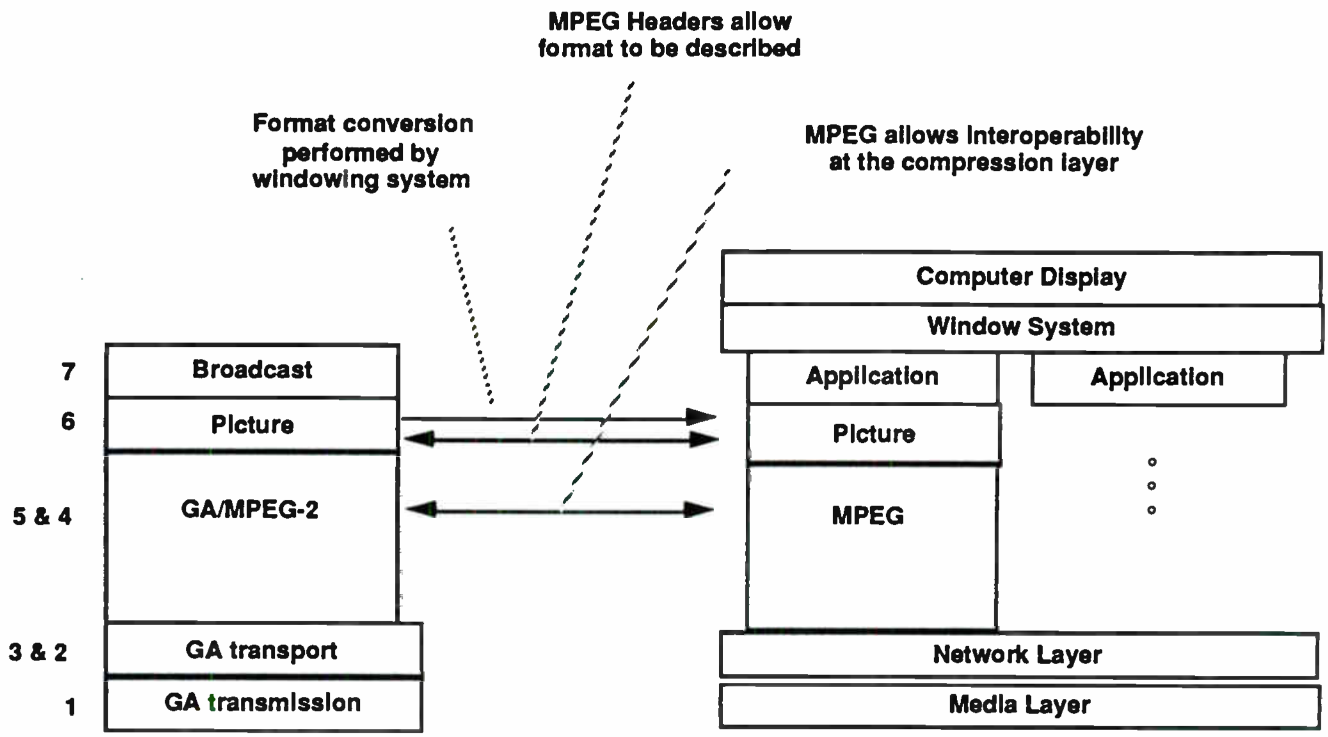

Figure 4-2. GA Inter-operability with Computers

Chapter IV

AUDIO COMPRESSION SYSTEM

1. Introduction

The AC-3 audio coding technology is a sophisticated and flexible system for the digital representation of high quality (including multi-channel) sound. A wide range of bit rates and audio coding modes are supported. Features have been incorporated which allow the coded AC-3 data stream to service nearly all listeners, not just those with a particularly ideal listening situation. A section of the data stream referred to as *bit stream information* (bsi) has a number of fixed and optional fields that describe the coded bit stream in some detail. Some of these fields, concerned with audio levels and types of audio production facilities, allow coded bit streams from different sources to be decoded with improved uniformity of level and intended frequency response.

2. Technical Details

2.1. Overview

An AC-3 serial coded audio bit stream is made up of synchronization frames containing 6 coded Audio Blocks (AB), each of which represent 256 audio samples. A Synchronization Information (SI) header at the beginning of each frame contains information needed to acquire and maintain synchronization. A Bit Stream Information (BSI) header may follow SI, and contains parameters describing the mode of the coded audio service(s). The Audio Blocks may be followed by an Auxiliary Data (Aux Data) field.

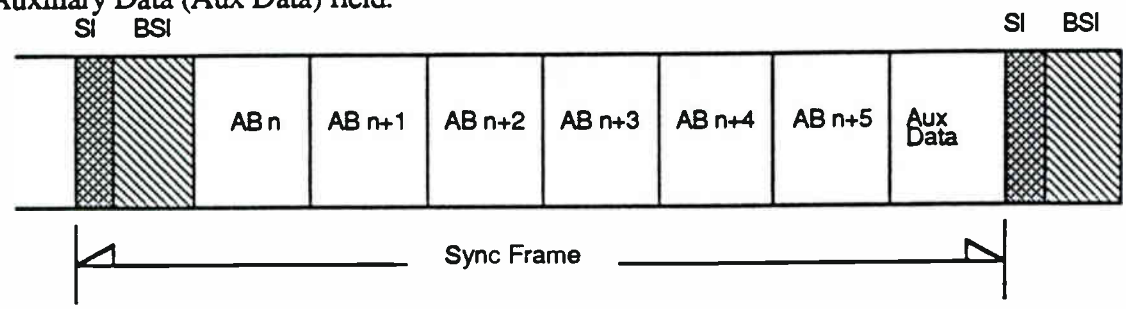

Figure 1 AC-3 Data Stream

2.2. Filterbank

AC-3 is a transform coder (see Figure 2) using the Time Domain Aliasing Cancellation (TDAC) transform. The filterbank is critically sampled, and of low computational complexity. Each input time point is represented in two transforms by means of the 50% overlap/add windows. The Fielder window is used which offers an optimum trade-off of close in frequency selectivity and far away rejection. The block size is 512 points which is the optimal transform length for most program material, being a good trade-off between frequency resolution (which determines coding gain and thus minimum bit rate), and complexity (primarily the buffer lengths and memory requirement in the decoder). The frequency resolution of the AC-3 filter bank is 93 Hz, uniform across the spectrum.

The 512-point transform is done every 256 points, providing a time resolution of 5.3 ms (at 48 kHz sampling rate). This time resolution is sometimes insufficient depending on the temporal characteristics of the input signal. During transients a finer time resolution is needed in order to prevent pre-noise artifacts at low bit-rates. The encoder controls the transform length, and during transients will switch to a 256-point transform for a time resolution of 2.7 ms. The algorithm which determines the optimum transform length resides only in the encoder, and may be improved or updated without affecting the decoders already in the field. The transition between blocks is seamless, and the aliasing cancellation of the transform is maintained along with critical sampling. During block switching, there is no increase in filterbank computation rate.

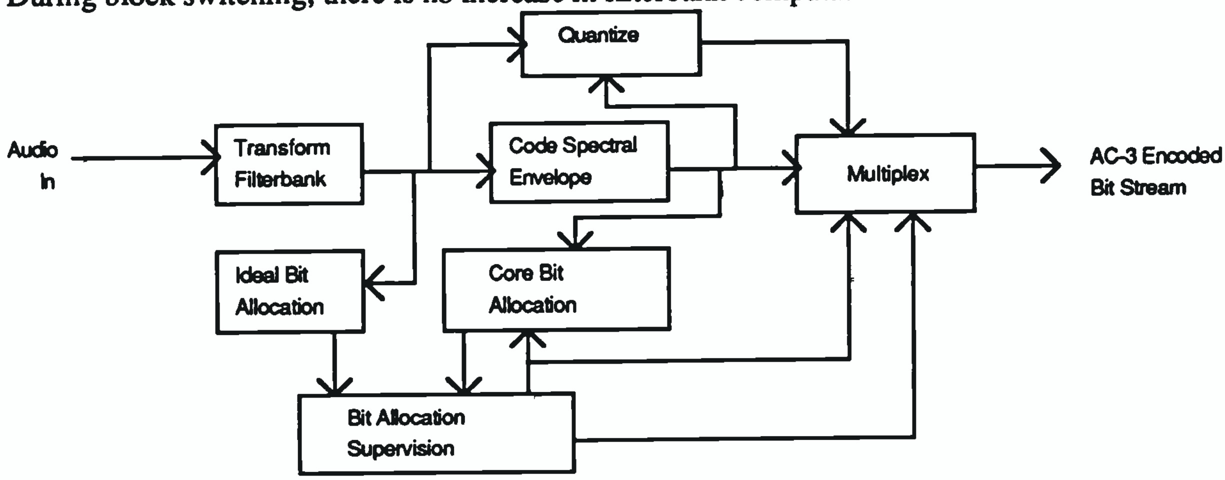

Figure 2 AC-3 Encoder

2.3. Spectral Envelope

The output of the 512-point TDAC transform gives 256 real-valued frequency coefficients. The logarithm of the absolute values of these coefficients forms the spectral envelope. The spectral envelope is coded with time and frequency resolution which is signal dependent. The frequency resolution of the spectral envelope can vary from the fundamental filterbank resolution of 93 Hz, up to 750 Hz, depending on the signal. The time resolution of the spectral evelope can vary from 5.3 msec to 32 msec, and is signal dependent. The algorithm which determines the time and frequency resolution of the spectral envelope is in the encoder, and may be updated or improved without affecting the decoders already in the field. The encoder decodes the encoded spectral envelope, so as to make use of the identical information that is available to the decoder. The decoded representation is used both as a reference for the quantization of the transform coefficients, and to drive the bit allocation routine.

2.4. Bit Allocation

The decoded spectral envelope has sufficient precision to be used for a psychoacoustic based bit allocation procedure. The encoder performs an iterative bit allocation routine altering the noise to mask ratio of all channels until the number of available bits is used up. Key hint information is coded and transmitted to the decoder which performs the core bit allocation routine only once and arrives at the identical bit allocation which was obtained by the encoder. While not as ideal as a bit allocation routine that uses the full knowledge available at the encoder, this approach saves the overhead required to explicitly transmit the bit allocation to the decoder, and allows more frequent updating of the allocation to be performed. The bits are allocated to each channel out of a common bit pool. Each individual channel may be allocated a differing number of bits.

The bit allocation may be modified by one of two methods. First, parameters of the psycoacoustic model may be altered by side information which is transmitted from the encoder to the decoder. These psycoacoustic model parameters are used in the core bit allocation routine which is computed in both the encoder and decoder. Second, the encoder may optionally send additional side information in order to explicitly alter some or all of the allocation values derived in the core bit allocation routine.

An AC-3 encoder may compute an ideal bit allocation based on full knowledge of the input signal, and compare the results of this allocation to that of the core routine. If the encoder determines that improved subjective results can be obtained by altering the parameters of the psycoacoustic model used by the core routine, it can do so and send the appropriate side information to the decoder. If the results of the core routine are still sub-optimal, the encoder may send additional side information to explicitly modify the output of the core bit allocation routine.

Since the bit allocation is dependent on the spectral envelope, the bit allocation has time and frequency resolution equal to that of the spectral envelope. The bit allocation may change as often as every 5.3 msec, and the frequency resolution may be as fine as 93 Hz.

The core bit allocation routine used by AC-3 only requires the use of a rudimentary ALU with a pair of 16 bit accumulators. Only simple instructions such as ADD, SUBTRACT, COMPARE, AND, OR, and conditional branch are required. The psychoacoustic model used by the AC-3 encoder may be changed at any time without requiring a change in the installed base of decoders. The performance of AC-3 may thus improve over time as more knowledge is gained about the human auditory system.

2.5. Channel Coupling

At high frequencies the human ear determines directionality based on the signal envelope rather than on the particular signal waveform. Additionally, the human auditory system cannot independently detect directionality for multiple signals which are very close together in frequency. During periods of high bit demand, AC-3 takes advantage of these phenomena by selectively "coupling" (a form of combination) channels together at high frequencies. The method used maintains, within a critical band (a narrow region of frequency), the original powers of both the individual signals and the individual speaker outputs in order to avoid audible artifacts.

Coupling derives coding gains by combining number of individual channel transform coefficients (at a given frequency) into a common coefficient (at that frequency). The combinations only occur at high frequencies, above the "coupling frequency". The set of combined coefficients form the "coupling channel". Only the spectral envelope of the coupling channel is transmitted (instead of all of the individual spectral envelopes), along with the quantized values of the coupled coefficients (instead of all of the individual coefficients). The coupled coefficients are formed into bands, with widths similar to critical band widths of the human auditory system. For each band in each coupled channel a "coupling coordinate" is also transmitted. The coupling coordinate is used by the decoder to reproduce each coupled band out of each coupled channel loudspeaker at the original signal power level.

The channels which are coupled, the threshold frequency above which coupling takes place, and the widths of the coupled bands is determined by the encoder, and may dependent on the audio program content. The encoder algorithms used to determine these items may be updated at any time without requiring changes to decoders already in the field.

2.6. Synchronization and Acquisition

The AC-3 bit stream syntax has been designed to allow rapid synchronization and acquisition. The sync word is 16 bits long and has good autocorrelation properties. The probability of randomly finding a sync within the encoded data in a 32 msec sync frame is less than 3% (assuming byte level registration of the transport system). The first sync found will be correct 97% of the time, and it will be found within 32 msec. The sync frame may be verified by use of an embedded CRC check. If the CRC checks, one can be assured that true sync has been achieved, and decoding of audio may begin.

3. Features

3.1. Rates and Modes

The AC-3 algorithm supports industry standard sample rates of 48 kHz, 44.1 kHz, and 32 kHz. The Grand Alliance system may not support the use of more than one sample rate. Nominal bit rates are indicated by a bit field which indicates (via table lookup dependent on sample rate) the number of bytes between sync codes. Nineteen nominal bit rates of 32, 40, 48, 56, 64, 80, 96, 112, 128, 160, 192, 224, 256, 320, 384, 448, 512, 576, and 640 kb/s are defined in the AC-3 algorithm, with an additional 13 rate codes reserved for future definition. Again, the Grand

Alliance system may not support the use of all available bit rates. The main audio service mode can range from simple monophonic (1/0) through stereo (2/0) and up to all combinations of multiple channels (2/1, 2/2, 3/0, 3/1, 3/2). Any of the modes may optionally use a low frequency effects channel (0.1 channel).

3.2. Error Handling

The AC-3 syntax includes the use of 16 bit CRC checks for error detection. Some implementations of receiver architecture allow error flags to be delivered to the audio decoder to identify portions of the bit stream which may contain errors. Concatenation of the use of error flags and CRC checking is a useful technique to reduce the frequency of reproduced errors.
The audio decoder must handle an occasional errored packet or dummy packet without significant audio disturbance. In the event of an error, previous audio data may be repeated. The window used for the FFT 50% overlap/add process is a very good window to crossfade the repeated time segment into the output time waveform. The implementation of error concealment does require some additional memory in the decoder, since old data must be kept available until a new data is verified as error free. Since the input data rate to the decoder is much lower than the output data rate, it is most efficient to keep a section of the previous input data.

3.3. Program Related Information

Several types of static program related information are contained in the bit stream information section of the audio frame. Program related information which is dynamic (such as dynamic range control) is coded into each individual audio block.

3.3.1. Center Mix Level

When a program with three front channels is mixed down to one or two channels for mono or stereo reproduction, artistic considerations sometimes require a different mixture of the Center channel signal with respect to the Left and Right signals. In order that the original multi-channel program mix not be compromised for the sake of mono or stereo reproduction, the AC-3 syntax allows different center channel downmix coefficients to be indicated to the decoder.

3.3.2. Surround Mix Level

When multi-channel sound which contains surround information is mixed down to stereo or mono, the optimum mixing level of the surround content which is mixed into the output channels may vary depending on the program type. The AC-3 syntax allows differing levels for the surround mixing coefficients to be indicated to the decoder.

3.3.3. Dialogue Level

In order to assure that different AC-3 encoded programs reproduce a uniform level of dialogue, there is an indication of dialogue level encoded into the data stream. Audio decoders make use of this information in order to reproduce all encoded programs with uniform loudness, consistent with the requirements of ATSC document T3-186.

3.3.4. Dynamic Range Control

The coded audio blocks contain a dynamic range control word which is used to modify the output level of the decoder. This word is encoded by the broadcaster or production studio in order to intentionally restrict the dynamic range of the reproduced sound to that which is suitable for the broad audience. The AC-3 decoder is required to, by default, reproduce audio with the intended (by those artistically responsible for the program) restricted dynamic range. The decoder may permit the listener the option to restrict the use of the dynamic range control word during reproduction, which will have the effect of reproducing more (or all) of the original program dynamic range. The coded AC-3 bit stream can thus serve a wider range of audience, including those who require restricted dynamic range reproduction as well as those who wish to experience the full dynamic range of the original audio production.
The amplitude resolution of the dynamic range gain control word is less than 0.25 dB. This resolution, coupled with the smoothing effect of the gentle overlapping blocks of the TDAC alternating transforms, insures that there will be no audible modulation products generated by

changes in the control word. During transient events, the control word may be updated as often as every 2.6 msec. The encoder may use any amount of time look ahead practical (including off-line disc based processing) to generate the most intelligent gain control words. Any algorithm may be used to generate these words (even direct manual control) and all decoders in the field will respond correctly.

3.3.5. Production Mix Room Level, Type

Optional fields allow the indication of the type of mixing room used to prepare the program (large cinema or small studio, resulting in slightly differing equalization) as well as the absolute sound pressure level (SPL) of the mixing session. The SPL code may be used along with the dialogue level indication to allow the reproduction of the program at the identical level used during the mixing session. This information may also be used to adjust the reproduced frequency response to compensate for acoustic reproduction at a different level SPL level so as to maintain the same perceived frequency response (true loudness compensation based on differential equal loudness hearing contours).

3.3.6. Time Synchronization

The AC-3 data stream may be tagged with an hour, minute, second, frame, and fractional frame time code which is useful for synchronization with SMPTE time code. This information is useful to keep audio and video bit streams in sync in the production environment, and may be useful in the home environment as systems become more complex and interoperable.

3.3.7. Additional Information

Additional information may be appended to bit stream information and will be ignored by decoders not intended to recognize it.

3.4. Associated Services

The AC-3 syntax and the Grand Alliance transport system allow AC-3 bit streams to convey associated services. Each associated service may be one of the following types: visually impaired, hearing impaired, commentary, dialogue, or emergency flash message. Each associated service may have an optional indication of language.

3.5. Auxiliary Data

If the audio data rate is restricted to less than the indicated rate, the excess capacity will result in unused bits at the end of the data frame. This auxiliary data will be ignored by the audio decoder, and may be used for any purpose.

4. Compatibility

4.1. Compatibility with Mono and Stereo Reproduction

The multi-channel AC-3 bit-stream can serve listeners requiring mono and stereo outputs. When fewer than the full number of output channels is required, the particular down mix is done in the decoder and can be optimum for each listening situation. The program originator can indicate which downmix coefficients are appropriate for a given program. The down mix may be done in the frequency domain, so that only the desired number of output channels needs to be transformed back into the frequency domain and buffered for output. This saves on complexity for AC-3 decoders which only have mono or stereo outputs.

4.2. Compatibility with Dolby Surround Encoded Programs

When operated in the 2/0 mode, AC-3 will convey 2 channel Dolby Surround matrix encoded soundtracks without degrading the performance subsequently obtained when further decoding the 2 signals into 4 signals with a Dolby Pro Logic surround decoder. The AC-3 coder makes use of internal re-matrixing in order to avoid the unmasking of coding artifacts which may occur when matrix decoding. Re-matrixing places the dominant coding error in the same location as the dominant signal energy, and avoids the case of a loudspeaker reproducing only coding artifacts which may not be masked by signals from other loudspeakers. This feature is especially important when operating at low data rates, such as 2/0 mode at 192 kb/s.

4.3. Compatibility with Dolby Surround Decoders

The AC-3 decoder can provide a 2 channel stereo output from a multi-channel bit stream which is compatible with Dolby Surround matrix decoding. This is a useful feature due to the large number of surround decoders already in use in the field which only have 2 channel inputs.

5. ATSC Document T3-186

Portions of the ATSC document T3-186 are concerned with audio coding. AC-3 has been designed with the ATSC concerns in mind, and satisfies all of the requirements of T3-186. The audio related sections of T3-186 are referenced below, and the methods by which AC-3 satisfies the requirements are described.

5.1. T3-186 Section 1.5.4, Multi-Channel Audio

This section states the requirement for 5 channel coding, with the low frequency enhancement channel optional. Monophonic and two channel stereophonic modes are also required, along with the ability to decode the full 5 channel service into stereo or mono. AC-3 offers all of the necessary coding modes, which are further elucidated below. Lower cost mono or stereo AC-3 decoders may be built which only partially decode a full 5.1 channel data stream, mix the channels down in the frequency domain, and perform the inverse filter bank only on the needed output channels (see section 7 on complexity).

5.2. T3-186 Section 1.5.5, Multiple Languages

This section states the requirement to handle multiple languages. Multiple AC-3 data streams may be provided, each conveying a main service in a different language. The AC-3 bit stream information syntax allows a main audio service to be tagged with an internationally recognized language code. Associated services such as commentary and dialogue may be conveyed by AC-3 data streams and each of these may also be tagged with a language code.

5.3. T3-186 Section 1.5.6, Audio Services to the Visually and Hearing Impaired

The section states the requirement to allow associated services such as VI or HI to be provided along with a main service. The AC-3 syntax and Grand Alliance transport system allows single channel associated services to be delivered. These services can be tagged as to type: visually impaired, hearing impaired, or commentary.

5.4. T3-186 Section 1.5.7, Uniform Loudness

This section deals with uniform loudness. The AC-3 data stream contains a field for a reference level indication, and the decoder must use this information to adjust the reproduce level. This allows the inter-cutting of bit streams from different sources with different loudness calibrations and headroom without undesirable level variations occurring in the level of reproduced dialogue.

5.5. T3-186 Section 1.5.8, Dynamic Range Control

This section considers dynamic range control. AC-3 contains an integral dynamic range control system. The broadcast encoder, or another piece of equipment, may insert codes into the data stream which the decoder uses to dynamically alter the reproduced level. This allows the broadcaster to intentionally compress the dynamic range, but allows the listener the option to reproduce either the compressed or the original (or something in-between) dynamic range.

5.6. T3-186 Section 1.5.11, Error Correction and Concealment for Audio Services

The section directs the audio system to take effective methods to minimize audible disturbances caused by uncorrectable errors. The AC-3 data stream contains error detecting codes which will allow the AC-3 decoder to determine if the audio data is valid. If the audio data is not valid error concealment is performed, in which a previous block of data is decoded instead of the current errored block. The repeated block is innocuously spliced in with the gentle FFT window function and this process is typically inaudible. In the case of a television set, the transport layer may deliver error flags to indicate that certain packets of data may be in error. In the event entire AC-3 bit stream is removed from the ATV set and delivered to another piece of equipment (interoperability), the error flag may no longer be available. In this case, the final AC-3 decoder can still perform error concealment triggered by the imbedded error detecting codes.

5.7. T3-186 Section 3.1.1, Independent Coding Modes

AC-3 can support all of the following bit-rates for an independently coded channel.
Independent 1/0: 32 kb/s, 40 kb/s, 48 kb/s, 56 kb/s, 64 kb/s, 80 kb/s, 96 kb/s, 112 kb/s, 128 kb/s, 160 kb/s, 192 kb/s, 224 kb/s, 256 kb/s, 320 kb/s, 384 kb/s.
(Note: for reasons of simplicity, the Grand Alliance system may not require decoders to recognize all modes, and thus a subset of these may be selected. A reasonable subset would those suggested by the ATSC: 64kb/s, 96 kb/s, and 128 kb/s although some consideration will be given to an additional lower rate.)

5.8. T3-186 Section 3.1.2, Composite Coding Modes

AC-3 can support the following composite coding modes at bit-rates exceeding the recommended minimum indicated below. Any of the modes may optionally support the low frequency effects channel.

Composite 3/2: ≥ 256 kb/s
Composite 3/1: ≥ 256 kb/s
Composite 3/0: ≥ 192 kb/s
Composite 2/2: ≥ 256 kb/s
Composite 2/1: ≥ 192 kb/s
Composite 2/0: ≥ 112 kb/s

5.9. T3-186 Section 3.1.2, Associated Audio Data

The reference level field in the AC-3 data stream (discussed under 1.5.7 above) satisfies the 3.1.3.1 headroom requirement. The dynamic range control codes (discussed under 1.5.8 above) satisfies the 3.1.3.2 dynamic range requirement. The AC-3 data stream has optional fields available for additional information satisfying the 3.1.3.3 requirement for descriptor information, and the 3.1.3.4 requirement for user bits.

6. AC-3 Decoder Architecture

An AC-3 decoder may be efficiently implemented by five functional blocks, as shown in Figure 4. The first block contains an 8-bit serial-parallel converter and input buffer RAM. The coded AC-3 data stream enters the decoder serially and is converted to bytes and stored in RAM.

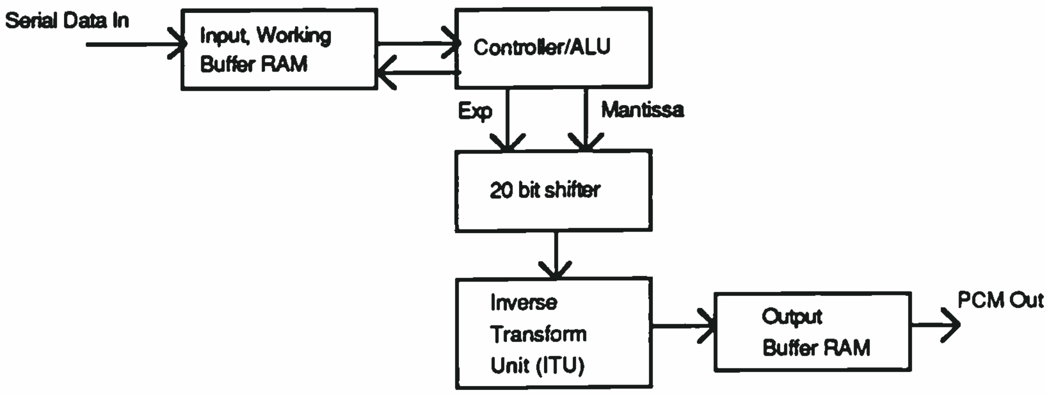

Figure 4 AC-3 Decoder

The second block contains a simple Controller/ALU incorporating two 16-bit accumulators, but no multiplier. The ALU requires the following instructions: add, subtract, compare, and, or, xor, conditional branch, and shift. The third block contains a semi-logarithmic shifter capable of implementing a 0 to 20-bit right shift on mantissas prior to inverse transformation. The semi-log design saves routing area compared to a straightforward 2^5-bit log-shifter. The Controller/ALU coordinates the input buffer and the 20-bit shifter. These three blocks perform the following functions:

1. Accept bit stream input interrupts from the serial-to-parallel converter and load the RAM.
2. Determine frame boundaries (frame synchronization).
3. Unpack control data.
4. Decode the spectral envelope.
5. Perform the core bit allocation routine.
6. Unpack mantissas, right shift up to 20 bits, and deliver data to the ITU.

The fourth circuit block, the Inverse Transform Unit (ITU), performs the frequency-to-time domain conversion (reconstruction synthesis filterbank). The ITU incorporates its own controller circuitry, and may operate somewhat autonomously from the previous circuit elements. Audio blocks representing 256 audio samples are inverse transformed using an efficient fast TDAC transform technique which minimizes computation and RAM requirements. The ITU requires the following instructions: multiply, multiply-add, and multiply-subtract.

Chapter V
TRANSPORT SYSTEM

The Grand Alliance Transport System

Table of Contents

The Grand Alliance Transport System

1. Introduction

This document provides a description of the functionality and format of the Grand Alliance transport system. While tutorial in nature, the document provides sufficient technical detail to serve as the operational specification of the transport layer. Issues related to terrestrial broadcast and cable delivery of the ATV service, in the context of ACATS discussions, are addressed in this document. The authors have attempted to make this a stand-alone document, though the reader would gain additional insight from associated ISO-MPEG documents on this and related topics.

In developing the specification of the Grand Alliance transport layer, we have drawn upon the collective experience of the member companies in developing individual systems, as well as the excellent body of work created by the ISO-MPEG standards process. While any system design requires intelligent tradeoffs to be made, selection of a format based on fixed-length packets has maintained a number of simultaneous goals.

1.1. Program vs. Transport stream multiplexing

In general there are two approaches for multiplexing elementary bit streams from multiple applications on to a single channel. One approach is based on the use of fixed length packets and the other on variable length packetization. Both approaches have been used in the MPEG-2 standard. As illustrated in Fig. 1.1, the video and audio elementary streams in both cases go through an initial stage of PES packetization (discussed in greater depth later), which results in variable length PES packets. The process of generating the transmitted bit streams for the two approaches is shown to involve a difference in processing only at the final multiplexing stage.

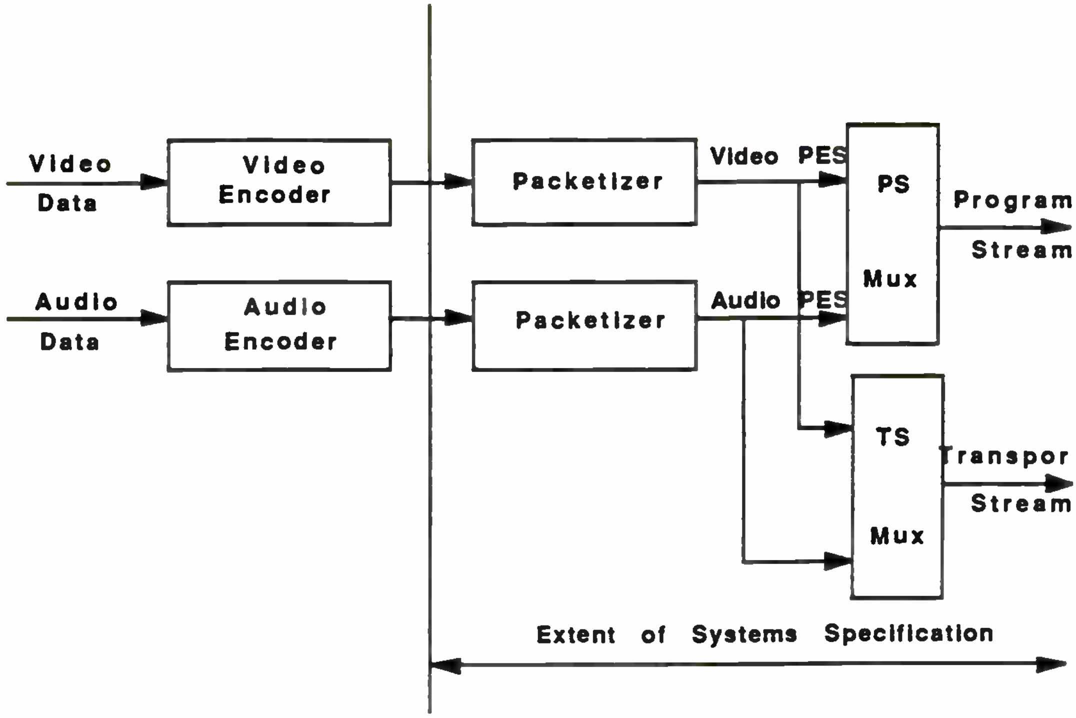

Fig 1.1. Comparison of system level multiplexing approaches.

Fig. 1.2 gives examples of bit streams for the both program and transport stream approaches, in order to clarify their difference. As shown in Fig. 1.2b, in a program stream approach, PES packets from various elementary bit streams are multiplexed by transmitting the bits for the complete PES packets in sequence, thus resulting in a sequence of variable length packets on the channel. (As shown in the diagram, each PES packet is preceded by a PES packet header.) In contrast to this approach, in the transport stream approach selected for the GA system, the PES packets (including the PES headers) are transmitted as the payload of fixed length transport packets. Each transport packet is preceded by a transport header which includes information for bit stream identification. As illustrated in Fig. 1.2a, each PES packet for a particular elementary bit stream occupies a variable number of transport packets, and data from various elementary bit streams are generally interleaved with each other at the transport packet layer, with identification of each elementary bit stream being facilitated by data in the transport headers. New PES packets always start a new transport packet in the GA system, and stuffing bytes are used to fill packets with partial PES data.

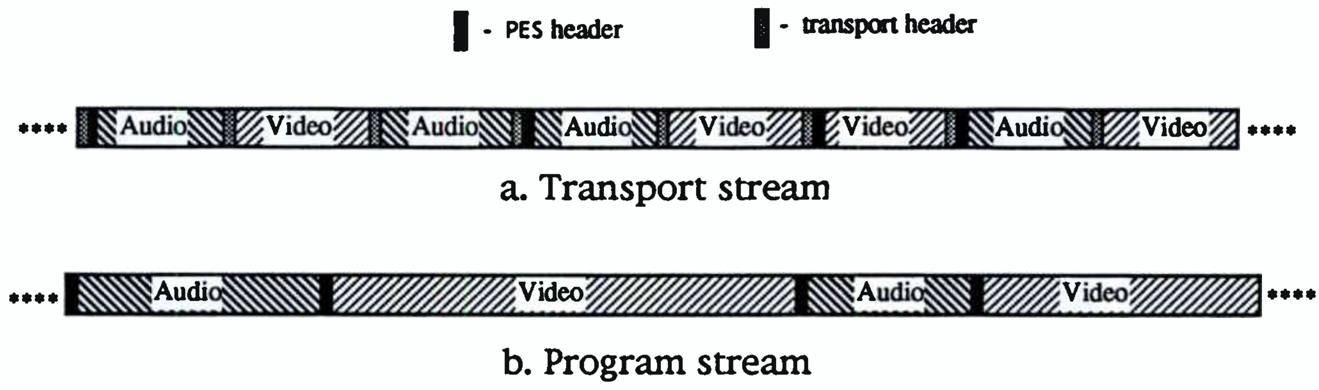

Fig. 1.2. Illustration of bit streams for the two packetization approaches.

The two multiplexing approaches are motivated by completely different application scenarios. Transport streams are defined for environments where errors and data loss events are likely, including storage applications and transmission on noisy channels. Program streams on the other hand are designed for relatively error-free media, e.g. CD-ROMs. Errors or loss of data within PES packets can potentially result in complete loss of synchronization in the decoding process in this case. The definition of program stream approach within MPEG-2 is also motivated by the requirement for compatibility with MPEG-1.

The transport stream approach of MPEG-2 has been found to support most of the functional requirements for the GA system, and hence forms the basis of the GA system definition. As will become clearer from discussions in the following sections, the variable packet-size based program stream approach does not meet these requirements in many aspects. Furthermore compatibility with MPEG-1 systems (which is based on the program stream concept) is not a concern for the GA. Note that the GA system transport system can still identify and carry MPEG-1 video and audio services. In general, the program and transport streams both address the same general layers of protocol functionality and therefore it does not make much sense to attempt to carry a program bit stream within a transport stream or vice-versa. Transcoding between the two formats is however feasible and one could in theory build an interface that connects from a GA bit stream to a program stream decoder. The need for such functionality is not anticipated.

Another point to note is that in other ATV scenarios such as CATV and DBS, defacto standards are being set based on the use of the fixed length packetization approach. The approach for the GA system is consistent with the need for simple interoperability with these scenarios.

1.2. Advantages of the fixed length packetization approach

The fixed length packetization approach offers a great deal of flexibility and some additional advantages when attempting to multiplex data related to several applications on a single bit stream. These are described in some detail in this section.

Dynamic Capacity Allocation:

While digital systems are generally described as flexible, the use of fixed length packets offers complete flexibility to allocate channel capacity among video, audio and auxiliary data services. The use of a packet id (or PID) in the packet header as a means of bit stream identification makes it possible to have a mix of video, audio and auxiliary data which is flexible and which need not be specified in advance. The entire channel capacity can be reallocated in bursts for data delivery. This capability could be used to distribute decryption keys to a large audience of receivers during the seconds preceding a popular pay-per-view program, or download program-related, computer software to a "smart receiver."

Scalability:

The transport format is scalable in the sense that availability of a larger bandwidth may also be exploited by adding more elementary bit streams at the input of the multiplexer, or even multiplexing these elementary bit streams at the second multiplexing stage with the original bit stream. This is a critical feature for network distribution, and also serves interoperability with a cable plant's capability to deliver a higher data rate within a 6 MHz channel.

Extensibility:

Because there will be possibilities for future services that we cannot anticipate today, it is extremely important that the transport architecture provide open-ended extensibility of services. New elementary bit streams could be handled at the transport layer without hardware modifications, by assigning new packet IDs at the transmitter and filtering on these new PIDs in the bit stream at the receiver. Backward compatibility is assured when new bit streams are introduced into the transport system since existing decoders will automatically ignore new PIDs. This capability could possibly be used to compatibly introduce "1000-line progressive formats" or "3D- HDTV" by sending augmentation data along with the normal ATV data.

Robustness:

Another fundamental advantage of the fixed length packetization approach is that the fixed length packet can form the basis for handling errors that occur during transmission. Error correction and detection processing (which precedes packet demultiplexing in the receiver subsystem) may be synchronized to the packet structure so that one deals at the decoder with units of packets when handling data loss due to transmission impairments. Essentially, after detecting errors during transmission, one recovers the data bit stream from the first good packet. Recovery of synchronization within each application is also aided by the transport packet header information. Without this approach, recovery of synchronization in the bit streams would have been completely dependent on the properties of each elementary bit stream.

Cost Effective Receiver Implementations:
A fixed-length packet based transport system enables simple decoder bit stream demultiplex architectures, suitable for high speed implementations. The decoder does not need detailed knowledge of the multiplexing strategy or the source bit-rate characteristics to extract individual elementary bit streams at the demultiplexer. All the receiver needs to know is the identity of the packet, which is transmitted in each packet header at fixed and known locations in the bit stream. The only important timing information is for bit level and packet level synchronization.

MPEG-2 Compatibility:
The GA transport system is based on the MPEG-2 system specification. While the MPEG-2 system layer has been designed to support many different transmission and storage scenarios, care has been taken by MPEG, as well as the Grand Alliance, to limit the burden of protocol inefficiencies caused by this generality in definition.

An additional advantage of MPEG-2 compatibility is interoperability with other MPEG-2 applications. The MPEG-2 format is likely to be used for a number of other applications, including storage of compressed bit streams, computer networking, and non-HDTV television delivery systems. MPEG-2 transport system compatibility implies that GA transport bit streams may directly be handled in these scenarios (ignoring for the moment the issue of bandwidth and processing speed).

While the GA transport format conforms to the MPEG-2 systems format, it will not exercise all the capabilities defined in the MPEG-2 transport. Therefore, a GA System decoder need not be fully MPEG-2 systems compliant, in that it will not be able to decode any arbitrary MPEG-2 systems bit streams. However, all MPEG-2 decoders should be able to decode the GA bit stream syntax at the transport system level. Documents defining the extent to which the MPEG capabilities are supported in the GA transport have been submitted to the MPEG committee and have contributed to the current working draft of the standard. (See Attachment 1.) MPEG-2 standard features not supported in the GA specification were constrained[1] if they were deemed to not be applicable to broadcast/cable delivery of ATV.

In the development of the GA transport specification, the intent has never been to limit the design by the scope of the MPEG-2 systems definition. If the MPEG-2 standard is unable to efficiently meet the requirements of the GA system, a deviation from MPEG would be in order. The Advisory Committee would be notified should there be a future deviation from MPEG, with justification for the change.

1.3. Overview of the transport subsystem

Fig. 1.3. illustrates the organization of a GA transmitter-receiver pair and the location of the transport subsystem in the overall system. The transport resides between the application (e.g. audio or video) encoding/decoding function and the transmission subsystems. At its lowest layer, the encoder transport subsystem is responsible for formatting the encoded bits and multiplexing the different components of the program for transmission. At the receiver, it is responsible for recovering the bit streams for the individual application decoders and for the corresponding error signaling. (At a higher layer, multiplexing and demultiplexing of multiple programs within a single bit stream can be achieved with an additional system level multiplexing or demultiplexing stage before/after the modem in the transmitter/receiver.) The transport subsystem also incorporates other higher level functionality related to identification of applications and, as illustrated, synchronization of the receiver. This document will describe these functions in greater detail.

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[1] The constraint takes the form of a limitation of functionality. In this instance, certain flags will be permanently configured, and some fields will not appear in the Grand Alliance bitstream. This allows a simpler decoder, as it will not be necessary to handle elements not used for the Grand Alliance application. Often constraints are grouped as a "profile" when acknowledged by the MPEG standards body.

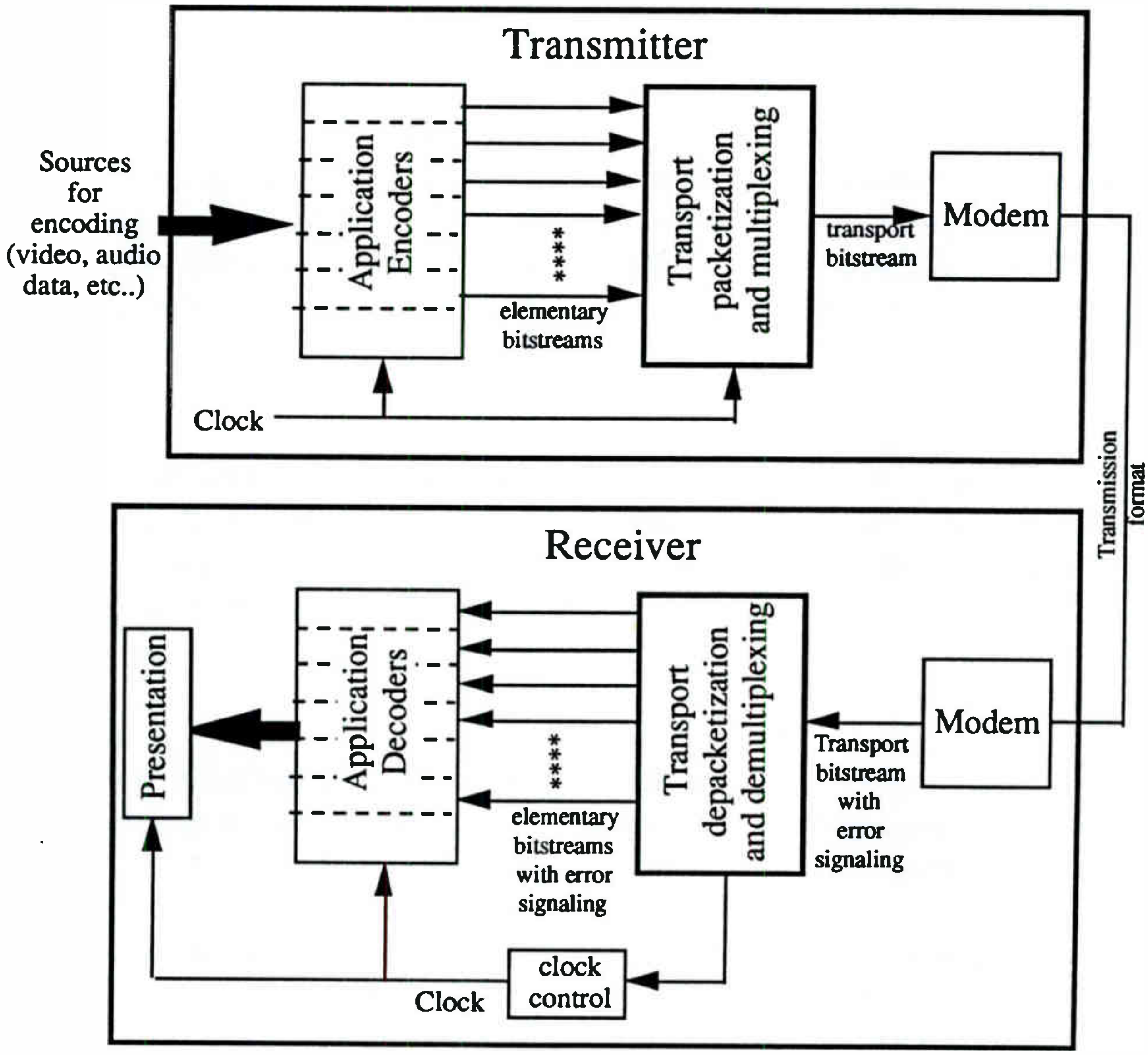

Fig. 1.3. Sample organization of functionality in a transmitter-receiver pair for a single GA program.

As described earlier, the data transport mechanism in the GA system is based on the use of fixed length packets that are identified by headers. Each header identifies a particular application bit stream (also called an *elementary bit stream*) which forms the payload of the packets. Applications supported include video, audio, data, program and system control information, etc.. As indicated earlier, the elementary bit streams for video and audio are themselves be wrapped in a variable length packet structure called the packet elementary stream (PES) before transport processing. The PES layer provides functionality for identification, and synchronization of decoding and presentation of the individual application. The format and functionality of a PES packet is described in a later section.[2]

Moving one level up in the description of the general organization of the GA bit streams, elementary bit streams sharing a common time base are multiplexed, along with a control data stream, into *programs*. These programs and an overall system control data stream are then asynchronously multiplexed to form a multiplexed *system*. The organization is described in detail in a later section. Note that programs in the GA system are analogous to today's NTSC broadcast channels.

At this level, the GA transport is also quite flexible in two aspects:

1. It permits you to define programs as any combination of elementary bit streams, for example the same elementary bit stream could be present in more than one program (e.g., two bit streams with the same audio), a program could be formed by combining a basic elementary bit stream and a supplementary elementary bit stream (i.e., bit streams for scalable decoders), programs could be tailored for specific needs (e.g., regional selection of language for broadcast of secondary audio), etc....

2. Flexibility at the systems layer allows different programs to be multiplexed into the system as desired, and allows the system to be reconfigured easily when required. The procedure for extraction of programs from within a system is also simple and well defined.

The GA format provides other features that are useful for both normal decoder operation and for the special features required in broadcast and cable applications. These include

1. Decoder synchronization
2. Conditional access
3. Local program insertion, etc....

The elements of these features that are relevant to the standard definition process will be discussed in detail.

The GA bit stream definition directly addresses issues related the storage and playback of programs. Although, this is not directly related to the ATV transmission problem, it is a fundamental requirement for creating programs in advance, storing them and playing them back at the desired time. The programs are stored in the same format in which they are transmitted, i.e., as

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[2]Note that the PES layer is not required for all applications. Its use is mandated for both the video and audio in the GA system.

transport bit streams. The GA bit stream format also has the hooks in it to support the design of consumer digital products based on recording and playback of these bit streams, including the use of the "trick modes" that one is familiar with for current analog VCRs. It should be noted that the issues related to storage and play back of digitally compressed video bit streams are quite different from those that need to be considered for analog systems such as NTSC.

1.4. General bit stream interoperability issues

The question has been raised frequently about the bit stream level interoperability of the GA system. There are two sides to this issue. One is whether the GA transport bit stream can be carried on other communication systems, and the other is the ability of the GA system to carry bit streams generated from other communication systems.

The first aspect of transmitting GA bit streams in different communication systems has been addressed to some extent (e.g., for ATM interoperability) in the design of the protocol, and is described in more detail in later sections. In short, there is nothing that prevents the transmission of a GA bit stream as the payload on a different transmission system. It may be simpler to achieve this functionality in certain systems, e.g., CATV, DBS, ATM, etc.., than in others, e.g., computer networks based on protocols such as FDDI, IEEE 802.6, etc.. Since ATM is expected to form the basis of future broadband communications, it is believed that the issue of bit stream interoperability has been addressed for one of the more important transmission scenarios of the future. This is discussed in more detail in Chapter 9.

The other aspect is of transmitting other, non-GA, bit streams within the GA system. This makes more sense for bit streams linked to TV broadcast applications, e.g., CATV, DBS, etc.., but is also possible for other "private" bit streams. This function is achieved by transmitting these other bit streams as the payload of identifiable transport packets. The only requirement is to have the general nature of these bit streams recognized within the GA system context. Note that there is also a certain minimum system level processing defined by the GA that needs to be implemented to extract all (even private) bit streams. The details are made clearer in the sections that follow. It is also important to remember that the GA system is essentially a broadcast system and hence any private transmissions that may be based on a two way communications protocol will not be directly supported, unless this functionality is provided external to the GA system definition.

2. The packetization approach and functionality

The GA transport bit stream consists of fixed length packets with a fixed and a variable component to the header field as illustrated in Fig. 2.1.

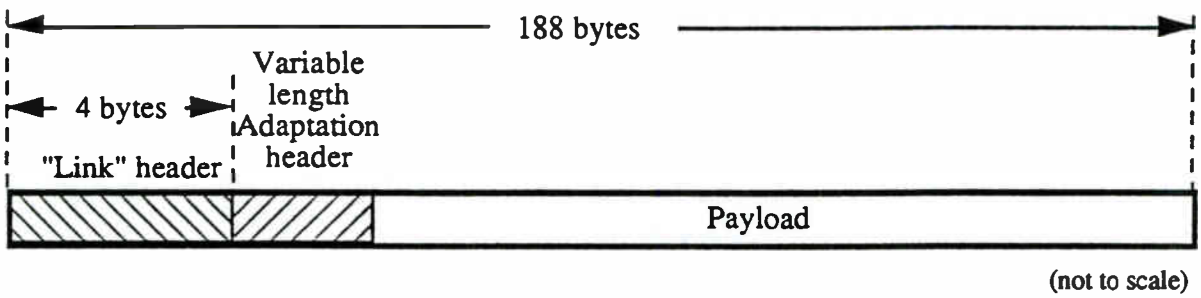

Fig. 2.1. GA Transport packet format

Each packet consists of 188 bytes. The choice of this packet size is motivated by a few factors. The packets need to be large enough so that the overhead due to the transport headers do not become a significant portion of the total data carried. They should not be too large that the probability of packet error becomes significant under standard operating conditions (due to inefficient error correction). It is also desirable to have packet lengths in tune with the block sizes of typical, block oriented, error correction approaches, so that packets may be synchronized to error correction blocks, and the physical layer of the system can aid the packet level synchronization process in the decoder. Another motive for the particular packet length selection is interoperability with the ATM format. The general philosophy is to transmit a single GA transport packet in four ATM cells. There are, in general, several approaches to achieve this functionality. Chapter 9 includes a discussion of some example approaches. If this interface is to be standardized, the issue will be settled outside the scope of the GA/ACATS process.

The contents of each packet and the nature of this data are identified by the packet headers. The packet header structure is layered and may be described as a combination of a fixed length "link" layer and a variable length adaptation layer. Each layer serves a different functionality similar to the link and transport layer functions in the OSI layers of a communications system. This link and adaptation level functionality is directly used for the terrestrial link on which the GA bit stream is transmitted. However these headers could also be completely ignored in a different system (e.g. ATM), in which the GA bit stream may just remain the payload to be carried. In this environment, the GA bit stream headers would serve more as an identifier for the contents of a data stream rather than as a means for implementing a protocol layer in the overall transmission system.

2.1. The "link" layer

The link layer is implemented using a four byte header field. The format of the header field is described in greater detail in a later section. Some of the important functions that are enabled by the header elements are described here.

2.1.1. Packet synchronization

Packet synchronization is enabled by the sync_byte, which is the first byte in a packet. The sync_byte has the same fixed, pre-assigned, value for all GA bit streams. In some implementations of decoders the packet synchronization function is done at the physical layer of the communication link (which precedes the packet demultiplexing stage), in which case this sync_byte field may be used for verification of packet synchronization function. In other decoder implementations this byte may be used as the primary source of information for establishing packet synchronization. The standard does not specify the details of the approach to be used to implement this function in a decoder but only provides the hooks in the bit stream to facilitate the function.

2.1.2. Packet Identification

As discussed earlier, an important element in the link header is a 13 bit field called the PID or Packet ID. This provides the mechanism for multiplexing and demultiplexing bit streams, by enabling identification of packets belonging to a particular elementary or control bit stream. Since the location of the PID field in the header is always fixed, extraction of the packets corresponding to a particular elementary bit stream is very simply achieved once packet synchronization is established by filtering packets based on PIDs. The fixed packet length makes for simple filter and demultiplexing implementations suitable for high speed transmission systems.

2.1.3. Error handling

Error detection is enabled at the packet layer in the decoder through the use of the continuity_counter field. At the transmitter end, the value in this field cycles from 0 through 15 for all packets with the same PID that carry a data payload (as will be seen later, the GA transport allows you to define packets that have no data payload). At the receiver end, under normal conditions, the reception of packets in a PID stream with a discontinuity in the continuity_counter value indicates that data has been lost in transmission. The transport processor at the decoder then signals the decoder for the particular elementary stream about the loss of data. This signaling approach is not included in the standard.

Because certain information (such as headers, time stamps, and program maps) is very important to the smooth and continuous operation of a system, the GA transport system has a means of increasing the robustness of this information to channel errors by providing a mechanism for the encoder to duplicate packets. Those packets that contain important information will be duplicated at the encoder. At the decoder, the duplicate packets are either used if the original packet was in error or are dropped. Semantics for identifying duplicate packets are described in the description of the continuity_counter.

2.1.4. Conditional Access

The transport format allows for scrambling of data in the packets. Each elementary bit stream in the system can be scrambled independently. The GA standard specifies the descrambling approach to be used but does not specify the descrambling key and how it is obtained at the decoder. The key must be delivered to the decoder within a time interval of its usefulness. There is "private" data capacity at several locations within the GA transport stream where this data might be carried. Two likely locations would be 1) as a separate private stream with it's own PID, or 2) a private field within an adaptation header carried by the PID of the signal being scrambled. The security of the conditional access system is ensured by encrypting the descrambling key when sending it to the receiver, and by updating the key frequently. As mentioned before, the key encryption, transmission, and decryption approaches are not a part of the standard and could differ in different versions of the ATV delivery system. There is no system imposed limit on the number of keys that can be used and the rate at which these may be changed. The only requirement in a receiver to meet the standard is to have an interface from the decryption hardware (e.g., a Smart-card) to the decoder that meets the standardized interface spec. The decryption approach and technology is itself not a part of the standard. For the purposes of testing, to demonstrate feasibility, only the descrambling function will be tested and the encryption keys will probably be made directly available at the decoder. Note that the generalized systems layer definition includes a mechanism for transmitting key information, including the process of identifying bit streams carrying key information and the definition of the tables that enable this function.

Information in the link header of a transport packet describes if the payload in the packet is scrambled and if so, flags the key to be used for descrambling. The header information in a packet is always transmitted in the clear, i.e., unscrambled. The amount of data to be scrambled in a packet is variable depending on the length of the adaptation header. It should be noted that some padding of the adaptation field might be necessary for certain block mode algorithms. Conditional access is discussed in greater detail in a later section.

Note that the general MPEG-2 transport definition provides the mechanism to scramble at two levels, within the PES packet structure and at the transport layer. Scrambling at the PES packet layer is primarily useful in the program stream (which is not supported in the GA system), where there is no protocol layer similar to the transport to enable this function. In the GA system scrambling will be implemented only at the transport layer.

2.2. The Adaptation layer

The adaptation header in the GA packet is a variable length field. Its presence is flagged in the link level section of the header. The functionality of these headers is basically related to the decoding of the elementary bit stream that is extracted using the link level functions. Some of the functions of this layer that are important to the functioning of the GA system are described here.

2.2.1. Synchronization and timing

Synchronization of the decoding and presentation process for the applications running at a receiver is a particularly important aspect of real time digital data delivery systems such as the GA system. Since received data is expected to be processed at a particular rate (to match the rate at which it is generated and transmitted), loss of synchronization leads to either buffer overflow or underflow at the decoder, and as a consequence, loss of presentation/display synchronization. The problems in dealing with this issue for a digital compressed bit stream are different from those for analog NTSC. In NTSC, information is transmitted for the pictures in a synchronous manner, so that one can derive a clock directly from the picture synch. In a digital compressed system the amount of data generated for each picture is variable (based on the picture coding approach and complexity), and timing cannot be derived directly from the start of picture data. Indeed, there is really no natural concept of synch pulses (that one is familiar with in NTSC) in a digital bit stream.

The solution to this issue in the GA system is to transmit timing information in the adaptation headers of selected packets, to serve as a reference for timing comparison at the decoder. This is done by transmitting a sample of a 27 MHz clock in the program_clock_reference (PCR) field, which indicates the expected time at the completion of the reading of that field from the bit stream at the transport decoder. The phase of the local clock running at the decoder is compared to the PCR value in the bit stream at the instant at which it is obtained, to determine whether the decoding process is synchronized. In general, the PCR from the bit stream does not directly change the phase of the local clock but only serves as an input to adjust the clock rate. Exceptions are during channel change and insertion of local programming. As mentioned earlier, the nominal clock rate in the GA decoder system is 27 MHz. A point to note here is that the standard only specifies the means of transmitting synchronization information to a receiver but does not specify the implementation of the synch recovery process. Note also that the audio and video sample clocks in the decoder system are locked to the system clock derived from the PCR values. This simplifies the receiver implementation in terms of the number of local oscillators required to drive the complete decoding process, and has other advantages such as rapid synch acquisition.

Details of the format for the PCR are given in a later section. Note that in this implementation the encoder and decoder system clocks are set completely independent of the modem clock. This makes for a clean separation of functionality when implementing the two subsystems, and leads to simpler interfaces. This also makes it simpler to interface GA transmitters and receivers at the transport interface to modems which may be used for transmission on other media such as CATV, DBS, computer networks, etc..

2.2.2. Random entry into the compressed bit stream

Random entry into the application bit streams such as video and audio is necessary to support functions such as program tuning and program switching. Random entry into an application is possible only if the coding for the elementary bit stream for the application supports this functionality directly. For example, a GA video bit stream supports random entry through the concept of Intra (or I-) frames that are coded without any prediction, and which can therefore be decoded without any prior information. The beginning of the video sequence header information preceding data for an I-frame could serve as a random entry point into a video elementary bit stream. In general, random entry points should also coincide with the start of PES packets where they are used, e.g., for video and audio. The support for random entry at the transport layer comes from a flag in the adaptation header of the packet that indicates whether the packet contains a random access point for the elementary bit stream. In addition, the data payload of packets that are random access points also start with the data that forms the random access points into the elementary bit stream itself. This approach allows the discarding of packets directly at the transport layer when switching channels and searching for a resynchronization point in the transport bit stream, and also simplifies the search for the random access point in the elementary bit stream once transport level resynchronization is achieved.

A general objective is to have random entry points into the programs as frequently as possible, to enable rapid channel switching.

2.2.3. Local program insertion

This functionality is important for inserting local programming, e.g., commercials, into a bit stream at a broadcast headend. In general, there are only certain fixed points in the elementary bit streams at which program insertion is allowed. The local insertion point has to be a random entry point but not all random entry points are suitable for program insertion. For example, for GA video, in addition to being a random entry point, the VBV_delay (video buffer verifier delay) needs to be at a certain system defined level to permit local program insertion.[3] This is a requirement to control the memory needed at the decoder for buffering data and to prevent buffer overflow or underflow. Local program insertion also always takes place at the transport packet layer, i.e., the data stream splice points are packet aligned. Implementation of the program insertion process by the broadcaster is aided by the use of a splice_countdown field in the adaptation header that indicates ahead of time the number of packets to countdown until the packet after which splicing and local program insertion is possible. The insertion of local programming usually results in a discontinuity in the values of the PCR received at the decoder. Since this change in PCR is completely unexpected (change in PCR values are usually only expected during program change), the decoder clock could be thrown completely out of synchronization. To prevent this from happening, information is transmitted in the adaptation header of the first packet after the splicing point to notify the decoder of the change of PCR values (so that it can change the clock phase directly instead of attempting to modify the clock rate). In addition there are constraints on 1) the length of the bit stream that is to be spliced in, to assure that the buffer occupancies at the decoder both with and without the splice would be consistent, and 2) the initial VBV value assumed when encoding the bit stream to be spliced in, in order to prevent decoder buffer underflow or overflow.

The details of the syntax elements that support splicing and local program insertion are described in the chapter on the transport format. More specifics of the particular implementation for the GA system will be described in a separate section.

In addition to the functions described above, the adaptation header includes capabilities for extension of the header, for supporting new functionality, and also for defining data that is private and whose format and meaning are not defined in the public domain. These elements may be useful in extending the GA transport beyond the current range of expectations for usage.

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[3] The `VBV_delay` information is computed and transmitted as a part of the header data for a picture in the compressed video bit stream. It defines how full the decoder video buffer should be just before the bits for the current picture are extracted from the buffer, if the decoder and encoder processes are synchronized.

3. Higher Level Multiplexing functionality

As described earlier, the overall multiplexing approach can be described as a combination of multiplexing at two different layers. In the first layer one forms program transport streams by multiplexing one or more elementary bit streams at the transport layer, and in the second layer the program transport streams are combined (using asynchronous packet multiplexing) to form the overall system. The functional layer in the system that contains both this program and system level information that is going to be described is called the PSI or Program Specific Information.

3.1. Single Program Transport Multiplex

A GA program transport bit stream[4] is formed by multiplexing individual transport packetized elementary bit streams (with or without PES packetization) sharing a common time-base, and a control bit stream that describes the program. Each elementary bit stream, and the control bit stream (also called the elementary stream map in Fig. 3.1), are identified by their unique PIDs in the link header field. The organization of this multiplex function is illustrated in Fig. 3.1. The control bit stream contains the program_map_table that describes the elementary stream map. The program_map_table includes information about the PIDs of the transport streams that make up the program, the identification of the applications that are being transmitted on these bit streams, the relationship between these bit streams, etc.. The details of the program_map_table syntax and the functionality of each syntax element are given in a later section. The identification of a bit stream carrying a program_map_table is done at the system layers to be described next.

The transport syntax allows a program to be comprised of a large number of elementary bit streams, with no restriction on the types of applications required within a program. A program transport stream does not need to contain compressed video or audio bit streams, or, for example, it could contain multiple audio bit streams for a given video bit stream. The data applications that can be carried are flexible, the only constraint being that there should be an appropriate stream_type ID assignment for recognition of the application corresponding to the bit stream in a GA decoder. The list of application types that will be a supported in the initial GA system are given in the section on services supported by the GA system. Note that the initial selection of applications does not limit the future. (Indeed, it is quite impossible for one to anticipate all possible future applications!)

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[4]The terminology can be confusing. A **program** is analogous to a channel in NTSC. A **program stream** refers to a particular bitstream format, described in the introduction, that is not being used in the GA system. A **program transport stream** is a term used to describe a transport bitstream that has been generated for a program.

Note that many of the link level functions are carried out independently, without program level coordination, for the different elementary bit streams that make up a program. This includes functions such as PID manipulation, bit stream filtering, scrambling and descrambling, definition of random entry packets, etc.. The coordination between the elements of a program is primarily controlled at the presentation (display) stage based on the use of the common time base. This common time base is set up by the fact that all elementary bit streams in a program derive timing information from a single clock, the information for which is transmitted via the PCR on one of the elementary bit streams that constitute the program. The data for timing of presentation is present in the elementary bit streams for individual applications.

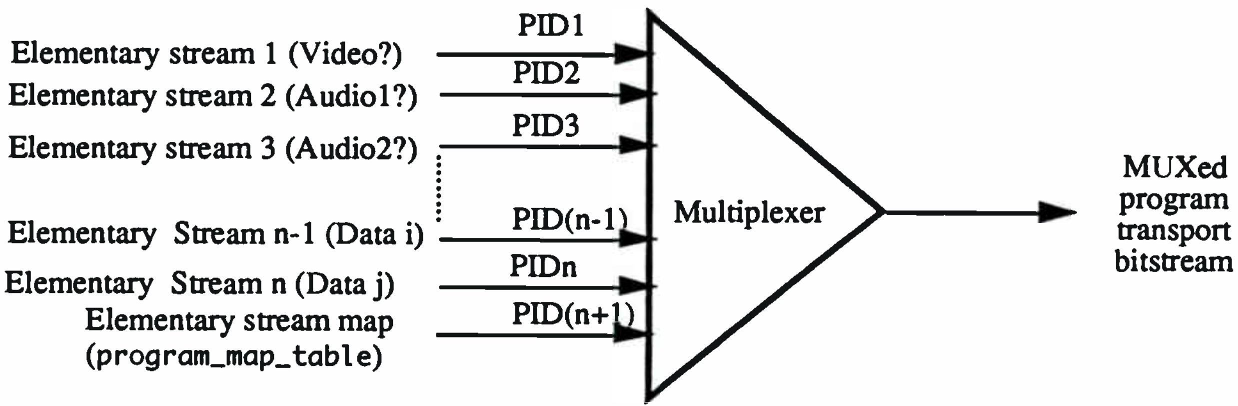

Fig. 3.1. Illustration of the multiplex function to form a program transport stream.

Although there is no restriction on the PID values that can be assigned within a program transport multiplex, in light of the fact that PIDs need to be unique at a system level, a standardized PID assignment approach should be considered for the GA system. A suggestion is to use the LSB bits in the PIDs to identify the stream type.

3.2. System Multiplex

The system multiplex is the process of multiplexing different program transport streams. In addition to the transport bit streams (with the corresponding PIDs) that define the individual programs, a system level control bit stream with PID = 0 is defined. This bit stream carries the program_association_table that maps program identities to their program transport streams. The program identity is represented by a number in the program_association_table. A program corresponds to what is traditionally called a channel, e.g., HBO™, ESPN™, etc.. The map indicates the PID of the bit stream containing the program_map_table for a program. Thus, the process of identifying a program and its contents takes place in two stages: first one uses the program_association_table in the PID = 0 bit stream to identify the PID of the bit stream carrying the program_map_table for the program, in the next stage one obtains the PIDs of the elementary bit streams that make up the program from the appropriate program_map_table. Once this step is completed the filters at a demultiplexer can be set to receive the transport bit streams that correspond to the program of interest.

The system layer of multiplexing is illustrated in Fig. 3.2. Note that during the process of system level multiplexing, there is the possibility of PIDs on different program streams being identical at the input. This poses a problem since PIDs for different bit streams need to be unique. A solution to this problem lies at the multiplexing stage, where some of the PIDs could be modified just before the multiplex operation. The changes have to be recorded in both the program_association_table and the program_map_table. Hardware implementation of the PID reassignment function in real time is helped by the fact that this process is synchronous at the packet clock rate. The other approach, of course, is to make sure up front that the PIDs being used in the programs that make up the system are unique. This is not always possible with stored bit streams.

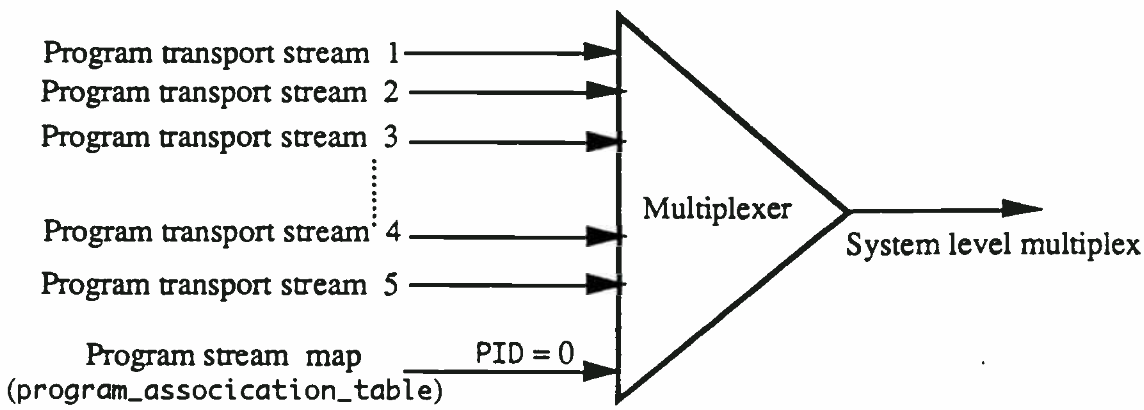

Fig. 3.2. Illustration of the multiplex function to form the system level bit stream.

Note that the architecture of the GA bit stream is scalable. Multiple system level bit streams can be multiplexed together on a higher bandwidth channel by extracting the program_association_tables from each system multiplexed bit stream and reconstructing a new PID = 0 bit stream. Note again that PIDs may have to be reassigned in this case.

Note also that in all descriptions of the higher level multiplexing functionality no mention is made of the functioning of the multiplexer and multiplexing policy that should be used. This function is not a part of the standard and is up to individual designers. Because its basic function is one of filtering, the transport demultiplexer will function on any GA bit stream regardless of the multiplexing algorithm used.

Fig 3.3 illustrates the entire process of extracting elementary bit streams for a program at a receiver. It also serves as one possible implementation approach (although not the most efficient! In practice the same demultiplexer hardware could be used to extract both the program_association_table and the program_map_table control bitstreams.). This also represents the minimum functionality required at the transport layer to extract any application bit stream (including those that may be private).

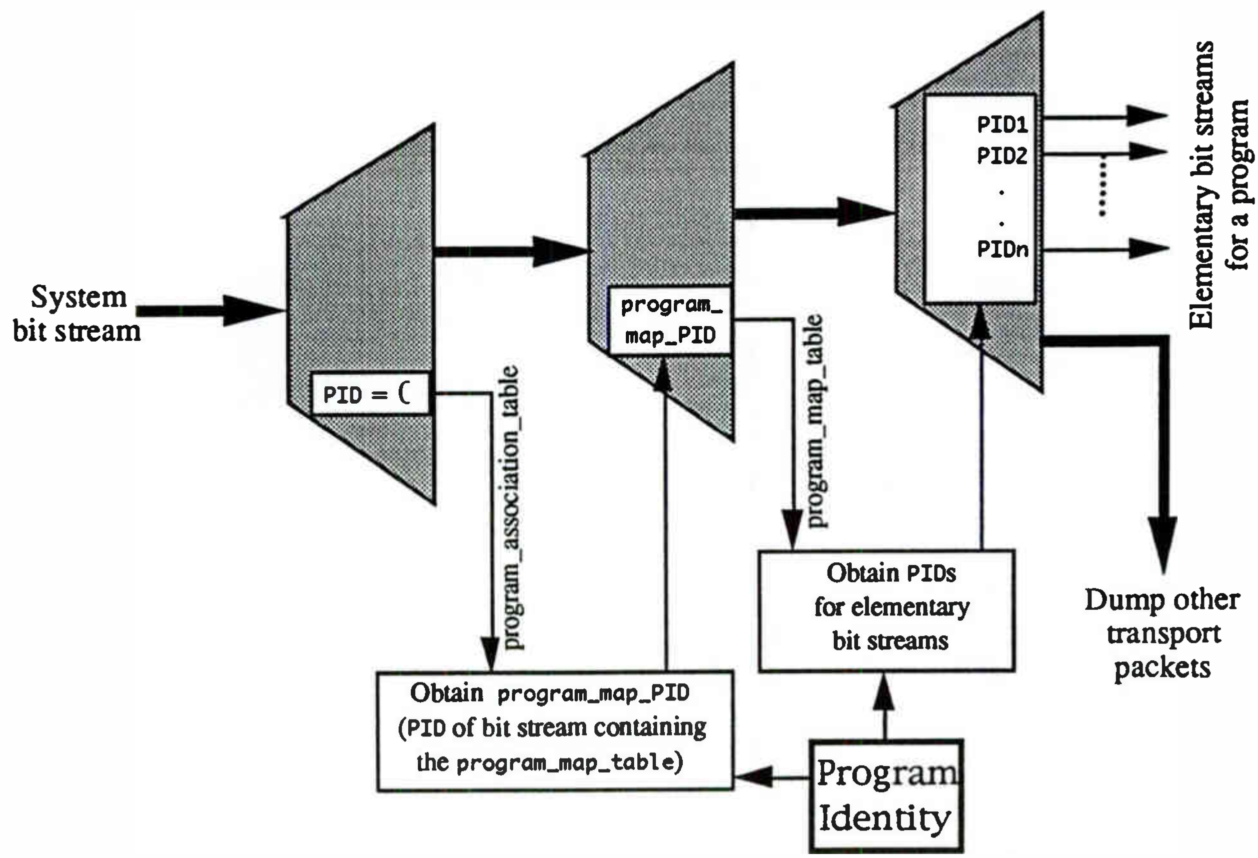

Fig 3.3. Illustration of transport demultiplexing process for a program.

Note that once the packets are obtained for each elementary bit stream in the program, further processing stages of obtaining the random entry points for each component bit stream, decoder system clock synchronization, presentation (or decoding) synchronization, etc.., need to take place before the receiver decoding process reaches normal operating conditions for receiving a program.

It is important to clarify here that the layered approach to defining the multiplexing function does not necessarily imply that program and system multiplexing should always be implemented in separate stages. A hardware implementation that includes both the program and system level multiplexing within a single multiplexer stage is allowed, as long as the multiplexed output bit stream has the correct properties as defined in this document.

4. Features and Services Supported by the GA ATV System

As we have demonstrated, the GA Transport architecture has been designed to be maximally flexible and is capable of supporting a vast number of services through its system multiplex. The transport, however, is not the only means by which information can be delivered in the ATV system. The GA video syntax also provides means for delivery of predefined information.

Although this is a global issue for the GA ATV system, it is appropriate to review the services supported in the context of the transport specification, since this sub-system has ultimate responsibility for the multiplexing of supported services. In this chapter, we will review a number of functions identified by ATSC document T3/186 and the SMPTE headers/descriptors group and describe how they are carried within the ATV system. In some cases, these functions are a matter to be communicated between the studio that sources the program and the final encoder that transmits the bit stream. While the final transmitted bit stream does not necessarily need to carry this information to the receiver, it is important to see the capabilities of the ATV system to carry this information in a network distribution capacity.

4.1. Features Supported within the Grand Alliance Video Syntax

Pan & Scan:
Pan & scan information is supported within the GA video syntax. This information is transmitted as an extension within the picture layer syntax. The pan & scan extension allows decoders to define a rectangular region which may be panned around the entire coded image. This facility could be used to identify a 4:3 aspect ratio window within a 16:9 coded image.

Field/Frame Rate and Film Pull-down:
The GA video syntax provides means for transmitting the frame rate of the coded bit stream. This allows the encoder to maximize coding efficiency by not transmitting redundant fields, and signals the decoder the proper order for displaying the decoded pictures. The GA syntax supports frame rates of 23.976, 24, 29.97, 30, 59.94 and 60 Hz as well as an extension for future capabilities. The frame rate syntax is found within the video sequence layer.

Picture Structure Information:
This information details the sampling structure used in the coded image, including samples per line, lines per frame, and scanning format (interlace or progressive). This information is supported by the video syntax and is found within the sequence layer.

Picture Aspect ratio:
The video syntax provides a field for sample aspect ratio within the sequence layer. This information combined with the picture structure fields, allows the picture aspect ratio to be determined.

Color field Identification:
This information is supported by the GA video syntax and helps the decoder re-encode the image to an NTSC compatible output with reduced NTSC artifacts.

Colorimetry
Information on the colorimetry characteristics of the video to be encoded are supported by syntax in the video sequence layer. This includes description of the color primaries, transfer characteristics and the color matrix coefficients.

4.2. Features Supported as Multiplexed Services within the Grand Alliance Transport System

As mentioned earlier, the GA transport scheme provides great flexibility for multiplexing a variety of services in addition to video and audio elementary streams. Several possible services that could be supported under this transport definition are summarized below.

Audio Compression Types and Language Identification:
The transport layer syntax defines a program map which permits identification of individual audio services by their compression algorithm as well as signaling the presence of a secondary language channel that can be selected by the viewer.

Program Information
This service could be provided to the decoder as an ancillary data service with its own PID. This could take the form of a TV guide that is personalized by the service provider. The information would require only a low refresh rate that would not consume a significant amount of the channel bandwidth.

Other Program-related Information

There is a large body of program-related information that could be identified for use at the decoder. MPEG-2 systems syntax, on which the GA system is based, currently supports copyright notification, but has not defined a separate capability for program classification data. This and other program-related information could be addressed as private data.

4.3. Support of Closed Captioning

The Grand Alliance has been participating in dialogue with the EIA committee on ATV Closed Caption Standards to ensure proper support of this important feature. The standards group reviewed a number of important requirements for carriage of the closed caption service in a digitally compressed ATV system. In response to those stated needs, the Grand Alliance has indicated that closed captioning would be carried as user data within the video picture layer.

Support of closed caption in this manner, allows a fixed amount of channel capacity to be dedicated to the service, while maintaining absolute synchronization with each ATV frame. Carriage of this data as a separate service would require a separate synchronization mechanism as exists within the audio decoder. This synchronization was an essential requirement stated by the standards group. Additionally, this mechanism allows for relatively easy editing of the closed caption data downstream. It is expected that the closed caption data will require a dedicated allocation of 9600 bits per second.

Our work with this standards body will continue as we progress to final specifications, and our work will be updated to the Advisory Committee.

4.4. Features not Anticipated to be Transmitted by the Grand Alliance System to the Consumer Receiver

Telecine Source Identification:

Source identification information that could inform the encoder whether the video was originally film or video could assist the detection of redundant fields and thereby improve coding efficiency. There is no syntax provided in the video bit stream to carry this external information, however it could be multiplexed in at the systems level as an ancillary data service.

Image Processing History:

It is projected that future encoders could take advantage of knowledge of any algorithms that have been applied to the image sequence. This is knowledge that would need to be passed from the studio to the encoder, and would not be sent to the decoder. It could be carried in an ancillary data stream in a wider bandwidth network distribution system.

Scene Change:
Automatic scene change detection algorithms are used in some encoders to improve coding efficiency. There has not been an interface specified for this information to be transmitted to the encoder, but it would aid in the coding algorithm. Such scene change information, if supported by a production facility, could provide useful information to the video encoder at both the compression and transport levels and we look forward to working with standards bodies to make this effective.

Conditional Access Identification:
Implementation of Conditional Access systems is supported by the transport syntax, with bits defined in the packet header. Delivering information about the conditional access information, including key information is an issue that would be addressed as private data in the GA syntax. This issue is covered further in a separate chapter on Conditional Access.

Universal Identifier:
Definition of Universal Identifier information has not yet been finalized by SMPTE. The issue of the registration descriptor is addressed more completely in chapter 9.

5. The Transport format and protocol

This chapter defines the syntax elements for the transport layer of the GA bit stream. All syntax elements need to be recognized at some level in a GA receiver. Most trigger a response in the transport decoder. A few are present for interoperability with MPEG-2.

5.1. Link level headers

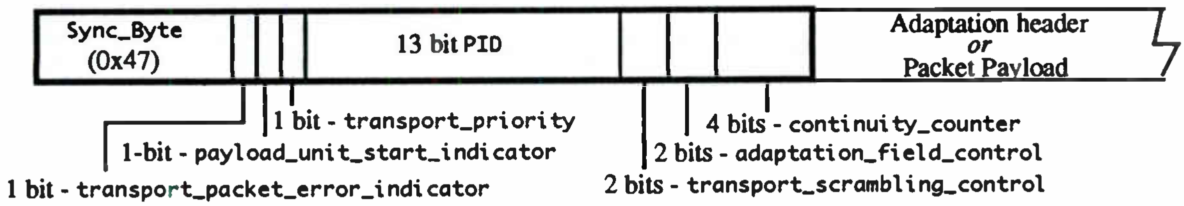

Fig. 5.1. Link header format for the GA system transport packet.

Fig. 5.1. shows the link layer headers with the functionality assigned to each bit. Some of these have been discussed earlier. The table below spells out these functions in detail. These general functions may not all be used on the broadcast channel, but are useful for transmitting the same bit stream over other links, including cable links, computer networks, etc.. In short, these provide interoperability features.

field	General function	GA usage
Sync_byte Value - 0x47	Packet synchronization.	As defined. See section 2.1.1.
transport_packet_error_ indicator	Indicates if packet is erroneous 0 - no error 1 - erroneous packet	Can be used for error signaling from modem to transport demultiplexer. If this bit is set the payload is not supposed to be used.
payload_unit_start_ indicator	Indicates if a PES packet header or the start of a table containing program specific information (PSI) is present in the payload in this packet. The PES packet header always begins the payload of the packet. The starting byte of the PSI table in the packet is indicated using a pointer field to be descibed later. 0 - no PES header or start of PSI table present 1 - PES header present	As defined for resynch into the transport stream.

transport_priority	Priority indicator at input to transmission channels/networks which support prioritization. 0 - lower priority 1 - higher priority	The GA transmission system is currently not expected to support prioritization. If it does, this bit will be set during transport packetization process, to route packets to the transmission path with the appropriate priority.
PID	Packet Identifier for mux/demux.	As defined. See section 2.1.2.
transport_scrambling_ control	Indicates the descrambling key to use for the packet 00 - not scrambled others - user defined.	00 - not scrambled 10 - "even" key 11 - "odd" key 01 - reserved See section 2.1.4.
adaptation_field_control	Indicates if an adaptation field follows 00 - reserved 01 - no adaptation field, payload only 10 - adaptation field only, no payload 11 - adaptation field followed by payload	As defined.
continuity_counter	Increments by one for each packet with a given PID and transport priority. Used at the decoder to detect lost packets. Not incremented for packets with adaptation field of 00 or 10. If two consecutive transport packets of the same PID have the same continuity_counter value and the adaptation_field_control equals '01' or '11', the two transport packets shall be considered duplicate.	As defined. See section 2.1.3.

5.2. Adaptation level headers

The presence of the adaptation header field is signaled in the adaptation_field_control of the link layer as described before. The adaptation header itself consists of information useful for higher level decoding functions. The header format is based on the use of flags to indicate the presence of the particular extensions to the field.

The header starts with a fixed length component that is always present (if an adaptation header is transmitted). The format is shown in Fig. 5.2.

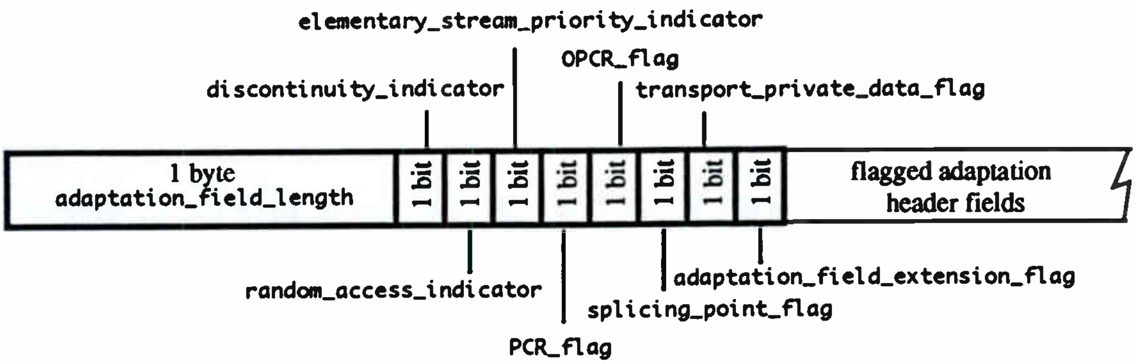

Fig. 5.2. Format for the fixed length component of the adaptation header.

The adaptation_field_length specifies the number of bytes that follow it in the adaptation header. The adaptation header could include stuffing bytes after the last adaptation header component field. (Stuffing bytes have a value of 0xff and are not interpreted at the decoder.) In this case the adaptation_field_length also reflects the stuffing bytes. The value in the adaptation_field_length field can also be used by the decoder to skip over the entire adaptation header, and to directly advance to the data payload in the packet if desired.

The presence of additional adaptation header fields is indicated by the state of the last five single bit flags shown in Fig. 5.2. (with a value of '1' indicating that a particular field is present). The three flags at the beginning do not result in extensions to the adaptation header and are described in the table below.

field	General function	GA usage
discontinuity_indicator	Indicates if there is a discontinuity in the PCR values that will be received from this packet onwards. This happens when bit streams are spliced. This flag should be used at the receiver to change the phase of the local clock.	As defined for local program insertion. See section 2.2.3.
random_access_indicator	Indicates that the packet contains data that can serve as random access point into the bit stream. As an example, these can correspond to the start of sequence header information in the GA video bit stream.	As defined. See section 2.2.2.
elementary_stream_priority_indicator	Logical indication of priority of the data being transmitted in the packet.	Not used. If set it is ignored at a GA decoder.

As mentioned earlier, the other components of the adaptation header appear based on the state of the flags shown in Fig. 5. The order in which these components appear in the bit stream is the same as the order of the flags. Based on the type of adaptation header information being conveyed, the data in these fields may be either fixed length or variable length. These fields are described in detail next.

5.2.1. The PCR and OPCR fields

The use of the PCR has been described in detail in section 2.2.1. This section deals mainly with the format for transmission.

Overall Format

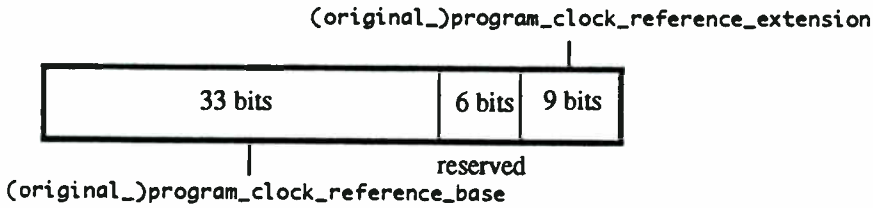

Fig. 5.3. The (O)PCR header format.

Functionality

field	General function	GA usage
PCR	Indicates intended time of arrival of last byte of the program_clock_reference_extension at target decoder. Used for synchronization of the system decoding process. This field can be modified during the transmission process.	As defined. The PCR will be transmitted at least once every 100 milliseconds.
OPCR	Indicates intended time of arrival of last byte of the original_program_clock_reference_extension at target decoder for a single program. This field is not modified during transmission.	May be used for recording and playback of single programs. Not used in the GA receiver in the decoding process.

Functional details of the (O)PCR format

The total PCR value is based on the state of a 27 MHz clock. The 9 bit extension field cycles from 0 to 299 at 27 MHz, at which point the value in the 33 bit field is incremented by one. (This results in the 33 bit field being compatible with the 33 bit field that are used for the 90 KHz clock of MPEG-1. Backward compatibility was a concern in the MPEG-2 system design.) The cycle time of the PCR value is approximately 26 hours.

5.2.2. The transport_private_data and adaptation_field_extension fields

Format

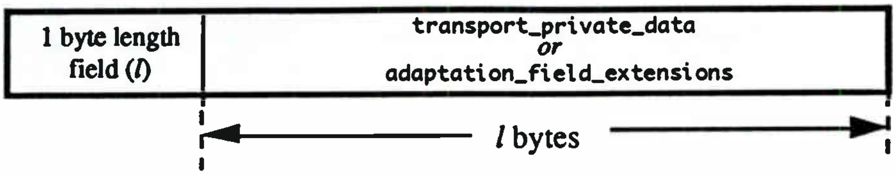

Fig. 5.4. The transport_private_data and adaptation_field_extension header format.

Functionality

field	General function	GA usage
transport_private_data	For private data not recognizable by the general MPEG decoders. Meant for short bursts of control information.	To be decided
adaptation_header_extensions	For future extensions of the adaptation header which may have not been thought of yet.	Not used currently.

5.2.3. The splice_countdown field

This field is useful for local program insertion as described in section 2.2.3.

Format: This is a one byte field that is present if the splicing_point_flag is set.

Functionality

General Function	GA Usage
Indicates number of packets present in the bit stream, with the same PID as current packet, until a splicing point packet. The splicing point packet is defined as the packet containing a point in the elementary bit stream from which point onwards data can be removed and replaced by another bit stream, so that the resulting transport bit stream is valid according to MPEG-2 rules. Transmitted as a 2s-complement value.	Used for supporting insertion of local programming.

5.3. PSIs and the pointer_field

As mentioned in chapter 3, the program_association_table and the program_map_tables that describe the organization of a multiplexed GA bit stream are a part of the PSI layer of the GA system. PSI tables, in general, are transmitted in the appropriate bit stream sequentially, without any gap between the tables. This implies that tables do not necessarily start at the beginning of a transport packet. This also implies that in order to decode specific tables, there needs to be some indication of where these begin in the bit stream. This functionality is achieved with the pointer_field. The pointer_field, if present, is the byte of the payload of a packet (after the link and adaptation headers). The pointer_field is present in the packet if a PSI table begins in the packet, an event which is signaled at the link level, by setting the payload_unit_start_indicator (described in section 5.1) to '1'.

The pointer_field indicates the number of bytes that follow it before the start of a PSI table. As an example a pointer_field value of 0x00 indicates that a new PSI table begins immediately following it.

5.4. The program_association_table

As discussed in section 3.2, the program_association_table is transmitted as the payload of the bit stream with PID = 0 and describes how program numbers associated with programs, (e.g., HBO™, ESPN™, etc.) map on to bit streams containing the program_map_tables for these programs. This section discusses the syntax of the table in some depth. The program_association_table may be transmitted as multiple program_association_segments with each segment having a maximum length of 1024 bytes. The transport decoder can extract individual table segments from the bit stream in whatever order it desires. As shown in Fig. 5.5, each table segment has a fixed length 10 byte header component for table segment identification, a variable length component that depends on the number of entries contained, and a 4 byte CRC-32 field.

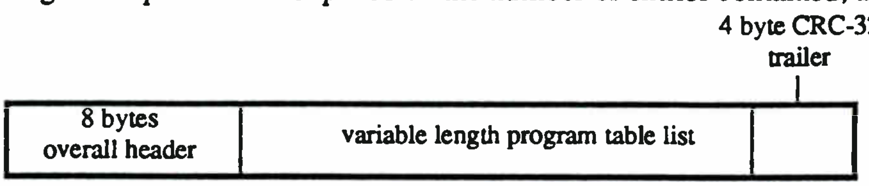

Fig. 5.5. High level overall picture of the program_association_segment.

i. The fixed length overall header component is shown below in Fig. 5.6 and is described in the table below.

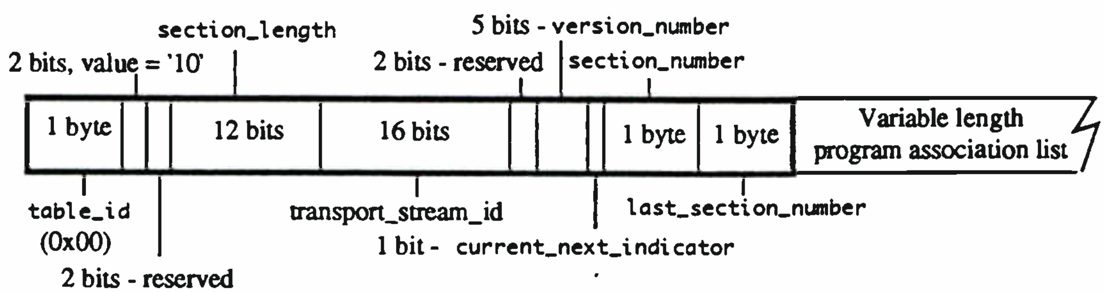

Fig. 5.6. Fixed length header component of the program_association_table.

field	General function	GA usage
table_id	Indicates the nature of the table. 0x00 indicates a program_association_table	As defined.

section_length	Length of the section of the program_association_table. This length includes all bytes following this field, up to and including the CRC. The two most significant bits of this field are set to 0, i.e., maximum value is 1024. This field allows the transport decoder to skip sections when reading from the bit stream if desired.	As defined.
transport_stream_id	Identication of a particular multiplex from several in the network.	*Should correspond to a channel number for the terrestrial application.*
version_number	Incremented each time there is a change in the program_association_ table being transmitted.	As defined
current_next_indicator	Value of 1 indicates that the map is currently valid. Value of 0 indicates that the map is not currently being used and will be used next.	As defined
section_number	Identifies the particular section being transmitted.	As defined
last_section_number	Section_number for the last section in the program_association_ table. Needed to confirm when an entireprogram_association_table has been received at the decoder.	As defined

The value of the reserved bits is undefined and the GA system should not interpret these. On the other hand the 2 bit '10' value following the table_id needs to be received correctly.

ii. The variable length component of the table consists of program_count number of fixed length entries corresponding to each program, and stuffing_bytes (to make up the program_association_segment_length). The format for each fixed length entry is shown in Fig. 5.7.

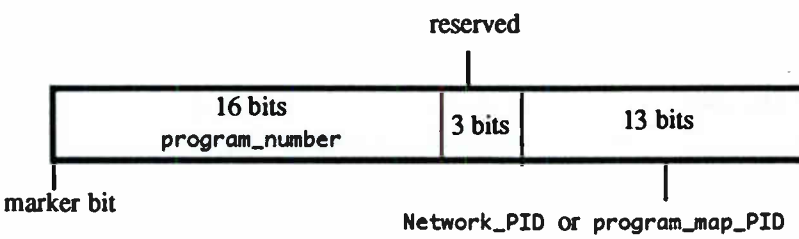

Fig. 5.7. Format of each entry in the program_association_table.

The program identity '0' is reserved for the network_PID, i.e., the PID of the bit stream carrying information about the configuration of the overall system. The nature of this bit stream is yet to be defined for the GA system. The format for this bit stream is completely open since it is meant to be a private bit stream. For all other program identities, the program_map_PID is the PID of the bit stream containing the program_map_table for the particular program.

iii. The program_association_table ends with a four byte CRC field that contains results of a CRC done over the entire program map segment, starting at the segment_start_code_prefix. The CRC is based on the polynomial $x^{32}+x^{26}+x^{23}+x^{22}+x^{16}+x^{12}+x^{11}+x^{10}+x^{8}+x^{7}+x^{5}+x^{4}+x^{2}+x+1$.

5.5. The program_map_table

As discussed in the previous section, the program_map_table is transmitted as the payload of the bit stream with PID = program_map_PID (as indicated in the program_association_table). The program_map_table carries information about the applications that make up programs. Basically each program_map_table is transmitted as a single TS_program_map_section. The format for a TS_program_map_section can be described as a combination of an overall header field, fields that describe each program within the table and a trailer CRC field, as shown in Fig. 5.8. The CRC used is the same as for the program_association_table. In general, each program_map_PID may contain more than one TS_program_map_section, each describing a different program.

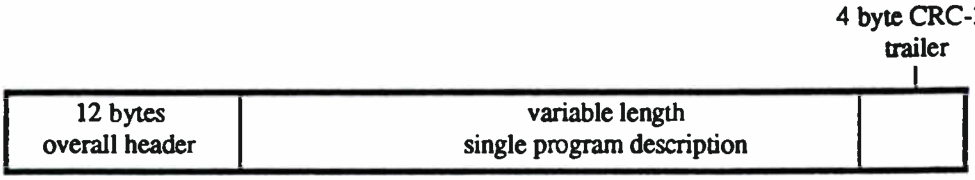

Fig. 5.8. High level overall view of the TS_program_map_section.

5.5.1. The overall TS_program_map_segment header format

The header format for a TS_program_map_section is shown in Fig. 5.9. The format of the first 8 bytes is the same and has similar functionality as that for the program_association_table. The similarity in format is intended to facilitate simple software decoding of the headers. Note that the table_id for TS_program_map_section is different from that for the program_association_table.

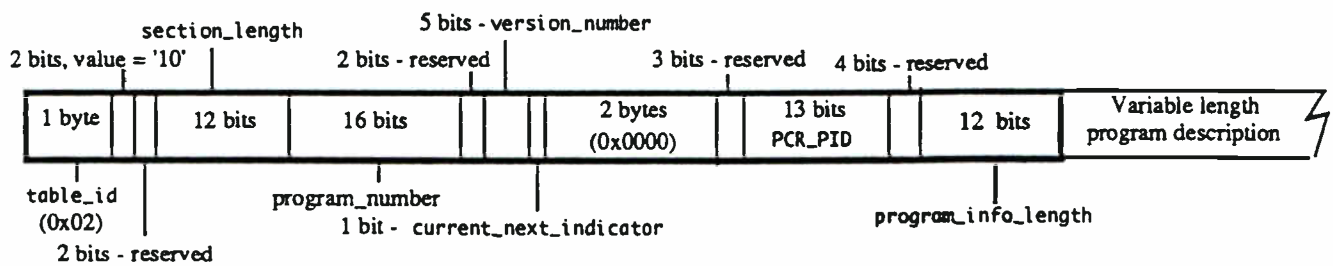

Fig. 5.9. Fixed length header for the TS_program_map_section.

The two bytes that were used to identify the transport_stream_id in the association table are now used to identify the program_number of the program whose description follows. The section identification functions are not required for this table since the description of each program is defined to have to fit into one section. Hence these fields are set to '0' as shown in the figure. The other common header fields between the program_association_table and the TS_program_map_section have functionality as desecribed in the table following Fig. 5.8. The additional header fields in a TS_program_map_section are the 13 bit PCR_PID field that identifies the PID of the particular packetized elementary bit stream in the program that contains the PCR values for the program, and the program_info_length field, that indicates the number of bytes of program_descriptors that follow this header field.

The overall program description that follows the header described above consists of the optional, variable length, program_descriptor field (whose length was indicated by the program_info_length field shown in Fig. 5.9), followed by a descriptions of each of the individual elementary bit streams that make up the program, i.e., there are one or more elementary bit stream descriptions for each TS_program_map_section.

5.5.2 An elementary stream description

Each elementary stream description for has a 5 byte fixed length component and a variable length elementary_stream_descriptor component as shown in Fig. 5.10.

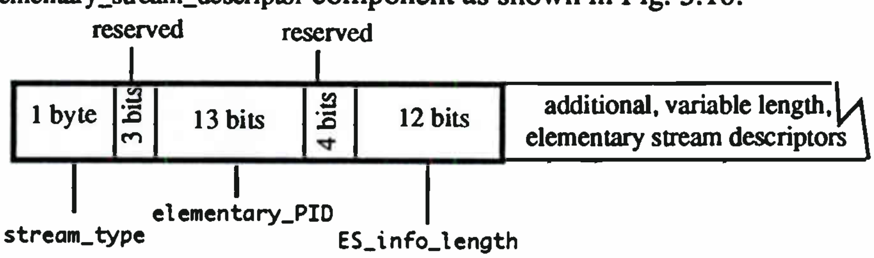

Fig. 5.10. The fixed length component of the elementary stream description.

The functionality of the different fields is as follows.

field	General function	GA usage
stream_type	Indicates the application being considered in this elementary stream.	??.
elementary_pid	Indicates the PID of the transport bit stream containing the elementary bit stream.	As defined
ES_info_length	Indicates the length of a variable length elementary_stream_descriptor field that immediately follows.	As defined.

As before reserved bits are ignored.

5.6. Descriptors

Descriptors are transmitted in the program_descriptor and the elementary_stream_descriptor fields to describe certain charaacterisitics of the program or the elementary bit stream. In general, each program_descriptor and the elementary_stream_descriptor can consist of number of individual descriptor field elements transmitted sequentially.

Two factors need to be considered in order to use descriptors. In the first place, there has to be an mechanism for indicating the presence of the descriptors. In the PSI tables that have been described, this functionality is achieved by the length field that preceeds the descriptor, with a value of zero indicating that no descriptor are present. A second function is the identification of the descriptor itself. This is achieved within the descriptor header itself, which consists of a one byte descriptor_tag field followed by a one byte descriptor_length field that specificies the number of bytes in the descriptor following the descriptor_length field. The set of valid descriptor_tags in the GA systems is the same as that defined for MPEG-2. The table of tags and the format for each descriptor are not described in this document.

6. The PES packet format

As described before, some elementary bit streams, including the GA video and compressed audio, will go through a PES layer packetization prior to the GA transport layer. The PES header carries various rate, timing, and data descriptive information, as set by the encoder. The PES packetization interval is application dependent. The resulting PES packets are of variable length with a maximum size of 2^{16} bytes, when the PES packet length field is set to its maximum value. This value is set to zero for the GA video stream, implying that the packet size is unconstrained and that the header information cannot be used to skip over the particular PES packet. This value of Note also that the PES packet format has been defined to also be of use as an input bit stream for Digital Storage Media (DSM) applications. Although the DSM format will not be used for GA broadcast application, some of the PES header fields related to the DSM functions are also described in this chapter. Note that the ability to handle input bit streams in the DSM format is not essential for a GA receiver, but may be useful for VCR applications.

The constraints on the length of the PES packets for GA video have to be settled upon. Among the decisions to be reached are whether PES packets should only start on GOP boundaries, whether all GOP boundaries should start a new PES packet, whether PES packets should be defined for each access unit (corresponding to a picture), etc.. Note that the format for carrying the PES packet within the GA transport is a subset of the general definition in MPEG. This choice was made to simplify the implementation of the GA receiver. Essentially, in the GA transport system, all data for a PES packet, including the header, are transmitted contiguously as the payload of transport packets. New PES packet data always starts a new transport packet, and PES packets that end in the middle of a transport packet are followed by stuffing bytes for the remaining length of the transport packet.

A PES packet consists of a PES_packet_start_code, PES header flags, PES packet header fields, and a payload (or data block), as shown in Fig. 6.1. It is created by the application encoder. The packet payload is a stream of contiguous bytes of a single elementary stream. For video and audio packets, the payload is a sequence of access units from the encoder. The access units correspond to the video pictures and audio frames.

3 bytes	1 byte	2 bytes	2 bits	14 bits	1 byte	Variable Length	
Packet Start Code Prefix	Stream ID	PES Packet Length	10	PES Header Flags	PES Header Length	PES Header Fields	PES Packet Data Block

Figure 6.1. Structural Overview of a PES Packet

Each elementary stream is identified by a unique stream_id. The PES packets from each encoder carry the corresponding stream_id. PES packets carrying various types of elementary streams can be multiplexed to form a program or transport stream in accordance with Part 1 of the MPEG-2 standard. This chapter deals with the organization of PES packets for MPEG-2 compliant streams only. Specifically, the stream_id field can take on a number of values, indicating the type of data in the payload (See Table for valid values). This section does not define the PES packet structure for 'private data', i.e. stream_id must be 'Reserved', 'Padding', or 'MPEG Audio' or 'MPEG Video'. For 'private data', the PES packet structure is user defined.

The preliminary fields, the packet_start_code_prefix, stream_id, and PES_packet_length, are described in the table below.

Field	Description	GA Usage
packet_start_code_prefix	Indicates the start of a new packet. Together with the stream_id, it forms the packet_start_code. Takes on the value 0 x 00 00 01.	As defined
stream_id	Specifies the type and number of the stream, to which the packet belongs. 1011 1100 – Reserved Stream 1011 1101 – Private Stream 1 1011 1110 – Padding Stream 1011 1111 – Private Stream 2 110x xxxx – MPEG Audio Stream Number xxxxx 1110 xxxx – MPEG Video Stream Number xxxx 1111 xxxx – Reserved Data Stream Number xxxx	As defined
PES_packet_length	Specifies the number of bytes remaining in the packet after the this Field.	0 x 00 00 - Only allowed value for video. Details for audio yet to be determined

6.1. PES header Flags

A breakdown of the PES header flags is shown in Figure 6.2. These flags are a combination of indicators of the properties of the bit stream and indicators of the existence of additional fields in the PES header. The following table describes the flags present the header. The flags not supported by the GA system are set to '0' and form the basis of some of the "constraints" discussed earlier. (These entries are shaded in the table.) Note that the abbreviations in the tables are from the flag names in Fig. 6.2.

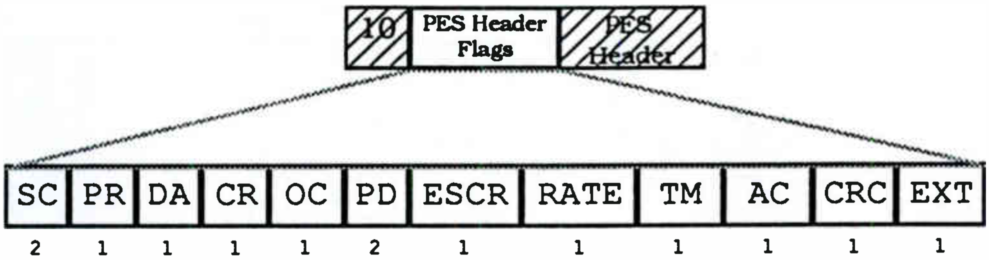

Figure 6.2. PES header flags in their relative positions (all sizes in bits)

Flag	Description	GA Usage
PES_scrambling_control (SC)	Indicates the scrambling of the PES packet payload 00 -- Not Scrambled 01 -- User Defined 10 -- User Defined 11 -- User Defined	Set to 00.
PES_priority (PR)	Indicates the priority of this packet with respect to other packets whose PR field is not set. 1 -- Higher priority 0 -- Same priority	GA does not care how this field is set. (It may be used by applications where necessary.)
data_alignment_indicator (DA)	Indicates the nature of alignment of the first start code occurring in the payload. The type of data in the payload is indicated by the data_stream_alignment_descriptor. 1 - Aligned 0 - No indication of alignment	Must be aligned for video. To be decided for Audio.
copyright (CR)	Indicates the copyright nature of the associated PES packet payload. 1 -- Copyrighted 0 -- Not copyrighted	The GA has yet to define its use of this field.
original_or_copy (OC)	Indicates whether the associated PES packet payload is the original program or a copy. 1 -- Original 0 -- Copy	The GA has yet to define its use of this field.
PTS_DTS_flags (PD)	This flag indicates whether the PTS or PTS and DTS are in the PES header. 00 -- Neither PTS nor DTS is present in header. 1x -- PTS field present 11 -- PTS and DTS field present in header.	The PTS flag is set when video data alignment indicator is set. The DTS may be included to signal the decoder of any special decoding requirements. The PTS transmissions should be spaced less than 700 msecs apart.
ESCR_flag (ESCR)	Indicates whether the Elementary Stream Clock Reference field is present in the PES Header.	Set to '0'.
ES_rate_flag (RATE)	Indicates whether the Elementary Stream Rate field is present in the PES Header.	Set to '0'.
DSM_trick_mode_flag (TM)	Indicates the presence of an 8 bit field describing the mode of operation of the DSM (Digital Storage Media). 1 -- Field present 0 -- Field not present	Set to '0' for the broadcast transmission. May be used for trick modes when using bit streams specifically generated for VCR type operations.
additional_copy_info_flag (AC)	Indicates the presence of the additional_copy_info field. 1 -- Field Present 0 -- Field not present	The GA has yet to define its use of this field.
PES_CRC_flag (CRC)	Indicates whether a CRC field is present in the PES packet.	Set to '0'.
PES_extension_flag (EXT)	This flag is set as necessary to indicate that extension flags are set in the PES header. Its use includes support of private data. 1 -- Field present 0 -- Field not present	As defined.

6.2. The PES header

The PES header immediately follows the field PES_header_length, which indicates the header size in bytes. The size of the header includes all the header fields, any extension fields, and stuffing_bytes. The flags described in the previous section indicate the organization of the PES header, i.e. which fields it does and does not contain. In essence, all the fields of the PES header are optional. Certain applications require particular fields to be set appropriately. For example, GA transport of video PES packets requires that the data_alignment_indicator be set. The trick mode flag is not set in this case. For DSM retrieval of video, the opposite is true. It is the application encoder's function to set the appropriate flags, and encode the corresponding fields. The fields are further described in the following sections. The association between the flags and the corresponding fields is obvious.

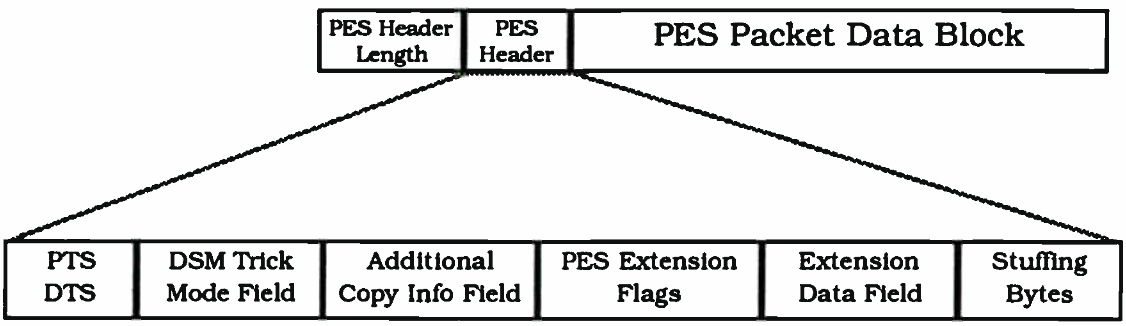

Figure 6.3. Organization of PES header.

The PES header Fields are organized according to Fig. 6.3 for the GA PES packets for video elementary streams. Most fields require marker bits to be inserted, as described later, in order to avoid the occurrence of long strings of 0's which could resemble a start code.

PTS and DTS

The presentation_time_stamp (PTS) informs the decoder of the intended time of presentation of a presentation unit, and the decoding_time_stamp (PTS) is the intended time of decoding of an access unit. An access unit is an encoded presentation unit. When it is encoded, the PTS refers to the presentation unit corresponding to the first access unit occurring in the packet. If an access unit does not occur in a PES packet, it shall not contain a PTS. Here, a video access unit is said to occur if the first byte of the picture start code is present in the PES packet payload, and an audio access unit occurs if the first byte of the synchronization word of an audio frame is present. Under normal conditions, the DTS may be derived from the PTS. So, it is not required to encode the DTS. Consequently, encoding the DTS may indicate special decoding requirements to the decoder. Under no circumstance does the DTS occur by itself; it must occur along with the PTS although the converse is not true. The PTS field is organized as shown, if it present without the DTS.

4 bits	36 bits
0010	PTS Field

PTS 32 ... 30	marker bit	PTS 29 ... 15	marker bit	PTS 14 ... 0	marker bit
3 bits	1 bit	15 bits	1 bit	15 bits	1 bit

Figure 6.4. Organization of the PTS field when only the PTS is encoded.

If both the PTS and DTS are sent, the following organization is required.

4 bits	36 bits	4 bits	36 bits
0011	PTS Field	0001	DTS Field

Figure 6.5. Organization of the PTS and DTS field when both PTS and DTS are encoded.

Here, the DTS field is defined in the same manner as the PTS field.

DSM Trick Mode Field[5]

The DSM trick mode field is an eight bit field, indicating the nature of the information encoded in the PES packet. The first three bits of this field form the identifier trick_mode_control, indicating the nature of the DSM mode. There are four modes to the DSM, as summarized in the table below.

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[5] It is emphasized once again that the ability to deal with trick mode bit streams is not a basic requirement for a GA receiver and this description is here only for completeness. This field is not present in the broadcast GA bit stream. Since the objective in this document is not to describe in detail the DSM mode of operation, all that is presented is the DSM syntax with some description of the respective fields. For more detailed information the reader is directed to the MPEG-2 Systems document.

Value	Description
'000'	Fast Forward
'001'	Slow Motion
'010'	Freeze Frame
'011'	Fast Reverse
'1xx'	Reserved

DSM Trick Modes

Depending on the value of trick_mode_control, a combination of four identifiers are encoded as described in the table below.

Identifier	Description
field_id	This identifier is valid for interlaced pictures only. It is a 2 bit field which identifies how the current frame is to be displayed. '00' -- Display field 1 only '01' -- Display field 2 only '10' -- Display complete frame '11' -- Reserved
frequency_truncation	This 2 bit field indicates the selection of coefficients from the DSM. '00' - Only DC coefficients are sent '01' - The first three coefficients in scan order on average '10' - The first six coefficients in scan order on average This field is for informational purposes only. i.e. the DSM may at times send more than the specified number of coefficients and at other times less. However, that information is not normative.
intra_slice_refresh	This 1 bit field indicates that each picture is composed of intra slices with possible gaps between them. The decoder should replace the missing slices by repeating the colocated sites from the previous decoded picture.
field_rep_cntrl	This field indicates how many times the decoder should repeat field #1 as both top and bottom fields alternatively. After field #1 has been displayed, the decoder should repeat field #2 the same number of times. This identifier being set to 0 is equivalent to a freeze frame with field_id being set to '10'.

Fast Forward and Fast Reverse Modes: The format in this case is shown in Fig. 6.6. The decoder is told how many coefficients are encoded, how the fields are to be displayed, and how to replace any missing slices in the access units.

trick_mode_control	field_id	intra_slice_refresh	frequency_truncation
3 bits	2 bits	1 bit	2 bits

Fig. 6.6. Trick Mode Field in Fast Forward and Fast Reverse Modes

Slow Motion Mode: In this mode, the DSM informs the decoder how many times a particular field is to be repeated. The identifier field_rep_cntrl is encoded as shown.

trick_mode_control	field_rep_control
3 bits	5 bits

Fig. 6.7. Trick Mode Field in Slow Motion Mode

Freeze Frame Mode: Only the identifier is encoded in this mode. It indicates to the decoder how to display the frozen picture. The field is organized as shown in Figure 6.8.

trick_mode_control	field_id	Reserved
3 bits	5 bits	5 bits

Fig. 6.8. Organization of Trick Mode Field for Freeze Frame Mode.

Additional Copy Info

This is a one byte field with a marker bit up front and 7 bits of information. The use of these seven bits is yet to be determined.

PES Extension Flags

The header could contain additional flags if the EXT flag (shown in Fig. 6.2) is set. These flags are transmitted in a one byte data field as shown in Fig. 6.9.

PES private data flag	pack header field flag	program packet sequence counter flag	STD Buffer Flag	Reserved	PES extension field flag
1 bit	1 bit	1 bit	1 bit	3 bits	1 bit

Figure 6.9. Organization of the PES Extension Flags Field.

The flags indicate whether further extensions to the PES header exist. The table below describes the nature of this additional data. As with the flags defined previously, the flag is set to '1' if the header field is present.

Field	General Description
PES_private_data_flag	Indicates whether the PES packet contains private data.
pack_header_field_flag	Indicates whether an MPEG-1 systems pack header or an MPEG-2 program stream pack header is present in the PES Header.
program_packet_sequence_counter_flag	Indicates the presence of a PES packet counter, which allows the decoder to retrieve the packets in the correct order, from the PES/program multiplex.
STD_buffer_flag	Indicates whether the STD_buffer_scale and the STD_buffer_size flags are encoded in the PES header.
PES_extension_field_flag	Indicates the presence of additional data in PES header.

For the GA system, as indicated in the constraint document submitted to MPEG, the flags that are shaded in the table are always set to '0'. It is likely that the EXT flag will set to '0' in the header flags field unless there is information to be transmitted in the PES_extension_field. This field is meant for functions that may have been missed in the initial design specification.

PES Extension Field

This field is shown in Fig. 6.10. The length of the PES_extension_field data is given by PES_extension_field_length.

marker bit	PES Extension Field Length	Reserved
1 bit	7 bits	

Figure 6.10. Organization of the Extension Field

7. Conditional Access

The transport protocol implements functions useful for supporting conditional access. The functionality that is available is flexible and complete in the sense of supporting all transmission aspects of applicable key encryption and descrambling approaches that may be used. Conditional access is also flexible in the sense that it can be exercised on a elementary stream by stream basis, including the ability to selectively scramble bit streams in a program if desired.

A conditional access system operates on the principle of randomizing the transmitted data so that unauthorized decoders cannot decode the signal. Authorized decoders are delivered a "key" which initializes the circuit which inverts the bit randomization. In subsequent discussion, we use the term scrambling to mean the pseudo-random inversion of data bits based on a "key" which is valid for a short time. We use the term encryption to mean the process of transforming the "key" into an encrypted key by a means which protects the key from unauthorized users. From a cryptographic point of the view, this transformation of the key is the only part of the system which protects the data from a highly motivated pirate. The scrambling portion of the process alone, in the absence of key encryption, can be defeated. Conditional Access (CA) is a blanket term for the system which implements the key encryption and distribution. The primary requirements which a scrambling and CA subsystem must meet for digital TV delivery are:

- Protection of programmer's revenues
  - robust against piracy.
- Private encryption system for each program provider.
- Standard consumer instruments
  - no secrets in consumer equipment
- Mobility of consumer equipment
- Consumer equipment should be cost effective

7.1. General Description

There are two features of the GA Transport system which support conditional access. The first feature is the two bit transport_scrambling_control field which signals the decoder whether the transport packet was scrambled or not. In the case that it was scrambled, the field identifies which scrambling key was used. As will be shown shortly, the use of two bits in the transport_scrambling_control to define the descrambling process is a necessary and sufficient bound for the key distribution function. The second feature is the ability to insert "private" data at several places in the GA Transport stream. These include entirely private streams and private fields in the adaptation header of the transport bit stream being scrambled. These private fields can be used to transmit the encrypted scrambling key to the decoding device.

The key distribution and usage process is clarified in Fig. 7.1. Basically, when the bit stream is scrambled, one descrambling key needs to be in use while the other is being received and decrypted. Two keys are transmitted at any time, with the keys being linked to a transport_scrambling_control value as shown in the figure. The transmission of a key should begin well before it is going to be used, to allow time to decrypt it. Note that this function does not bound the total number of keys that may be used during an entire transmission session.

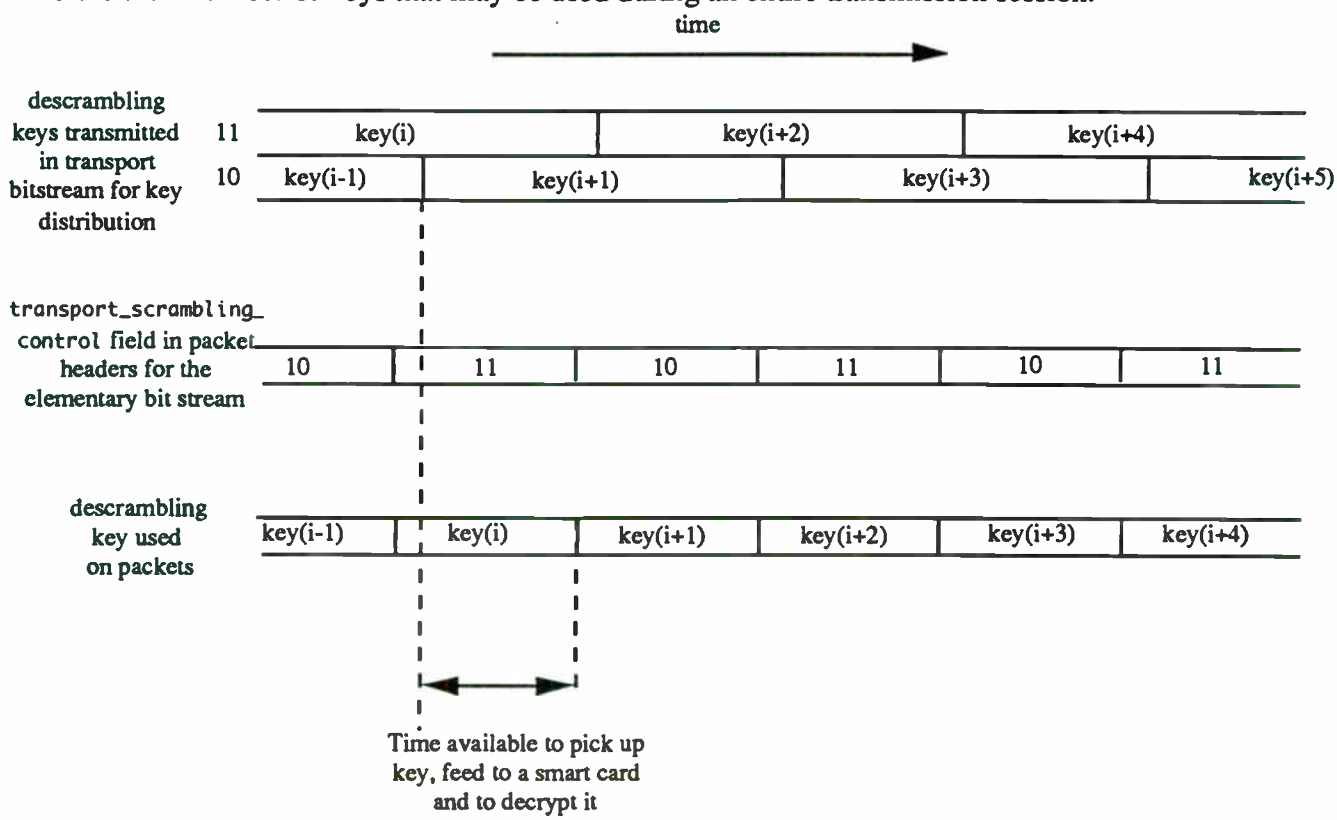

Fig. 7.1. Illustration of key distribution and usage process.

As stated previously, the amount of data to be scrambled in a packet is variable depending on the length of the adaptation header. It should be noted that some padding of the adaptation field might be necessary for certain block mode algorithms.

7.2. Example of Conditional Access Implementation

In this section, we go through a simple example of a conditional access implementation. Consider the receiver architecture shown in Figure 7.2. The high speed manipulations required to implement the descrambling are embedded in the transport demultiplexer, where they are shown as a DES block. Note that other scrambling schemes, such as stream ciphers based on a Pseudo Random Binary Sequence (PRBS), could be employed. The PRBS uses a shift register implementation, where it the initial register value is reset periodically for error robustness. The data security is achieved by the "key" which properly configures the descrambler. This element is delivered to the decoder through an ancillary data service, and is encrypted by the conditional access administrator. In the equipment at the customer's premises, the key is decrypted within the outboard Smart-Card. The Smart-Card interface will conform to ISO standard ISO-7816, which permits a variety of implementations and conditional access solutions.

The Smart-Card maintains a short list of two key's, commonly denoted as the "odd" key and the "even" key. The proper key to be used to descramble is signaled in the transport prefix in the transport_scrambling_control field. The transport_scrambling_control takes on one of the following 4 states:

transport_scrambling_control	*Description*
00	Not Scrambled
01	Reserved
10	"even" key
11	"odd" key

The scrambling in this example is a block cipher called the "Electronic Code Book" mode of the Digital Encryption Standard (DES). For DES, the key is a 56 bit binary sequence. The television electronics are "standard", while the Smart-Card implementation is proprietary to the service provider.

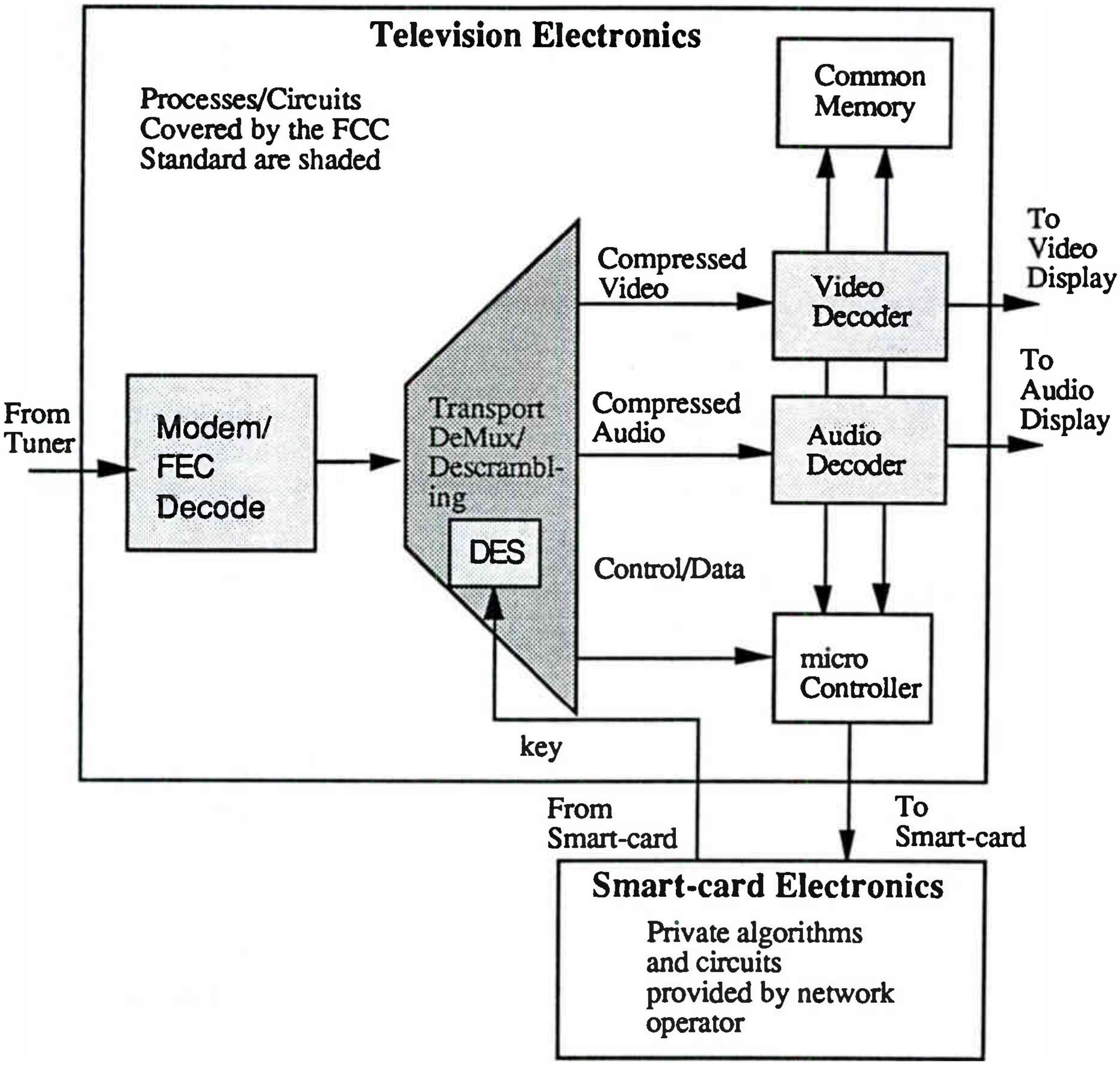

Fig. 7.2. Decoder with Example Conditional Access Implementation

There is significant flexibility for multihop encryption and nesting of encryption systems to ensure data security at every point in the transmission chain. Fig. 7.3 illustrates nested encryption systems, where the service provider provides authorization, keys and the scrambled data to the end user through Encryption System A. During transit, System B is employed by the carrier to protect the data while in the communications network. In fact, there are two implementation choices even for this segment of the delivery system. The provider can use both a second layer of scrambling in conjunction with a different authorization and key distribution. (This system need not comply with the transmission "Standard's" method.) Alternatively, System B could simply encrypt or scramble the encrypted keys distributed by System A, without the requirement of scrambling the actual service data. The main appeal of this method is that it is a low bandwidth/complexity solution, while its drawback is that it requires some knowledge of the System A key distribution method.

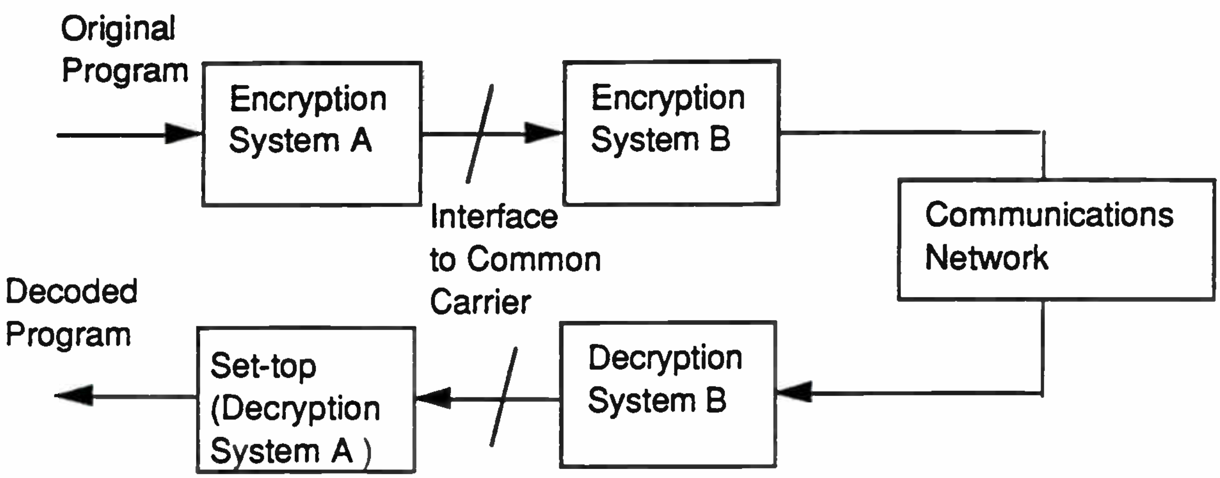

Figure 7.3. Nested encryption systems.

In Fig. 7.4, the two Encryption Systems are connected in series. System B is again used to protect the integrity of the data while in the communications network. System A is used by the local affiliate or cable company to authorize reception of the service within its own service area.

In fact, the two topologies discussed above can be combined, where either System A or System B could be a Nested or Series configuration. The number of nesting/series combinations can be arbitrarily large.

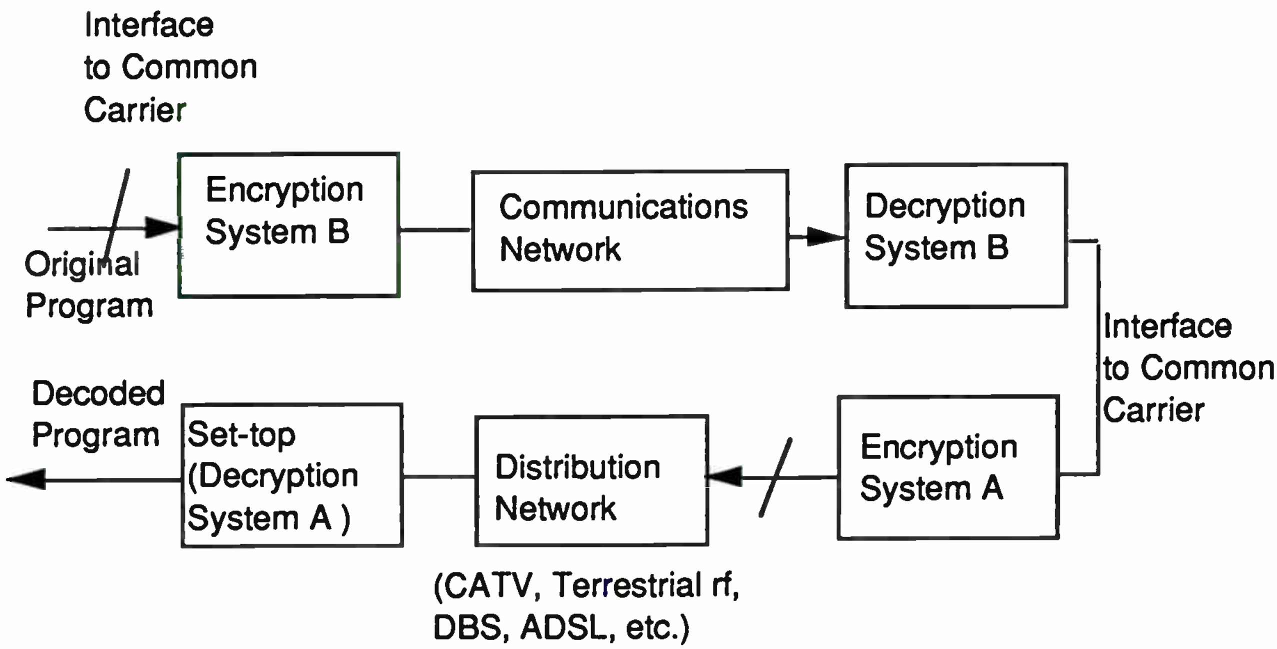

Figure 7.4. Series encryption systems.

8. Local Program Insertion

The Grand Alliance Transport supports insertion of programs and commercials, by use of flags and features dedicated to this purpose in the transport packets Adaptation Header. The use of these syntax elements will need to be within some imposed constraints to ensure proper operation of the video decoders. Furthermore, there will be some constraints on some of current common practices, imposed not by the GA transport, but rather by virtue of the compressed digital data format.

The functionality of program insertion and switching of channels at a broadcast head-end are quite similar, the difference being in the time constants involved in the splicing process, and also in the fact that in the program insertion the bit stream is switched back to the old program after insertion is complete, while in the channel switching case one most likely switches over to yet another program at the end of the splice. There are other detailed issues related to the hardware implementation that may differ for these two cases, including input source devices and buffering requirements. For example, if program insertion is to take place on a bit stream obtained directly from a network feed, and if the network feed does not include place-holders for program insertion, the input program transport stream will need to be buffered up for the duration of the program insertion. If the program is obtained from a local device, e.g., a video server or a tape machine, it may be possible to pause the input process for the duration of the program insertion. Neither of these is an issue for channel switching.

8.1. Systems level view

There are two layers of processing functionality to address when doing program insertion. The lower layer functionality is related to splicing of transport bit streams for the individual elements of the program. The higher level functionality is related to coordination of this process between the different elementary bit streams which make up the program transport stream. Fig. 8.1 illustrates the correct approach to implement program insertion.

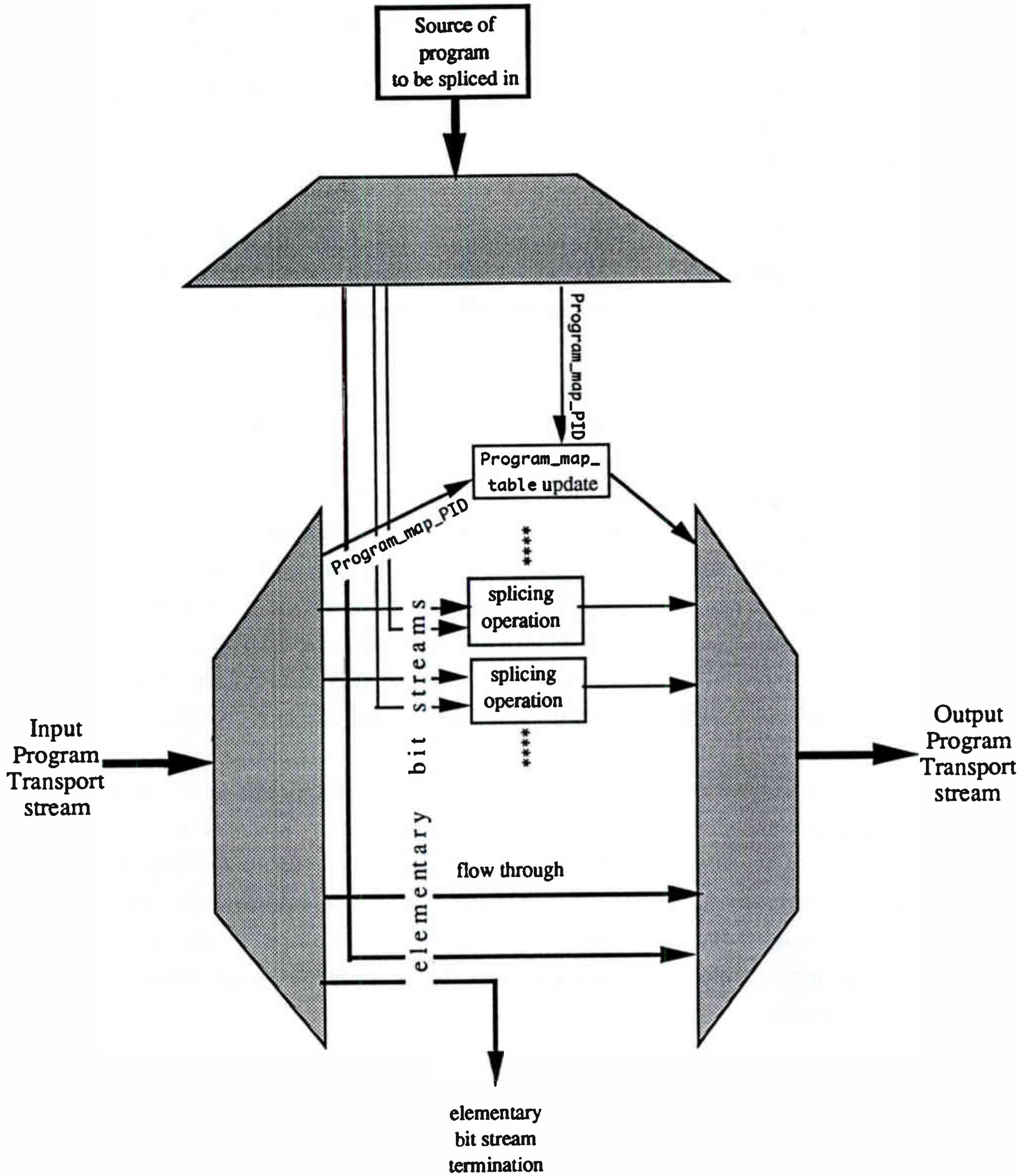

Figure 8.1 Example Program Insertion Architecture

The first step for program insertion to take place at a broadcast head-end is to extract (by demultiplexing) the packets, identified by the PIDs, of the individual elementary bit streams that make up the program, including the bit stream carrying the program_map_table. Once these packets have been extracted, as illustrated in Fig. 8.1, program insertion can take place on an individual PID basis. If applicable, some packets may be passed through without modification. There is also the flexibility to add and drop elementary bit streams. The splicing process for each PID is described in the next section.

When program insertion takes place, the program_map_table needs to be modified to reflect the properties of the program transport stream that is being spliced in. As described in the section on its syntax, the definition of the program_map_table allows the signaling of a change in the contents of a program transport stream ahead of time. The changes in the program definition could involve a change in the number of elementary bit streams that make up the program, either by addition or removal of bit streams, change in the PIDs used for the elementary bit streams, etc...

The bit-rate of the program after splicing should have a known relationship to the bit-rate of the program before splicing, in most scenarios. Unless dynamic bit-rate allocation is possible for a program at the system multiplexer (based on instantaneous bandwidth requirements), an increase in bit-rate after splicing can cause buffer overflow. A decrease in bit rate may be handled by transmitting null packets (packets with no information) or by allocating the extra bandwidth to other programs on a dynamic basis. These capabilities depend on the implementation of the system level multiplexing function, a function that is not a part of the GA Transport specification.

Bit rate constraints may also be imposed on individual elementary bit streams for the program that is inserted, e.g., for compressed video, input and output bit rates need to be the same. In a perfect program insertion set up, the splice points for the different elementary streams in a program should be coordinated to correspond to the same instant in time in the overall program (which may not correspond to the same instant in time for each elementary bit stream) to permit seamless transition. Additional constraints on selecting the splicing points exist for particular applications such as video (i.e., VBV_delay value).

8.2. Basics of elementary bit stream insertion

The interface for elementary bit stream insertion is at the transport layer of the protocol. This means that bit stream insertion always takes place in units of transport packets. The primary features enabling local elementary bit stream insertion are the discontinuity_indicator field and the splice_countdown fields in the transport header. The discontinuity_indicator signals the decoder that the PCR is changing to a new time base. This simply informs the decoder that the change in the bit stream is not due to an error in the channel, but rather is intended by the program provider. The implication for the decoder is that it should continue normal decoding, and it is the encoders responsibility to make sure that the bit stream has been constructed in a compliant manner (that is that decoders don't crash due to overflow or underflow.)

The splice_countdown field in the adaptation header is used to signal a head-end or intermediate digital switch that a subsequent packet is the point for switching in a new bit stream. The count-down is a positive number which decrements on each subsequent packet of that service. The value of "0" is the last packet in the original sequence, and the value "-1" is resident in the packet which should initiate the switch over. The count will continue to decrement for channel error resilience. The behavior is shown in Figure 8.2.

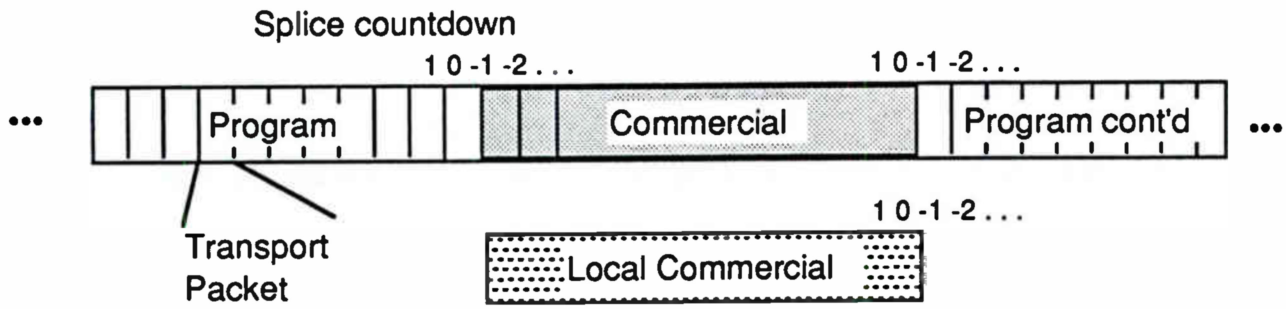

Figure 8.2. Local program insertion keyed on Splice countdown.

The affiliate or headend equipment sets itself up for the switch based on the descending countdown. At the "-1" point, the local commercial is inserted while the network commercial continues. At the second trailing "-1", the affiliate returns to the network feed. This technique can be used for either local programs or local commercials. It does depend on the video encoder constraining its bit generation at the splice points so that the decoder buffer does not overflow.

The GA transport encoder places some constraints on the encoding which are more stringent than the MPEG-2 requirements. The added constraint is that the PES header is followed immediately by a video access unit. This will speed acquisition. The first packet in an insertion will contain the PCR value, with the PCR discontinuity bit set to "1" to inform the decoder that a splice has occurred. The first payload in the stream will begin with a PES Header, which will have a PTS resident, so that the decoder can determine the display time immediately. Because the PES header also has the data_alignment_indicator set, the first data following the header will be the start of the video sequence layer. Consequently, the decoder has all the information available to begin decoding immediately after receiving the beginning of the spliced commercial. (In general, an MPEG-2 stream does not have these constraints imposed, and hence does not have guaranteed performance at the splice points.)

8.3. Restrictions

Compressed digital technology does impose some restrictions on affiliate operations which may differ from present practice. Although all present practices can be replicated by completely decoding and recoding the video, there is a desire to implement as much as possible in the compressed video domain. The following comments are made with respect to processing the compressed video.

Local commercials and network commercials will need to be strictly controlled to be the same number of packets, and the same number of frames of video. This is contrary to the present practice where local inserts may differ from the planned network inserts by several seconds.

A second restriction is that affiliate pix-in-pix and affiliate text overlays can only be accomplished by decoding and recoding.

8.4. Imperfect program insertion

It may not always be possible for the program insertion process to meet the precise requirements of a seamless splice. This could be due to several reasons including the presence of infrequent splice points in the incoming bit stream or non-availability of the hardware required for precise splicing at the network affiliate. There are two scenarios for imperfect splicing. In the first scenario the network affiliate attempts to splice in the entire program as a whole, without attempting to align each of the component elementary bit streams. In this case, the exact splicing can take place for only one of the elementary streams. Since video is the most important component of the program, perfect alignment will be obtained for video. In this case the output of the other elementary bit streams will not be presented at the output until synchronization of these bit streams is achieved. As an example, audio should be muted until its elementary bit stream is synchronized.

In the second and most uncoordinated splicing approach, the splicing takes place without any attempt at coordination with the input bit stream. In this case the video presentation process is affected around the splicing point. If the splicing takes place when the VBV level in the existing bit stream is less than it should be for perfect splice, there will be a period of time for which data is lost for the existing bit stream. In this case the decoder should freeze the last displayed frame. In the other case where VBV is fuller than expected, the decoder video data buffer may eventually overflow during the time period of the spliced in bit stream. The decoder will then have to initiate a resynchronization procedure in the middle of the program, freezing the display to the last decoded picture while this process is taking place. Note that when the process of splicing in a bit stream does not take place correctly, there will also be a disruption in service at the splice back to the original bit stream. It is the recommendation of the GA that this type of splicing be strictly prohibited, since it leads to a very noticeable interruption of service.

It is important to note that the process of facilitating frequent opportunities for splicing in a program bit stream is not within the control of the transport layer of the system. The transport only provides the mechanism of implementing the splice itself. Hence decisions on determining the possible frequency of commercial insertion should also involve the people involved in the design of the source coding algorithms for applications like video and audio.

9. Compatibility with other Transport Systems

The GA transport system is compatible with two of the most important alternative transport systems, namely the MPEG-2 transport stream definition, and also the ATM definition being finalized for Broadband ISDN. Furthermore, since several of the CATV (e.g., Digicipher II) and DBS systems being designed are considering use of the MPEG-2 Transport layer syntax, the degree of interoperability with such deployed systems should be quite high (possibly requiring a translation if the CATV or DBS system deploys a slightly incompatible MPEG-2 variant).

9.1. Interoperability with MPEG-2

In the development of the GA transport specification, the intent has never been to limit the design by the scope of the MPEG-2 systems definition. The GA system is interoperable with MPEG-2 decoders since the GA Transport is currently a constrained subset of the MPEG-2 Transport syntax. The constraints are imposed for reasons of increased performance of channel acquisition, bandwidth efficiency and decoder complexity. If, in the course of future work, the MPEG-2 standard is unable to efficiently meet the requirements of the GA system, a deviation from MPEG would be in order.

The ATV system requires definition of bit streams and services beyond the compressed video and audio services. A means of identifying such bit streams is necessary in the ATV system, but is not part of the MPEG-2 definition. There is a method of encoding such a registration descriptor when a an authority to administrate registration is identified. This identification is implemented by the registration_descriptor in the PSI stream.

9.2. Interoperability with ATM

The GA transport packet size is selected to ease transferring these packets in a link layer that supports Asynchronous Transfer Mode (ATM) transmission. There are several methods for mapping the Transport packet into the ATM format. Three techniques are presented, although the industry may converge to a different solution than those presented here.

9.2.1. ATM Cell and Transport Packet Structures

Figure 9.1 shows the format of an ATM cell. The cell consists of two parts: a five byte header and a forty-eight byte information field. The header, primarily significant for networking purposes, consists of the following fields:

GFC	a four bit Generic Flow Control field used to control the flow of traffic across the User Network Interface (UNI). Exact mechanisms for flow control are under investigation.
VPI	an eight bit network Virtual Path Identifier.
VCI	a sixteen bit network Virtual Circuit Identifier.
PT	a three bit Payload Type (i.e., user information type ID).
CLP	a one bit Cell Loss Priority flag (eligibility of the cell for discard by the network under congested conditions).
HEC	an eight bit Header Error Control field for ATM header error correction
AAL	ATM Adaptation Layer bytes (user specific header).

The ATM User Data Field consists of forty-eight bytes, where up to four of these bytes can be allocated to an Adaptation Layer.

Figure 9.2 shows the format of the Grand Alliance transport packet. A one hundred eighty-four byte packet data field (possibly including an optional and conditional adaptation field) is preceded by a four byte prefix.

9.2.2. Null AAL Byte ATM Cell Formation

The simplest method to form ATM cells from the Transport layer is the null AAL byte structure shown in Figure 9.3. The Transport packet is partitioned into forty-eight byte payloads, applied directly to the information fields of the ATM cell. The five byte ATM header is appended. Since the Transport packet length is not an integer multiple of the ATM cell payload, there will be only occasional alignment of the Transport header with the start of the ATM cell information field.

9.2.3. Single AAL Byte ATM Cell Formation

Alignment of the Transport packet and ATM cell is accommodated by parsing the Transport packet into forty-seven byte segments, shown in Figure 9.4. Four such segments will exactly encompass a Transport packet. A one byte AAL is appended, along with the five byte ATM header to fulfill the fifty-three byte ATM cell requirement. The AAL byte can carry useful information concerning the transport data within the ATM cell. It can be viewed as an adaptation field for the contained data, conveying the original position of the ATM payload within the Transport packet, for example, as well as other information. For example, ATM standards presently provide for five different AALs, such as AAL Type 1 for accommodating connection oriented constant bit rate services, and AAL Type 2 for handling connection oriented variable bit rate data services.

9.2.4. Dual AAL Byte ATM Cell Formation

An alternative solution to cell/packet alignment is shown in Figure 9.5. The transport header is discarded, and the remaining one hundred eighty-four byte payload is segmented into 46 byte increments. To these are added two AAL bytes and the five byte ATM header for each ATM packet. The idea here is that there may be a duplication in functionality in the ATM header and the link level transport header fields. A particular header field to consider for duplication of functionality is the PID. If the PID can be associated with a specific VPI and VCI used in the ATM headers of the packets carrying the data payload, and this PID mapping information can be sent to the destination terminal when the virtual path/circuit is set up (using the ATM signaling channel), it does not have to be transmitted for every GA packet. The PID can then be reconstructed at the destination (using the information transmitted at call setup), and can then be appended to the one hundred eighty-four byte payload (reconstructed from four ATM packets) to obtain the complete GA packets. Transport header information that cannot be reconstructed (e.g., adaptation field control) should be carried as a part of the two AAL bytes for each ATM cell, along with other additional information. Note that the above is only a suggested approach and does not represent a complete design.

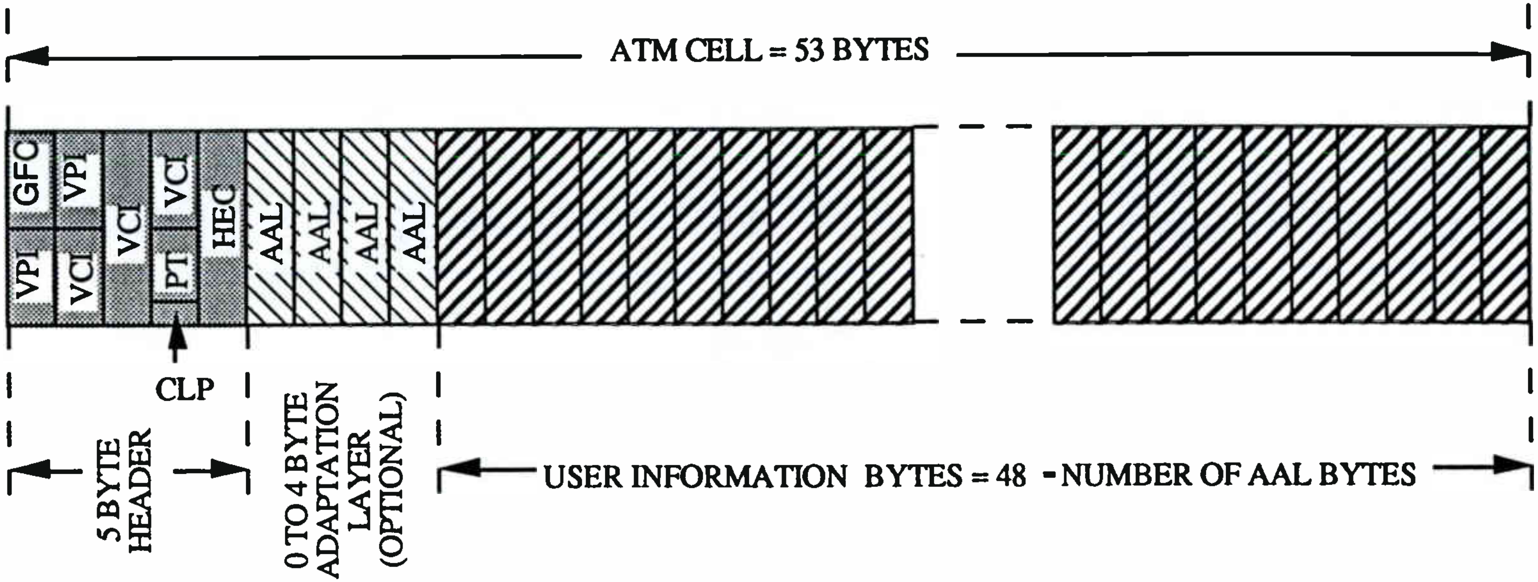

FIGURE 9.1 STRUCTURE OF THE ATM CELL

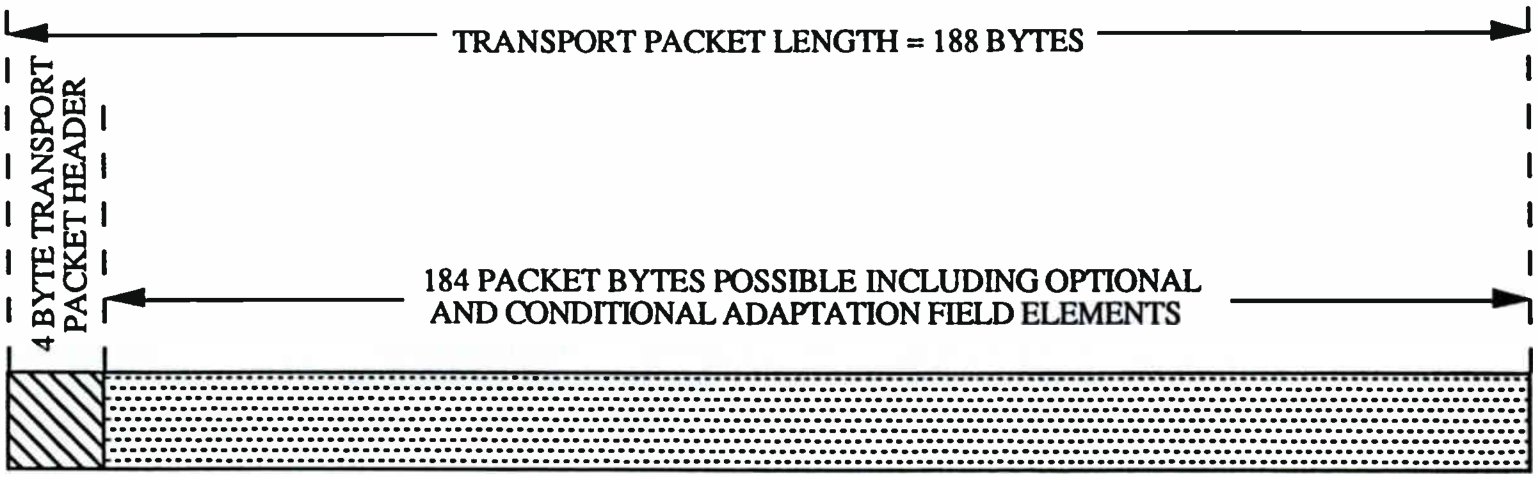

FIGURE 9 .2 STRUCTURE OF THE TRANSPORT PACKET

AAA 7/29/93

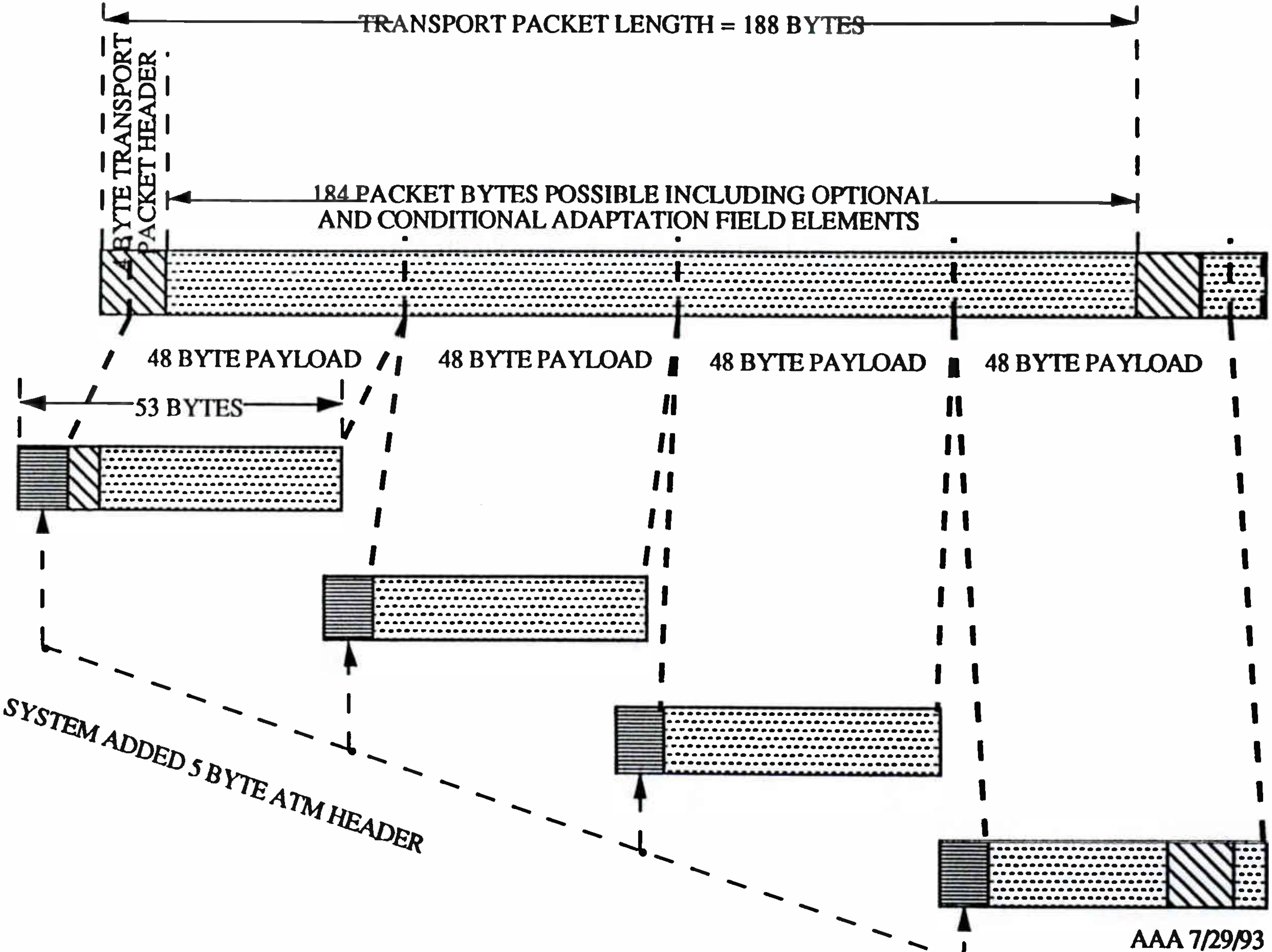

FIGURE 9.3. FORMATION OF ATM CELL WITH NO AAL BYTES

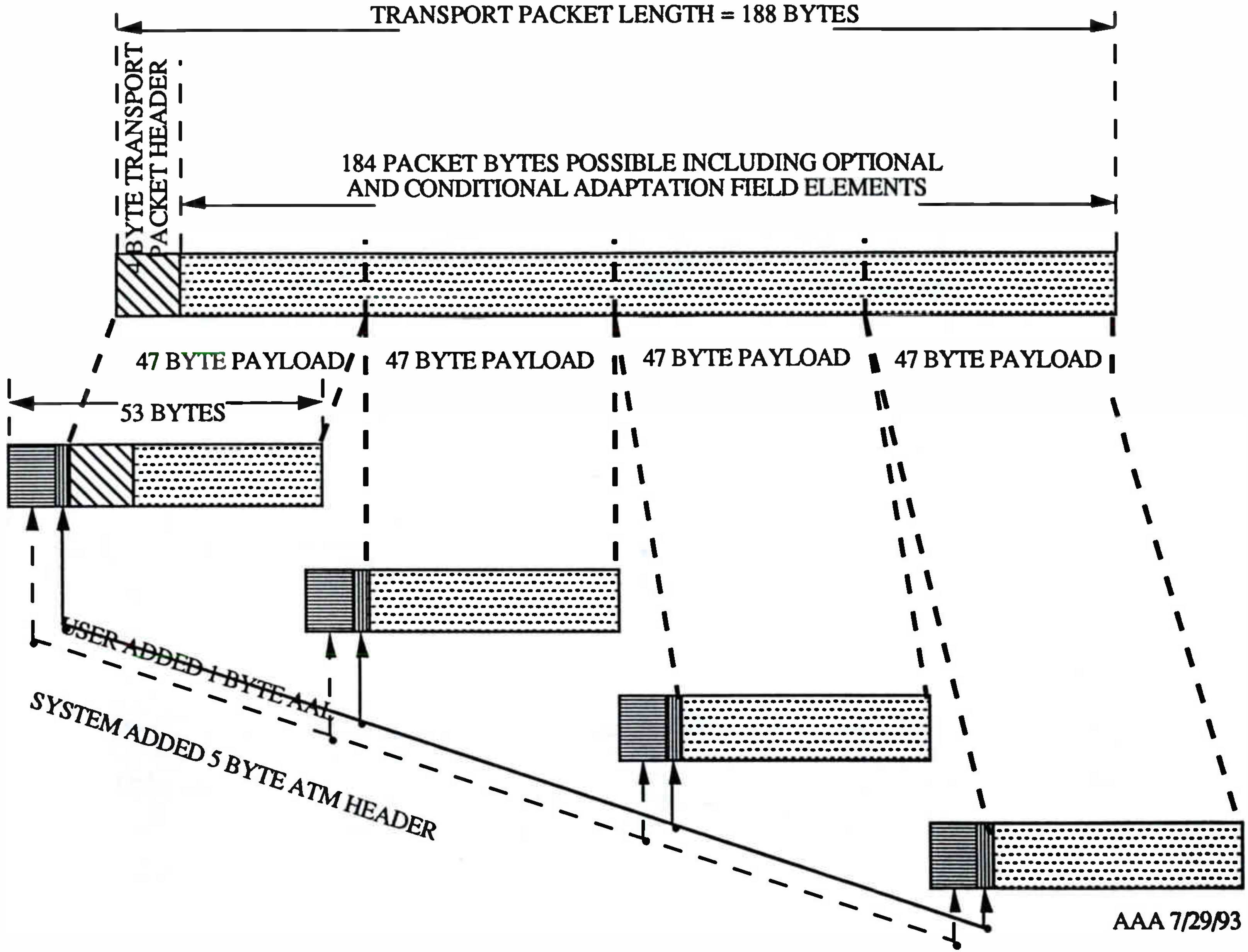

FIGURE 9.4. FORMATION OF THE ATM CELLS WITH A SINGLE AAL BYTE

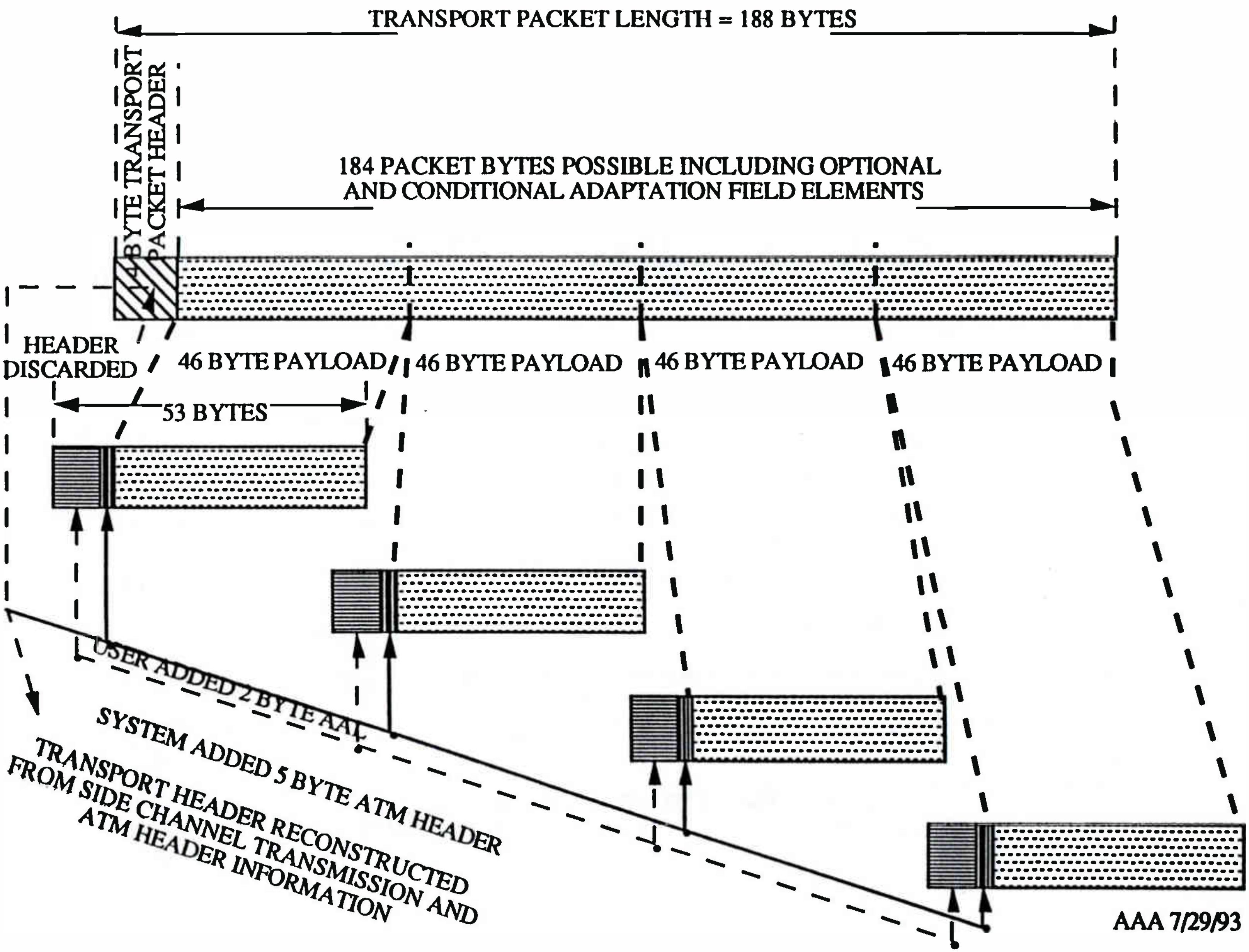

FIGURE 9.5. FORMATION OF AN ATM CELL WITH DUAL AAL BYTE

Chapter VI
TRANSMISSION SYSTEM

VSB TRANSMISSION SYSTEM DESCRIPTION

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GRAND ALLIANCE VSB TRANSMISSION SUBSYSTEM DESCRIPTION

1.0 INTRODUCTION AND SYSTEM OVERVIEW

The Grand Alliance (G-A) vestigial sideband (VSB) digital transmission system provides the basis for a family of transmission systems suitable for data transmission over a variety of media. This family shares the same pilot, symbol rate, data frame structure, interleaving, Reed-Solomon coding, and synchronization pulses. The VSB system has two modes: a simulcast terrestrial broadcast mode, and a high data rate cable mode. The terrestrial broadcast mode, which transmits an ATV signal on currently unusable taboo NTSC channels with minimal interference to or from NTSC channels, is optimized for maximum coverage area, and supports one ATV signal in a 6 MHz channel. The high data rate cable mode, which trades off some robustness for twice the data rate, supports two ATV signals in one 6 MHz channel.

The VSB transmission systems (both modes) take advantage of a pilot, a segment sync, and a training sequence for robust acquisition and operation. The two systems also share identical carrier, sync, and clock recovery circuits, as well as phase correctors and equalizers. Additionally, both systems use the same Reed-Solomon (R-S) code for forward error correction (FEC).

In order to maximize coverage area, the terrestrial broadcast mode incorporates both an NTSC rejection filter (in the receiver) and trellis coding. In contrast to the 4-VSB system tested during the first round of ATV testing, precoding at the transmitter is incorporated in the trellis code. When the NTSC rejection filter is activated in the receiver, the trellis decoder is switched to a trellis code corresponding to the encoder trellis code concatenated with the filter.

The cable mode, on the other hand, does not have as severe an environment to work in as that of the terrestrial system. Therefore, a higher data rate is transmitted in the form of more data levels (bits/symbol). For cost considerations, no trellis coding or NTSC interference rejection filters are employed.

VSB transmission inherently requires only processing the in-phase (I) channel signal, sampled at the symbol rate, thus optimizing the receiver for low cost implementation. The decoder only requires one A/D converter and a real equalizer operating at the symbol rate of 10.76 Msamples/second.

The parameters for the two VSB transmission modes are shown in Table 1.

2.0 TERRESTRIAL VSB SYSTEM DESCRIPTION

2.1 SYSTEM INFORMATION

The VSB transmission system transmits data according to the data frame depicted in Fig 1. The frame is organized into segments each with 836 symbols. Each transmitted segment consists of a four symbol segment sync followed by 832 data plus FEC symbols. The data contained in each transmitted 208 byte segment is a 188 byte MPEG-compatible data packet with 20 R-S parity bytes.

The exact symbol rate is:

4.5/286 Mhz x 684 = 10.76 MHz

The frequency of a segment may be calculated as follows:

f_{seg}=10.76 Msymbols/sec / 836 symbols/seg = 12.87 ksymbols/sec

For terrestrial broadcast mode, each segment corresponds to one R-S correction block of 208 bytes as follows:

208 bytes/block x 8 bits x 3/2 trellis = 3 bits/symbol x 832 symbols

For the high data rate cable mode, each segment corresponds to two R-S correction blocks of 208 bytes as follows:

2 blocks x 208 bytes/block x 8 bits = 4 bits/symbol x 832 symbols

As shown in Fig 1, each data frame begins with a first data field sync segment followed by 312 data segments, a second data field sync segment, another 312 data segments.

Except for the binary data segment and data field syncs, all other transmitted symbols are multi-level. For the terrestrial broadcast mode, 8-level symbols (3 bits/symbol) are transmitted while for the high data rate cable mode, 16-level symbols (4 bits/symbol) are used. These are called 8-VSB and 16-VSB, respectively.

The multi-level symbols combined with the data segment and data field syncs are used to suppress-carrier modulate a single carrier. However, before transmission, the lower sideband is removed. The resulting spectrum is flat, except for the band edges where a root-raised cosine response results in 620 KHz transition regions. The VSB transmission spectrum relative to an NTSC spectrum is shown in Fig 2. The cumulative distribution function (CDF) of the peak-to-average power ratio of a typical transmitted signal is plotted in Fig 3, with 99.9% of the power envelope within 6.3 dB of the average power.

At the suppressed carrier frequency of 310 KHz from the lower band edge, a small pilot is also added to the signal. The pilot is used in the VSB receiver to achieve carrier lock. Pilot power adds 0.3 dB to the total signal power but helps reduce implementation loss by more than that. With the aid of the pilot, the VSB transmission system achieves virtual theoretical performance (i.e. no implementation loss). The pilot is positioned in the vestigial sideband region of a cochannel NTSC signal and does not contribute to cochannel interference into NTSC.

The terrestrial VSB system was designed with robustness in mind. Forward error correction in the form of R-S and trellis coding, along with 1/3 data field interleaving, provide a rugged system that can endure both white noise and interference environments. The terrestrial VSB system can operate in signal-to-noise (S/N) environments of 14.9 dB (3×10^{-6}), as illustrated in the 8-VSB, 4-state segment error probability curve in Fig 4.

2.2 TRANSMITTER BROADCAST MODE

A functional block diagram of the VSB terrestrial broadcast transmitter is shown in Fig 5. Descriptions of each block follow.

2.2.1 DATA RANDOMIZER

A data randomizer is used on all input data to randomize the data payload (NOT including syncs, or R-S parity bytes). This ensures random data is transmitted even when constant data is applied to the system, as might happen when a data input is disconnected. Random data is important for optimum reception of the transmitter data signal.

The data randomizer exclusive OR's all the incoming data bytes with a 16-bit maximum length pseudo random sequence (PRS) locked to the data frame. The PRS is generated in a 16-bit shift register that has 9 feedback taps. Eight of the shift register outputs are selected as the fixed randomizing byte, where each bit from this byte is used to individually exclusive OR the corresponding input data bit.

2.2.2 REED-SOLOMON ENCODER

The Reed-Solomon (R-S) error protection code, known for its burst noise correction capability, is a very efficient block code for its parity overhead. Since it works with bytes (8-bits), any number of bit errors can occur within a byte and not affect its error correction ability. Therefore, there is no direct proportional relationship between byte error rate and bit error rate. Only at very low error rates do the two converge, but the bit error rate is never smaller than the byte error rate for the same S/N ratio. The measure of performance for block codes is its packet error rate versus signal-to-noise ratio or signal-to-interference ratio. Since ATV data is to be transmitted in MPEG transport packets, the most common method of describing data transmission performance in noise and interference conditions is probability of data packet error.

The R-S code used in the VSB transmission system is a t=10 (208,188) code. The data block size is 188 bytes, with 20 R-S parity bytes added for error correction. A total block size of 208 bytes is transmitted per data segment. A t=10 R-S code, with 20 parity bytes, can correct up to 10 byte errors per block.

2.2.3 INTERLEAVER

Although the R-S code is particularly powerful in protecting against burst errors, the data is interleaved for further protection. Burst errors result from both impairments and trellis decoding mistakes where a single error can propagate through the trellis decoder, multiply, and become a burst error. Long trellis decoders (large K) cause correspondingly longer burst errors to be created.

The goal of the interleaver is to spread the data bytes from the same R-S block over time so that a long burst of noise or interference is necessary to encompass more than 10 data bytes and overrun the R-S error protection. Since the R-S code can correct up to 10 byte errors per block regardless of how many individual bit errors have occurred within each byte, it is advantageous to group consecutive symbols from the trellis decoder into bytes to get the best burst error performance.

The interleaver employed in the VSB transmission system is an 87 data segment (intersegment) diagonal byte interleaver. Interleaving is provided to a depth of about 1/3 of a data field (5.5 msec deep). Intrasegment interleaving is also performed for the benefit of the trellis coding process, to be described later.

2.2.4 TRELLIS CODED MODULATION

The terrestrial broadcast VSB transmission mode employs a 2/3 rate (R=2/3) trellis code (with one unencoded bit). That is, one input bit is encoded into two output bits using a 1/2 rate convolution code while the other input bit is left unencoded. The signaling waveform used with the trellis code is an 8-level (3 bit) 1-dimensional constellation. The transmitted signal is commonly referred to as 8-VSB. To minimize the complexity of interleaving and trellis decoder hardware, as well as to minimize error propagation, a relatively simple (short) 4-state trellis encoder is used. Long trellis codes, which cause longer burst errors and require more interleaving, were not found to be beneficial.

Although trellis codes produce improvements in signal-to-noise ratio (S/N) threshold against white noise, they do not perform well for impulse or burst noise. Besides electromechanical sources of burst noise, burst noise is also caused by NTSC cochannel interference and phase noise which can cause data-dependent cross talk. To further reduce the effects of burst errors, and to simplify the trellis decoder when the NTSC rejection filter is used, trellis code interleaving is used. This uses twelve identical trellis encoders operating on interleaved data symbols. The code interleaving is accomplished by encoding symbols {0,12,24,36...} as one group, symbols {1,13,25,37} as a second group, symbols {2,14,26,38,...} as a third group, and so on for a total of 12 groups. The 12 groups are required since the creation of spectral nulls in the receiver requires a comb filter with delays of 12 symbols. This will be discussed further later.

The theoretical performance of the concatenated trellis and R-S code used in the terrestrial broadcast VSB system was shown earlier in Fig 4. A theoretical performance curve for the 4-VSB system as delivered for the first round of ATV testing is also shown for reference. The combination of short trellis codes and code interleaving is a superior error correction technique for use in channels with white gaussian noise and Rayleigh distribution fading.

2.2.5 SYNCHRONIZATION INFORMATION

Synchronization information is added to the digital data signal in order to facilitate packet and symbol clock acquisition and phase- lock during extreme noise and interference conditions. The encoded trellis data is passed through a multiplexer that inserts the various synchronization signals (data segment sync & data frame sync).

A two level (binary) 4-symbol Data Segment Sync is inserted into the 8-level digital data stream at the beginning of each data segment. The data segment sync embedded in random data is illustrated in Fig 6. A complete segment consists of 836 symbols: 4 sync symbols, and 832 data plus parity symbols. The segment sync is binary (2-level) in order to make packet and clock recovery rugged. The levels selected (±5) for segment sync insure robustness, but do not cause undue interference into cochannel NTSC signals. They also have an average value of zero so that when the DC pilot is added, they will not alter the desired value of pilot. The same sync pattern occurs regularly at 77.695 μsec intervals, and are the only signals repeating at this rate. The periodic recurrence of these four symbols makes possible their reliable detection at the receiver under severe noise and/or interference. Unlike the data, the four data segment sync symbols are not Reed-Solomon or trellis encoded, nor are they interleaved.

The data is not only divided into data segments, but also into data fields. Each data field starts with one complete data segment of Data Field Sync, as shown in Fig 7. The Data Field Sync signal is made up of binary (two-level) pseudo-random sequences of 511 and 63 symbols. The 12 symbols at the end of the reference signal are repeated from the previous active data segment to aid the trellis decoder in the receiver during cochannel interference when the NTSC rejection filter (12 symbol subtractive comb) is in the data path. Binary levels provide reliable detection for the Data Field sync under extremely adverse channel conditions.

The three concatenated 63 bit PRBS sequences in the field sync alternate polarity from one field to the next, providing field identification information to the receiver. Therefore, there are two data field syncs that make up one data frame (48.6 msec). The average value of the field sync symbols is close to zero so that when the DC pilot is added prior to transmission, they will not alter the desired value of pilot.

The Data Field Sync serves five purposes. First, it provides a means to determine the beginning of each data field. Second, it is also used by the equalizer in the ATV receiver as a training reference signal to remove intersymbol and other interferences. Third, it allows the receiver to determine whether the interference rejection filter (12 symbol subtractive comb) should be used. Fourth, it is used for system diagnostic measurements, such as signal-to-noise and channel response. Fifth, the phase tracker in the receiver uses the field sync to reset its circuitry and determine its loop parameters. Just like the data segment sync, the Data Field Sync is not Reed-Solomon or trellis encoded, nor is it interleaved.

2.2.6 PILOT INSERTION

A rugged system must be able to acquire a signal and maintain lock in the presence of very heavy noise and interference. A small pilot added to the suppressed carrier RF data signal is adequate to allow robust carrier recovery in the receiver during these extreme conditions.

A small (digital) DC level (1.25) is added to every symbol (data and syncs) of the digital baseband data plus sync signal (±1, ±3, ±5, ±7). This has the effect of adding a small in-phase pilot to the data signal. Digital addition of the pilot (at baseband) provides a highly stable and accurate pilot. The frequency of the pilot will be the same as the suppressed carrier frequency. Since the data is essentially guaranteed to be random over long intervals, all data states are equally probable. Using the above eight values for the random data symbols, the total average data power of the 8-level data signal is 21. After adding the pilot, the total average signal power is 22.56. Thus, the pilot represents an increased transmitted power of 0.3 dB. In the interference-limited environment, the pilot is not a contributing factor. Also, the pilot power has no significant effect on the ATV transmitter hardware (e.g. power dissipation or peak-to-average power ratio).

2.2.7 PRE-EQUALIZER FILTER

A pre-equalizer filter is available for use in over-the-air broadcasts, where the high power transmitter may have significant in-band ripple or roll off at band edges. This linear distortion can be detected by an equalizer in a reference demodulator ("ideal" receiver) located at the transmitter site that is receiving a small portion of the antenna signal feed (from a "sample"). The reference demodulator equalizer tap weights can be transferred into the transmitter pre-equalizer for pre-correction of transmitter linear distortion.

The pre-equalizer is an 80 tap, feedforward transversal filter. It operates on the I channel data signal (there is no Q channel data in the transmitter), and shapes the frequency spectrum of the IF signal so that it is a flat spectrum at the output of the high power transmitter that feeds the antenna for transmission.

2.2.8 VSB MODULATOR

The VSB modulator receives the 10.76 Msymbols/sec, 8-level trellis composite data signal (pilot plus syncs added). For minimal intersymbol interference, the data signal must be properly shaped (filtered) before it is transmitted over the 6 MHz channel. A linear phase raised-cosine Nyquist filter is employed in the concatenated transmitter and receiver, as shown in Fig 8. The system filter response is essentially flat across the entire band, except for the transition regions at each end of the band. Due to the vestigial-sideband nature of the transmitted signal, the same skirt selectivity on both sides is not required, although it is so implemented. The optimum system arrangement is to divide the roll off equally between the transmitter and receiver filters. Therefore, root-raised cosine filters are used. Fig 2 illustrates the ATV transmitter spectrum in comparison with a typical NTSC spectrum.

The transmitter VSB filtering is implemented by complex-filtering the baseband data signal, creating precision-filtered and stable in- phase and quadrature-phase modulation signals. This filtering process provides the root-raised cosine Nyquist filtering as well as the sin x/x compensation for the D/A converters. The orthogonal baseband signals are converted to analog form (D/A converters) and then modulated on quadrature IF carriers to create the vestigial sideband IF signal by sideband cancellation (phasing method). The nominal frequency of the IF carrier (and small in-phase pilot) is 46.69 MHz, which is equal to the IF center frequency (44.000 MHz) plus the data rate divided by 4 (10.762 MHz/4=2.6905 MHz). Additional adjacent channel suppression (beyond that achieved by sideband cancellation) is performed by a linear phase, flat amplitude response SAW filter. Adjacent channel energy spillage at the IF output is at least 57 dB down from the desired ATV signal power.

2.2.9 UPCONVERTER AND RF CARRIER FREQUENCY OFFSETS

Modern NTSC TV transmitters use a two-step modulation process. The first step usually is modulation of the data onto an IF carrier, which is the same frequency for all channels, followed by translation to the desired RF channel. The VSB transmitter applies this same two-step modulation process. The RF upconverter translates the filtered flat IF data signal spectrum to the desired RF channel. For the same coverage as an NTSC transmitter, the average power of the ATV signal will be at least 12 dB less than the NTSC peak sync power.

The frequency of the RF upconverter oscillator in ATV terrestrial broadcasts will typically be the same as that used for NTSC (except for NTSC offsets). However, in extreme cochannel situations, the ATV system is designed to take advantage of precise RF carrier frequency offsets with respect to the NTSC cochannel carrier. Since the VSB data signal sends repetitive synchronizing information (segment syncs), precise offset causes NTSC cochannel carrier interference into the VSB receiver to phase alternate from sync to sync. The VSB receiver circuits average successive syncs to cancel the interference and make data segment sync detection more reliable.

For ATV cochannel interference into NTSC, the interference is noise-like and does not change with precise offset. Even the ATV pilot interference into NTSC does not benefit from precise frequency offset because it is so small (11.3 dB below the data power) and falls far down the Nyquist slope (20 dB or more) of NTSC receivers.

Although it might be postulated that an ATV transmitter can be located so as to experience equal interference from two worst-case cochannel NTSC stations (e.g. three-way triangle), such a situation is so unlikely that the ATV signal is assumed to have only one dominant NTSC cochannel. The ATV cochannel pilot should be offset in the RF upconverter from the dominant NTSC picture carrier by an odd multiple half the data segment rate. A $56.9*f_H$ = 895.28 KHz offset is proposed between the pilot frequency and that of the cochannel NTSC picture carrier. In order to achieve this requirement, a 45.8 KHz spectrum shift of the VSB signal into the upper adjacent channel is required. This is illustrated (Fig 15) and further described later in the section on the receiver NTSC rejection filter.

For ATV to ATV cochannel interference, precise carrier offset prevents possible misconvergence of the adaptive equalizer. If perchance the two ATV Data Field Sync signals should fall within the same data segment time, the adaptive equalizer could misinterpret the interference as a ghost. To prevent this, a carrier offset of $f_{seg}/2$ = 6.437 KHz is proposed for close ATV-to-ATV cochannel situations. This causes the interference to have no effect in the adaptive equalizer.

2.3 RECEIVER BROADCAST MODE

Fig 9 shows the receiver block diagram of the VSB terrestrial broadcast transmission system. Descriptions of each block follow.

2.3.1 TUNER

The tuner, illustrated in Fig 10, receives the 6 MHz ATV signal from the antenna. It is a high-side injection double-conversion type with a first IF frequency of 920 MHz. This puts the image frequencies above 1 GHz, making them easy to reject by a fixed front end filter. This selection of first IF frequency is high enough so that the input bandpass filter selectivity prevents the local oscillator (978-1723 MHz) from leaking out the tuner front end and interfering with other UHF channels, yet it is low enough for second harmonics of UHF channels (470-806 MHz) to fall above the first IF bandpass. Harmonics of cable channels could possibly occur in the first IF passband but are not a real problem because of the relatively flat spectrum (within 10 dB) and small signal levels (-28 dBm or less) used in cable systems.

The tuner input has a bandpass filter that limits the frequency range to 50-810 MHz, rejecting all other non-television signals that may fall within the tuners image frequency range (beyond 920 MHz). In addition, a broadband tracking filter rejects other television signals, especially those much larger in signal power than the desired ATV signal power. This tracking filter is not narrow, nor is it critically tuned, as is the case of present day NTSC tuners that must reject image signals only 90 MHz away from the desired channel. Minimal channel tilt, if any, exists due to this tracking filter.

A 10 dB gain, wideband RF amplifier increases the signal level into the first mixer, and is the dominant determining factor of receiver noise figure (7-9 dB over entire VHF, UHF, and cable bands). The first mixer, a highly linear double-balanced design to minimize even harmonic generation, is driven by a synthesized low phase noise local oscillator (LO) above the first IF frequency (high-side injection). Both the channel tuning (first LO) and broadband tracking filtering (input bandpass filter) are controlled by microprocessor. The tuner is capable of tuning the entire VHF and UHF broadcast bands as well as all standard, IRC, and HRC cable bands.

The mixer is followed by an LC filter in tandem with a narrow 920 MHz bandpass ceramic resonator filter. The LC filter provides selectivity against the harmonic and subharmonic spurious responses of the ceramic resonators. The 920 MHz ceramic resonator bandpass filter has a -1 dB bandwidth of about 6 MHz. A 920 MHz IF amplifier is placed between the two filters. Delayed AGC of the first IF signal is applied immediately following the first LC filter. The 30 dB range AGC circuit protects the remaining active stages from large signal overload.

The second mixer is driven by the second LO, which is an 876 MHz voltage-controlled SAW oscillator. It is controlled by the frequency and phase-locked loop (FPLL) synchronous detector. The second mixer, whose output is the desired 44 MHz second IF frequency, drives a constant gain 44 MHz amplifier. The output of the tuner feeds the IF SAW filter and synchronous detection circuitry.

The tuner is made out of standard consumer electronic components, and is housed in a stamped metal enclosure.

2.3.2 CHANNEL FILTERING AND VSB CARRIER RECOVERY

Carrier recovery is performed on the small pilot carrier by an FPLL circuit, illustrated in Fig 11. The first LO is synthesized by a PLL and controlled by a microprocessor. The third LO is a fixed reference oscillator. Any frequency drift or deviation from nominal has to be compensated in the second LO. Control for the second LO comes from the FPLL synchronous detector, which integrally contains both a frequency loop and a phase-locked loop in one circuit. The frequency loop provides a wide frequency pull-in range of ±100 KHz while the phase-locked loop has a narrow bandwidth (less than 2 KHz).

During frequency acquisition, the frequency loop uses both the in-phase (I) and quadrature-phase (Q) pilot signals. All other data processing circuits in the receiver use only the I-channel signal. Prior to phase-lock, as is the condition after a channel change, the automatic frequency control (AFC) lowpass filter acts on the beat signal created by the frequency difference between the VCO and the incoming pilot. The high frequency data (as well as noise and interference) is mostly rejected by the AFC filter, leaving only the pilot beat frequency. After limiting this pilot beat signal to a constant amplitude (±1) square wave, and using it to multiply the quadrature signal, a traditional bipolar S-curve AFC characteristic is obtained. The polarity of the curve depends upon whether the VCO frequency is above or below the incoming IF signal. Filtered and integrated by the automatic phase control (APC) lowpass filter, this DC signal adjusts the tuner's second LO to reduce the frequency difference.

When the frequency difference comes close to zero, the APC loop takes over and phase-locks the incoming IF signal to the third LO. This is a normal phase-locked loop circuit, with the exception that it is bi-phase stable. However, the correct phase-lock polarity is determined by forcing the polarity of the pilot to be equal to the known transmitted positive polarity. Once locked, the detected pilot signal is constant, the limiter output feeding the third multiplier is at a constant +1, and only the phase-locked loop is active (frequency loop automatically disabled). The APC lowpass filter is wide enough to reliably allow ±100 KHz frequency pull-in, yet narrow enough to consistently reject all strong white noise (including data) and NTSC cochannel interference signals. The PLL has a bandwidth that is narrow enough to reject most of the AM and PM generated by the data, yet is wide enough to track out any phase noise on the signal (and, hence, on the pilot) out to about 2 KHz. Tracking out low frequency phase noise (as well as low frequency FM components) allows the phase tracking loop, discussed later, to be more effective.

The prototype receiver can acquire a signal and maintain lock at a signal-to-noise of 0 dB or less, and in the presence of heavy interference.

2.3.3 SEGMENT SYNC AND SYMBOL CLOCK RECOVERY

The repetitive data segment syncs are detected from among the synchronously detected random data by a narrow bandwidth filter. From the data segment syncs, a properly phased 10.76 MHz symbol clock is created along with a coherent AGC control signal. A block diagram of this circuit is shown in Fig 12.

The 10.76 Msymbols/sec I-channel composite baseband data signal (syncs and data) from the synchronous detector is converted by an A/D converter for digital processing. Traditional analog data eyes can be viewed after synchronous detection. However, after conversion to a digital signal, the data eyes cannot be seen due to the sampling process. A PLL is used to derive a clean 10.76 MHz symbol clock for the receiver.

With the PLL free-running, the data segment sync detector containing a 4-symbol sync correlator looks for the two level syncs occurring at the specified repetition rate. The repetitive segment sync is detected while the random data is not, enabling the PLL to lock on the sampled sync from the A/D converter, and achieve data symbol clock synchronization. Upon reaching a predefined level of confidence (using a confidence counter) that the segment sync has been found, subsequent receiver loops are enabled.

Data segment sync detection and clock recovery both work reliably at signal-to-noise ratios of 0 dB or less, and in the presence of heavy interference.

2.3.4 NON-COHERENT AND COHERENT AGC

Prior to carrier and clock synchronization, non-coherent automatic gain control (AGC) is performed whenever any signal (locked or unlocked signal, or noise/interference) overruns the A/D converter. The IF and RF gains are reduced accordingly, with the appropriate AGC "delay" applied.

When data segment syncs are detected, coherent AGC occurs using the measured segment sync amplitudes. The amplitude of the bipolar syncs, relative to the discrete levels of the random data, is determined in the transmitter. Once the syncs are detected in the receiver, they are compared to a reference value, with the difference (error) integrated. The integrator output then controls the IF and "delayed" RF gains, forcing them to whatever values provide the correct sync amplitudes.

2.3.5 DATA FIELD SYNCHRONIZATION

Data Field Sync detection, shown in Fig 14, is achieved by comparing each received data segment from the A/D converter (after interference rejection filtering to minimize cochannel interference), to ideal field #1 and field #2 reference signals in the receiver. Oversampling of the field sync is NOT necessary since a precision data segment and symbol clock has already been reliably created by the clock recovery circuit. Therefore, the field sync recovery circuit knows exactly where a valid field sync correlation should occur within each data segment, and only needs to perform a symbol by symbol difference. Upon reaching a predetermined level of confidence (using a confidence counter) that field syncs have been detected on given data segments, the Data Field Sync signal becomes available for use by subsequent circuits. The polarity of the three alternating 63 bit pseudo random (PN) sequences determine whether field 1 or field 2 is detected.

This procedure makes field sync detection robust, even in heavy noise, interference, or ghost conditions. Field sync recovery can reliably occur at signal-to-noise ratios of 0 dB or less, and in the presence of heavy interference.

2.3.6 INTERFERENCE REJECTION FILTER

The interference rejection properties of the VSB transmission system are based on the frequency location of the principal components of the NTSC cochannel interfering signal within the 6 MHz TV channel and the periodic nulls of a VSB receiver baseband comb filter.

Fig 15a shows the location and approximate magnitude of the three principal NTSC components: (1) the visual carrier (V) located 1.25 MHz from the lower band edge, (2) the chrominance subcarrier (C) located 3.58 MHz higher than the visual carrier frequency, and (3) the aural carrier (A) located 4.5 MHz higher than the visual carrier frequency.

The NTSC interference rejection filter (comb) is a one tap linear feed-forward filter, as shown in Fig 16. Fig 15b shows the frequency response of the comb filter, which provides periodic spectral nulls spaced 57* f_h (10.762 MHz/12, or 896.85 KHz) apart. There are 7 nulls within the 6 MHz channel. The NTSC visual carrier frequency falls close to the second null from the lower band edge. The 6th null from the lower band edge is correctly placed for the NTSC chrominance subcarrier, and the 7th null from the lower band edge is near the NTSC aural carrier.

Comparing Fig 15a and Fig 15b shows that the visual carrier falls 2.1 kHz below the second comb filter null, the chroma subcarrier falls exactly at the 6th null, and the aural carrier falls into the 7th null. (Note, the aural carrier is at least 7 dB below its visual carrier).

The comb filter, while providing rejection of steady-state signals located at the null frequencies, has a finite response time of 12 symbols (1.115 μsecs). Thus, if the NTSC interfering signal has a sudden step in carrier level (low to high or high to low), one cycle of the beat frequency (offset) between the ATV and NTSC carrier frequencies will pass through the comb filter at an amplitude proportional to the NTSC step size as instantaneous interference. Examples of such steps of NTSC carrier are: leading and trailing edge of sync (40 IRE units). If the desired to undesired (D/U) signal power ratio is large enough, data slicing errors will occur. However, interleaving will spread the interference and will make it easier for the Reed-Solomon code to correct them (R-S can correct up to 10 byte errors/segment).

Although the comb filter reduces the NTSC interference, the data is also modified. The 7 data eyes (8 levels) are converted to 14 data eyes (15 levels). Such a process is called partial response. The modified data signal can be properly decoded by the trellis decoder, and will be described in later sections. Note, because of time sampling, only the maximum data eye value is seen after A/D conversion.

The detail at the band edges for the overall channel is shown in Fig 15c and Fig 15d. Fig 15d shows that the frequency relationship ($56\frac{19}{22} * f_H$ between NTSC visual carrier and ATV carrier) requires a shift in the ATV spectrum with respect to the nominal channel. The shift equals +45.8 KHz, or about +0.76%. This is slightly higher than currently applied channel offsets and reaches into the upper adjacent channel at a level of about -40 dB. If that is another ATV channel, the nominal situation is restored. If it is an NTSC channel, the shift is below the (RF equivalent of the) Nyquist slope of an NTSC receiver where there is high attenuation, and it is slightly above its customary lower adjacent channel sound trap. No adverse effects of the shift have been found nor are they foreseen.

NTSC interference can be detected by the circuit shown in Fig 16, where the signal-to-interference plus noise ratio of the binary Data Field Sync is measured at the input and output of the comb filter, and compared to each other. This is accomplished by creating an error signal at each rejection filter port by comparing the received signal with a stored reference of the field sync. The errors are squared and integrated. After a predetermined level of confidence is achieved, the path with the largest signal-to-noise ratio (lowest interference energy) is switched in and out of the system automatically.

There is a reason to not leave the rejection comb filter switched in all the time. The comb filter, while providing needed cochannel interference benefits, degrades white noise performance by 3 dB. This is due to the fact that the filter output is the subtraction of two full gain paths, and since white noise is uncorrelated from symbol to symbol, the noise power doubles. If little or no NTSC interference is present, the comb filter is automatically switched out of the data path.

2.3.7 CHANNEL EQUALIZER

The equalizer/ghost canceler is used to compensate for linear channel distortions, such as tilt and ghosts. The distortions can come from the transmission channel or from imperfect components within the receiver. The equalizer delivered for testing uses a Least-Mean-Square (LMS) algorithm and adapts on the transmitted binary Data Field Sync as well as on the random data. A block diagram of the equalizer used for the G-A "bakeoff" testing is shown in Fig 17.

The advantage of adapting on a known training signal embedded within the random data signal is the guarantee of tap convergence, especially in extreme noise, interference, and ghost conditions. All the various loops within the receiver (carrier, clock, AGC, segment sync, frame sync) are independent of subsequent loops, including the equalizer. An optimum method is to use a two step process: equalization first on a binary training sequence to open the data eyes, and then equalization on the random data for high speed tracking of moving ghosts (e.g. airplane flutter).

The equalizer filter consists of two parts, a 78-tap feedforward transversal filter followed by a 177-tap decision-feedback section. The long decision-feedback provides more accurate ghost canceling due to reduced noise enhancement than that found in long feedforward equalizers. The equalizer operates at the 10.762 MHz symbol rate (T-sampled equalizer). Symbol rate sampling is made possible because the symbol clock has been accurately phased by the clock recovery circuitry BEFORE the equalizer. To aid convergence, the equalizer also includes an extra adder which is used to add or subtract a DC value to compensate for DC errors which can be caused by circuit offsets, nonlinearities, or shifts in the pilot due to ghosts.

For the final G-A system, the G-A intends to augment the equalizer with a blind equalization algorithm to improve airplane flutter performance.

2.3.8 PHASE TRACKING LOOP

The phase tracking loop is an additional decision feedback loop which further tracks out phase noise which has not been removed by the IF PLL operating on the pilot. Thus, phase noise is tracked out by not just one loop, but two concatenated loops. Because the system is already frequency-locked to the pilot by the IF PLL (independent of the data), the phase tracking loop bandwidth is maximized for phase tracking by using a first order loop. Higher order loops, which are needed for frequency tracking, do not perform phase tracking as well as first order loops. Therefore, they are not used in the VSB system.

A block diagram of the phase tracking loop is shown in Fig 18. The output of the real equalizer operating on the I signal is first gain controlled by a multiplier and then fed into a filter which recreates an approximation of the Q signal. This is possible because of the VSB transmission method, where the I and Q components are related by a filter function which is almost a Hilbert transform. The complexity of this filter is minor since it is a finite impulse response (FIR) filter with fixed anti-symmetric coefficients and with every other coefficient equal to zero. In addition, many filter coefficients are related by powers of two, thus simplifying the hardware design.

These I and Q signals are then fed into a de-rotator (complex multiplier), which is used to remove the phase noise. The amount of de-rotation is controlled by decision feedback of the data taken from the output of the de-rotator. Since the phase tracker is operating on the 10.76 Msymbol/sec data, the bandwidth of the phase tracking loop is fairly large, approximately 60 KHz. The gain multiplier is also controlled with decision feedback.

2.3.9 TRELLIS DECODER

To help protect the trellis decoder against short burst interference, such as impulse noise or NTSC cochannel interference, 12 symbol code interleaving is employed in the transmitter. As shown in Fig 19, the receiver uses 12 trellis decoders in parallel, where each trellis decoder sees every 12th symbol. This code interleaving has all the same burst noise benefits of a 12 symbol interleaver, but also minimizes the resulting code expansion (and hardware) when the NTSC rejection comb filter is active.

The trellis decoder performs the task of slicing and convolutional decoding. It has two modes; one when the NTSC rejection filter is used to minimize NTSC cochannel, and the other when it is not used. This is illustrated in Fig 20. The insertion of the NTSC rejection filter is determined automatically (before the equalizer), with this information passed to the trellis decoder. When there is little or no NTSC cochannel interference, the NTSC rejection filter is not used, and an optimal trellis decoder is used to decode the 4-state trellis-encoded data.

In the presence of significant NTSC cochannel interference, when the NTSC rejection filter (12 symbol, feedforward subtractive comb) is employed, a trellis decoder optimized for this partial response channel is used. This optimal code requires 8 states. This is necessary since the NTSC rejection filter, which has memory, represents another state machine seen at the input of the trellis decoder. In order to minimize the expansion of trellis states, two measures are taken: (1) special design of the trellis code, and (2) twelve-to-one interleaving of the trellis encoding. The interleaving, which corresponds exactly to the 12 symbol delay in the NTSC rejection filter, makes it so that each trellis decoder only sees a one-symbol delay NTSC rejection filter. By minimizing the delay stages seen by each trellis decoder, the expansion of states is also minimized. Only a 3.5 dB penalty in white noise performance is paid as the price for having good NTSC cochannel performance. The additional 0.5 dB beyond the 3 dB comb filter noise threshold degradation is due to the 12 symbol differential coding.

It should be noted that after the ATV transition period and NTSC is no longer being transmitted, the NTSC rejection filter and the 8-state trellis decoder can be eliminated from receivers.

Trellis decoder complexity is exponentially proportional to the number of states. The 4-state and 8-state decoders used in the VSB terrestrial broadcast mode are very simple and cost effective, yet result in about 1.50 dB improvement in signal-to-noise ratio over the 4-VSB system. Increasingly complex trellis codes, which provide small amounts of theoretical improvement in coding gain, will require increased interleaving complexity, and are also more susceptible to burst errors and implementation loss.

2.3.10 DATA DE-INTERLEAVER

The de-interleaver performs the exact inverse function of the transmitter interleaver. Its 1/3 data field depth, and intersegment "dispersion" properties allow noise bursts lasting about 170 μsecs to be handled. Even strong NTSC cochannel signals passing through the NTSC rejection filter and creating short bursts due to NTSC vertical edges, are reliably handled due to the interleaving and R-S coding process. The intrasegment interleaving is also performed, reassembling the trellis data from the 12 trellis decoders into their original form.

2.3.11 REED-SOLOMON DECODER

The trellis-decoded byte data is sent to the (208,188) t=10 R-S decoder, where it uses the 20 parity bytes to perform the byte-error correction on a segment-by-segment basis. Up to 10-byte errors/data segment are corrected by the R-S decoder. Any burst errors created by impulse noise, NTSC cochannel interference, or trellis-decoding errors, are greatly reduced by the combination of the interleaving and R-S error correction.

2.3.12 DATA DE-RANDOMIZER

The de-randomizer accepts the error-corrected data bytes from the R-S decoder, and applies the same PRS randomizing code to the data. The PRS code is generated identically as in the transmitter, using the same PRS generator feedback and output taps. Since the PRS is locked to the reliably recovered Data Field Sync (and not some codeword embedded within the potentially noisy data), it is exactly synchronized with the data, and performs reliably.

2.3.13 RECEIVER LOOP ACQUISITION SEQUENCING

The receiver incorporates a "universal reset" which initiates a number of "confidence counters" and "confidence flags" involved in the lock-up process. A universal reset occurs, for example, when tuning to another station or turning on the receiver.

The various loops within the VSB receiver acquire and lock-up sequentially, with "earlier" loops being independent from "later" loops. The order of loop acquisition is as follows:

* Tuner 1st LO synthesizer acquisition
* Non-coherent AGC reduces unlocked signal to within A/D range
* Carrier acquisition (FPLL)
* Data segment sync and clock acquisition
* Coherent AGC of signal (IF and RF gains properly set)
* Data field sync acquisition
* NTSC rejection filter insertion decision made
* Equalizer completes tap adjustment algorithm
* Trellis and R-S data decoding begin

Most of the loops mentioned above have confidence counters associated with them to insure proper operation. However, the build-up or let-down of confidence is not designed to be equal. The confidence counters buildup confidence quickly for quick acquisition times, but lose confidence slowly to maintain operation in noisy environments. The VSB receiver will work in S/N conditions of 0 dB or less as well as in severe interference situations.

3.0 HIGH SPEED CABLE MODE SYSTEM DESCRIPTION

The high data rate cable mode trades off transmission robustness (28.3 dB signal-to-noise threshold) for system data rate (43 Mbit/sec). Most parts of the cable mode VSB system are identical or similar to the terrestrial system. A pilot, data segment sync, and data field sync are all used to provide robust operation. The pilot in the cable mode also adds 0.3 dB to the data power. The symbol, segment, and field signals and rates are all the same, allowing either receiver to lock up on the other's transmitted signal. Also, the data frame definitions are identical. The primary difference is the number of transmitted levels (8 versus 16) and the use of trellis coding and NTSC interference rejection filtering in the terrestrial system.

The RF spectrum of the cable modem transmitter looks identical to the terrestrial system, as illustrated in Fig 21. Peak-to-average power ratio, shown in Fig 22, is very similar between the two VSB systems. The 16-VSB has slightly higher peak-to-average power ratio due to its 16-level random data. The error probability of 16-VSB, shown in Fig 23, is about 28.3 dB (3×10^{-6}), with forward error correction provided by Reed-Solomon. Fig 24 illustrates a typical data segment, where the number of data levels is seen to be 16 due to the doubled data rate.

Fig 25 shows the block diagram of the transmitter. It is identical to the terrestrial VSB system except the trellis coding is replaced with a Mapper which converts data to multi-level symbols. This conversion is performed by the trellis coding in the terrestrial VSB system. The receiver, shown in Fig 26, is identical to the VSB terrestrial receiver, except that the trellis decoder is replaced by a slicer, which translates the multi-level symbols into data. Also note that no NTSC interference rejection filter is required in the cable mode since strong cochannel signals are not present on cable. The lack of trellis coding and NTSC cochannel interference rejection results in a cost savings in the cable modem.

4.0 SUMMARY

The VSB transmission system provides high performance and low cost in both the terrestrial and cable modes. The terrestrial mode combines R-S and trellis coding with an NTSC rejection filter for maximum coverage area during and after the ATV transition period. Both the terrestrial and cable modes make use of a pilot, segment sync, and training signal for virtually no implementation loss. This means that theoretical performance is possible today without relying on future improvements. This performance is also achieved with minimal hardware complexity.

Factors contributing to low cost receivers are:

Simple Carrier Recovery (FPLL) using small pilot
Standard technology IF SAW filters
High immunity to phase noise (phase tracker) for inexpensive tuners
Symbol-spaced sampling for all data processing circuits
Single A/D converter (I channel only)
Simple Clock Recovery using data segment syncs
Real T-sampled Adaptive Equalizer operating at 10.76 MHz

5.0 FUTURE CONSIDERATIONS

During the first round of ATV testing, the 4-VSB system demonstrated a bi-rate capability whereby some data was transmitted more robustly in exchange for a lower net data rate. As delivered for G-A testing, the VSB systems will be configured to provide all the data at equal robustness. Bi-rate data can be a future option if considered important by the Advisory Committee. Bi-rate transmission in the trellis-coded VSB transmission system will have the same data and robustness tradeoffs as in the 4-VSB system already evaluated by the advisory committee.

The cable mode has the capability of being flexible in that the data rate can be traded for a lower white noise threshold, all under headend control. A given 6 MHz RF channel on the cable can be set to have a lower data rate by sending fewer bits/symbol, thus gaining white noise performance. The receiver knows what multi-level data is being transmitted by reading a binary code (always transmitted 2-level during the data field sync). This provides a cable operator with system flexibility as channels are added to the cable system.

6.0 REFERENCES

[1] Zenith [September 1991] Digital Spectrum Compatible HDTV: Technical Details. Monograph published by Zenith Electronics Corp & AT&T Bell Laboratories. FCC ACATS Document SS/WP1-0193.

[2] AT&T and Zenith Electronics Corporation, Oct. 26, 1992, DSC-HDTV System Improvements, submitted to Technical Sub-Group of the ATV Special Panel on Proposed ATV System Improvements.

[3] Certification Presentation, Systems Subcommittee, Working Party 1, Washington, D.C., November 6, 1991.

[4] VSB Transmission System Technical Details, Dec. 17, 1993, by Zenith Electronics Corp., submitted to Transmission Expert Group of the Technical Sub-Group of the ATV Special Panel.

TABLE 1 - VSB Parameters

Parameter	**Terrestrial Mode**	**High Data Rate Cable Mode**
Channel Bandwidth	6 MHz	6 MHz
Excess Bandwidth	11.5%	11.5%
Symbol Rate	10.76 MSPS	10.76 MSPS
Bits per Symbol	3	4
Trellis FEC	2/3 rate	None
Reed-Solomon FEC	T=10 (208,188)	T=10 (208,188)
Segment Length	836 Symbols	836 Symbols
Segment Sync	4 symbols per segment	4 symbols per segment
Frame Sync	1 per 313 segments	1 per 313 segments
Payload Data Rate	19.3 Mb/s	38.6 Mb/s
NTSC Co-Channel Rejection	NTSC Rejection Filter in receiver	N/A
Pilot Power Contribution	0.3 dB	0.3 dB
C/N Threshold	14.9 dB	28.3 dB

Figure 1

VSB Data Frame

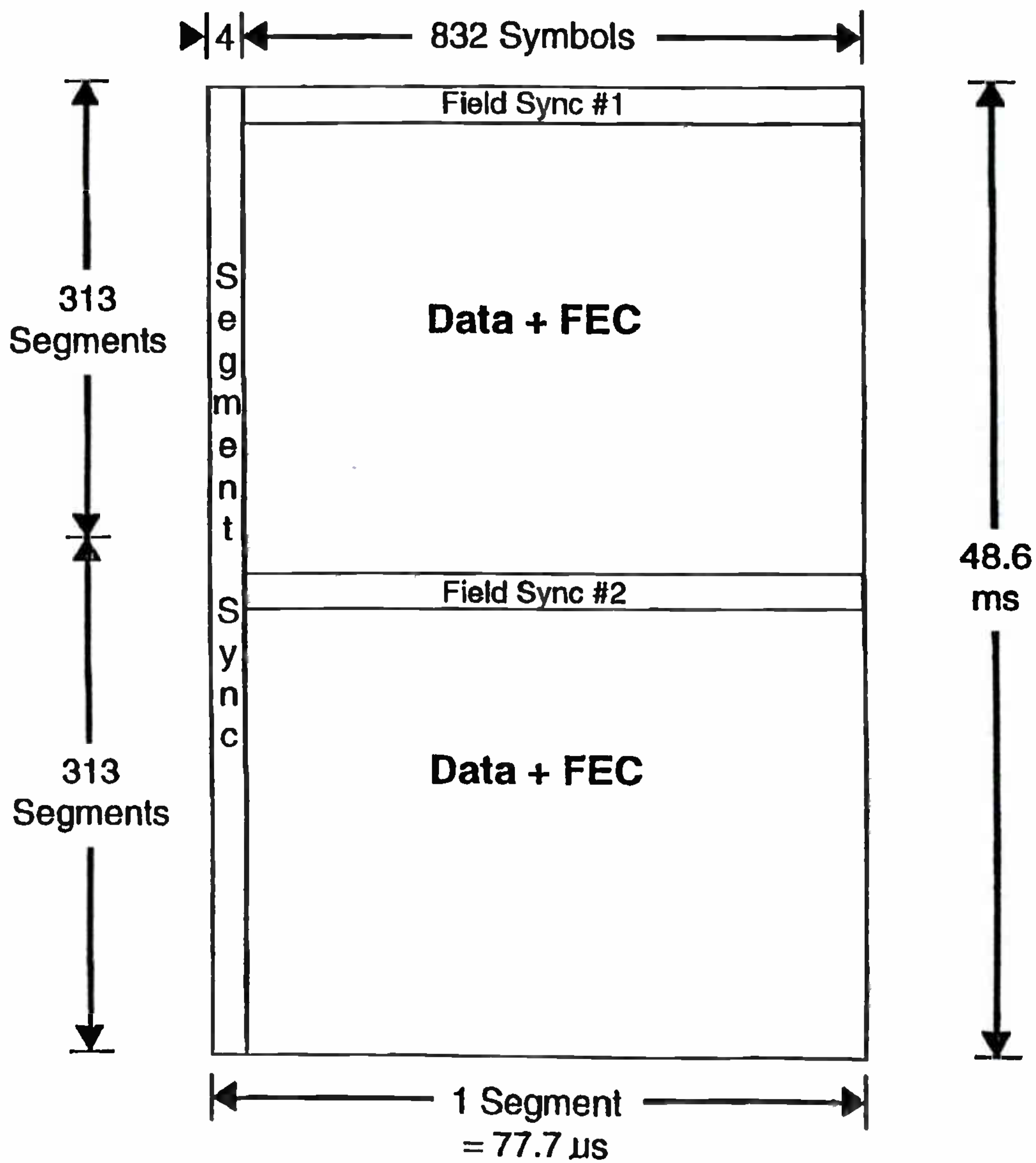


Figure 2

VSB AND NTSC CHANNEL OCCUPANCY

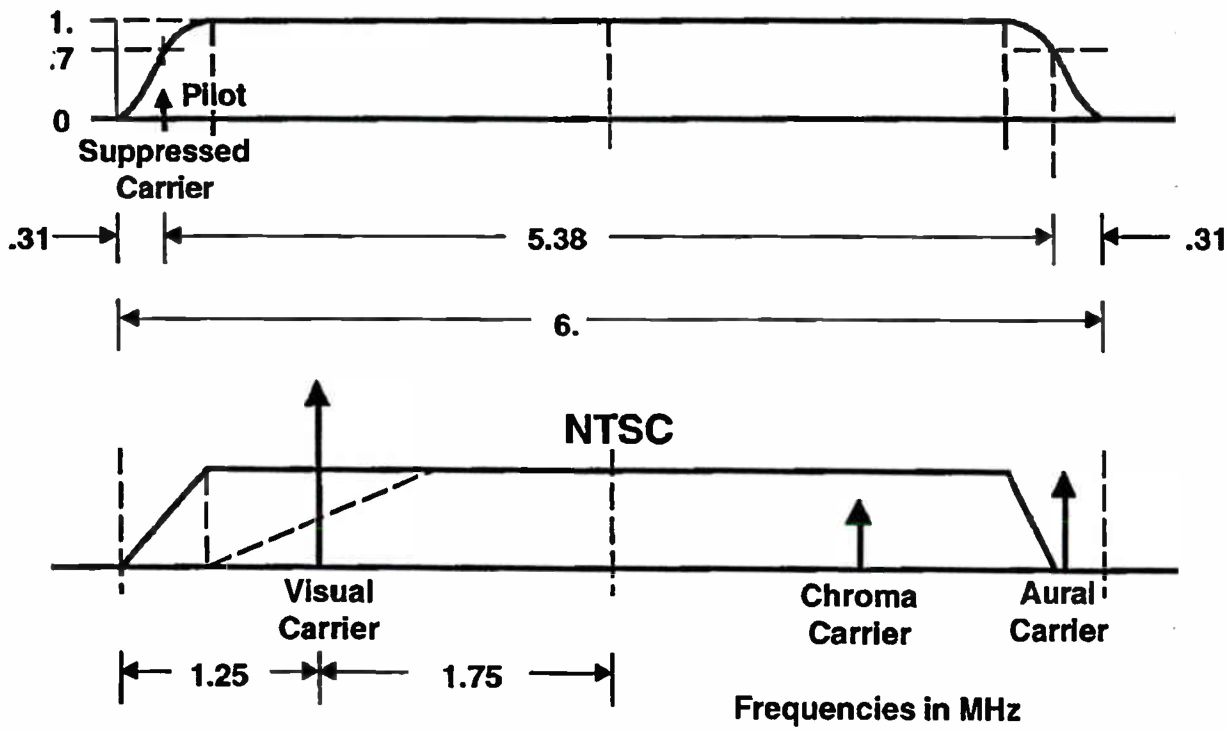


Figure 3

CUMULATIVE DISTRIBUTION FUNCTION OF 8-VSB PEAK-TO-AVERAGE POWER RATIO

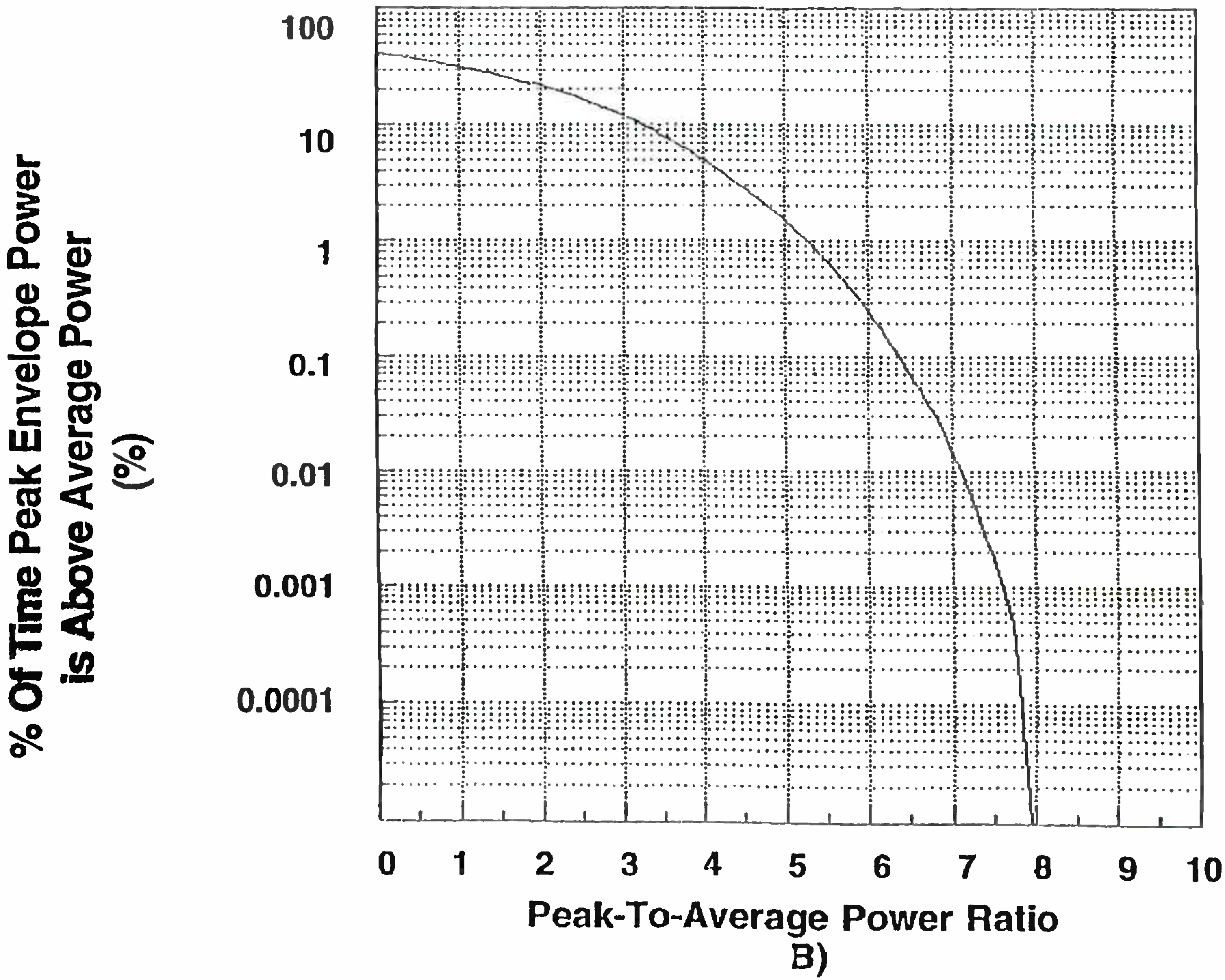


Figure 4

SEGMENT ERROR PROBABILITY
8vsb w/4state trellis; RS=(208,188)

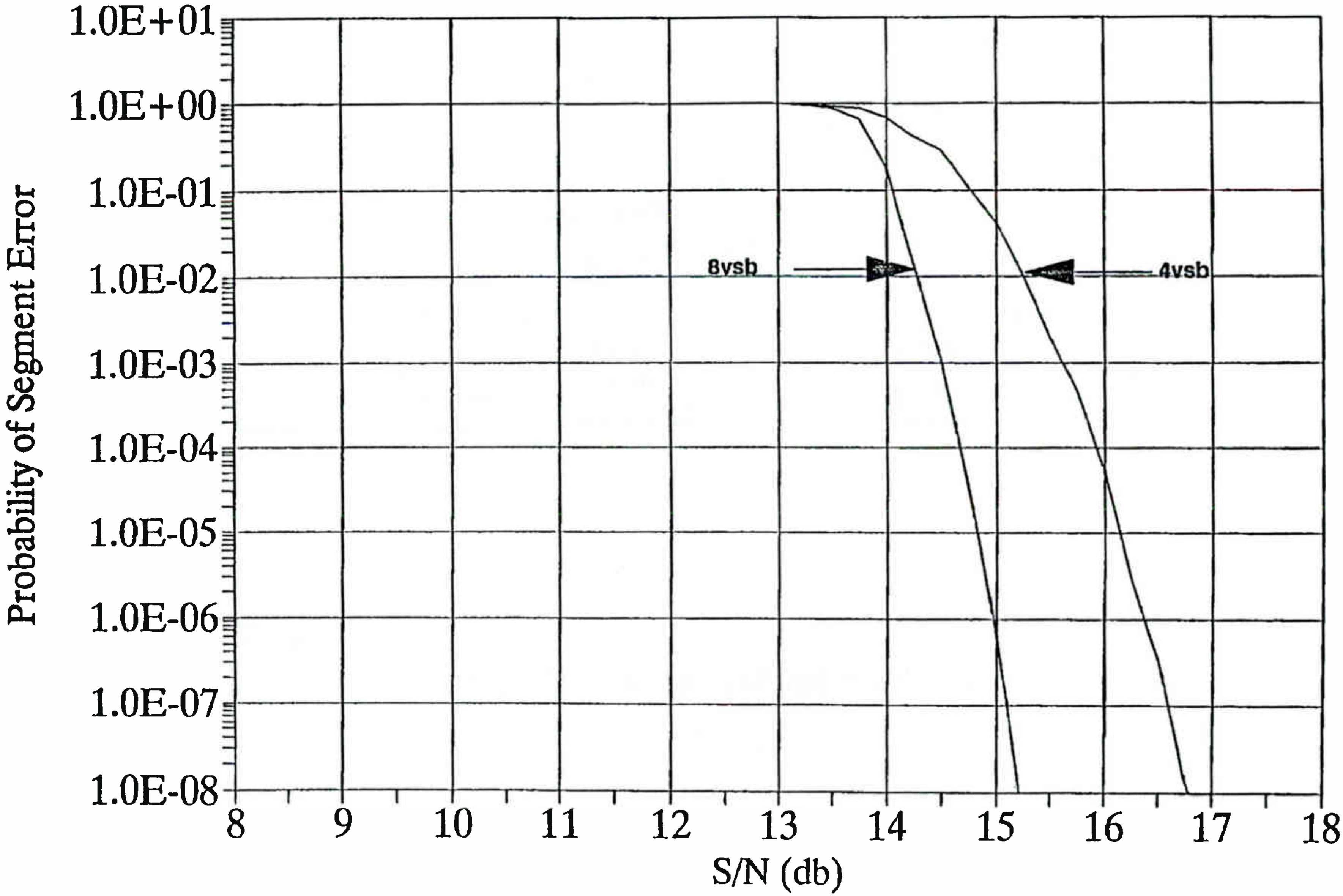


Figure 5

VSB TRANSMITTER

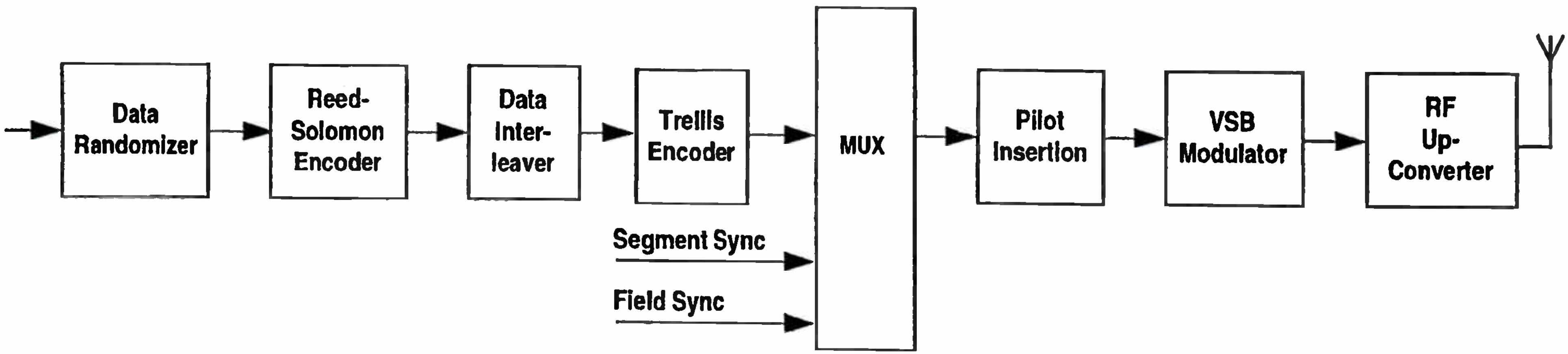


Figure 6

VSB DATA SEGMENT (Terrestrial)

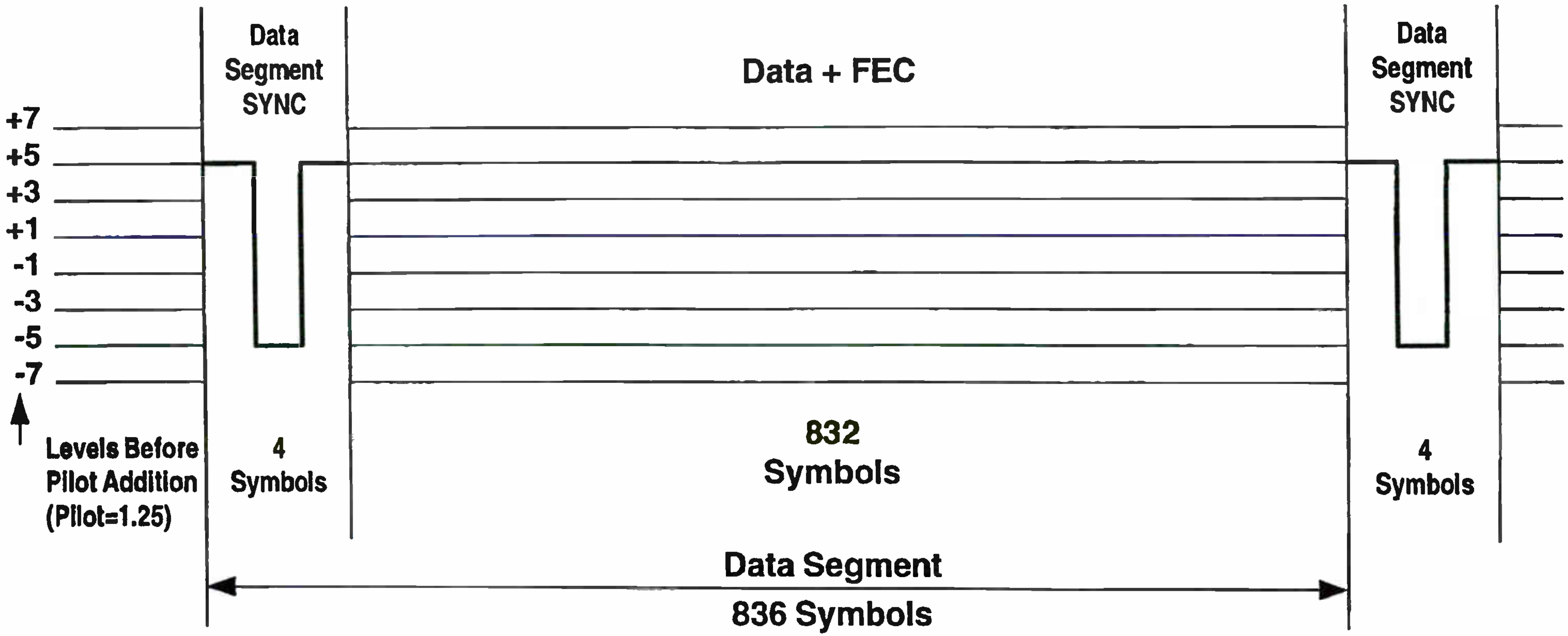
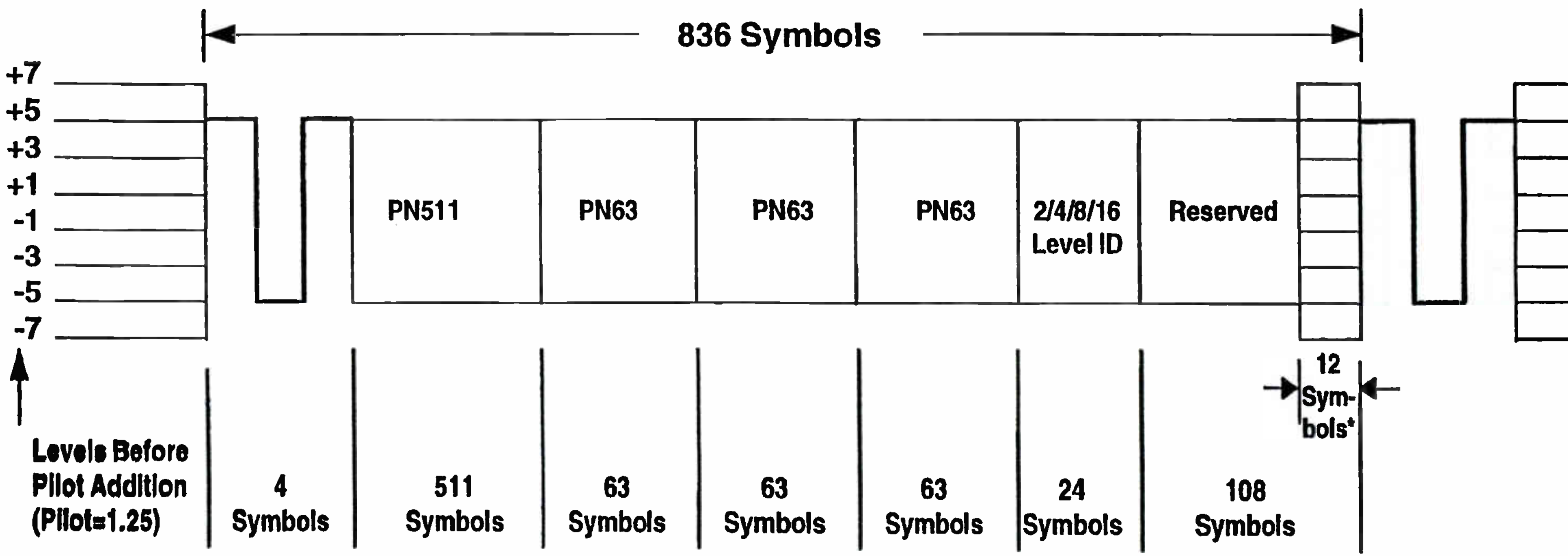


Figure 7

VSB DATA FIELD SYNC



* For 8-VSB the last 12 symbols of the previous segment are duplicated in the last 12 reserved symbols of the field sync.

Figure 8

NOMINAL VSB SYSTEM CHANNEL RESPONSE
(Linear Phase Raised Cosine Nyquist Filter)

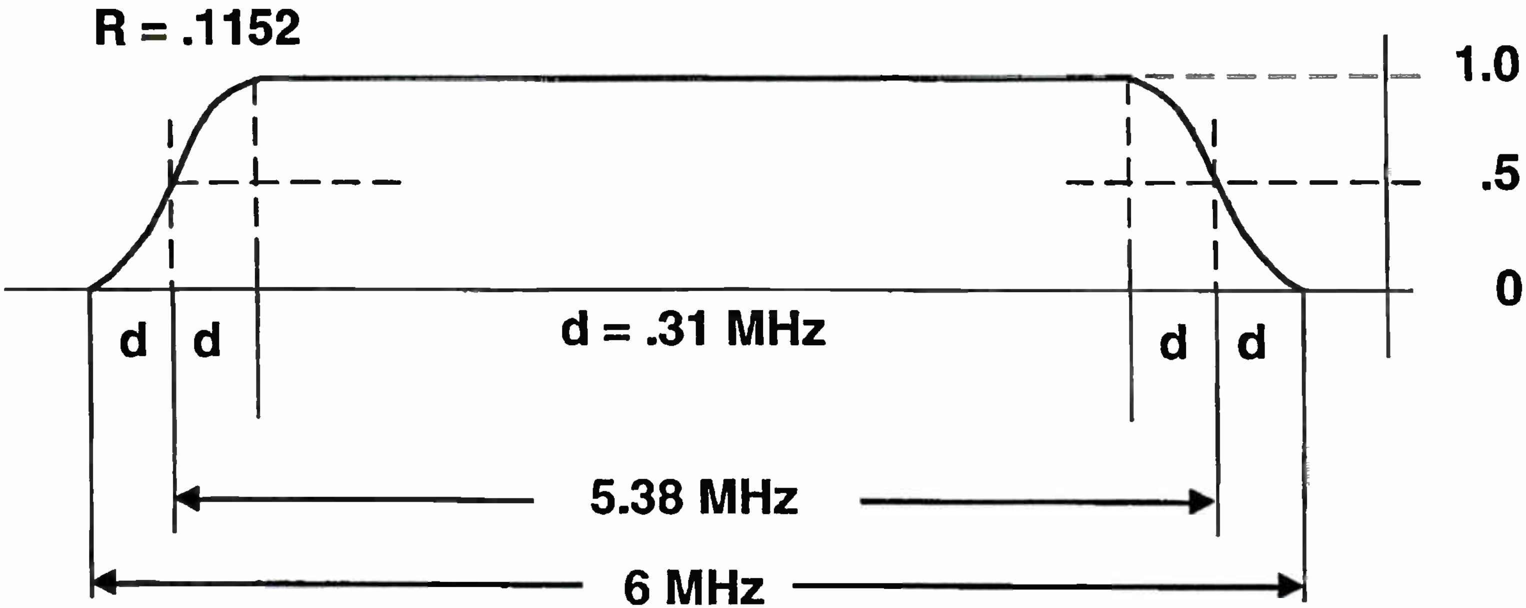


Figure 9

VSB RECEIVER

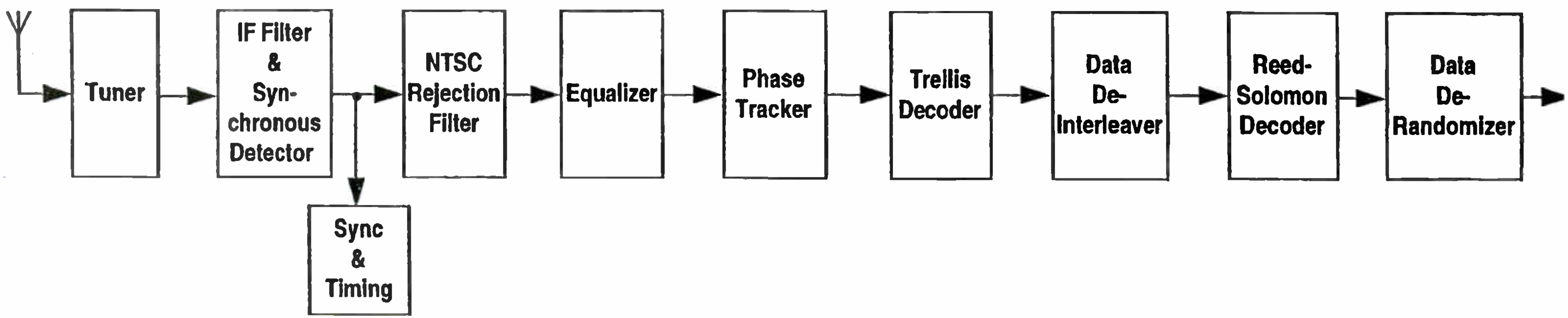


Figure 10

TUNER BLOCK DIAGRAM

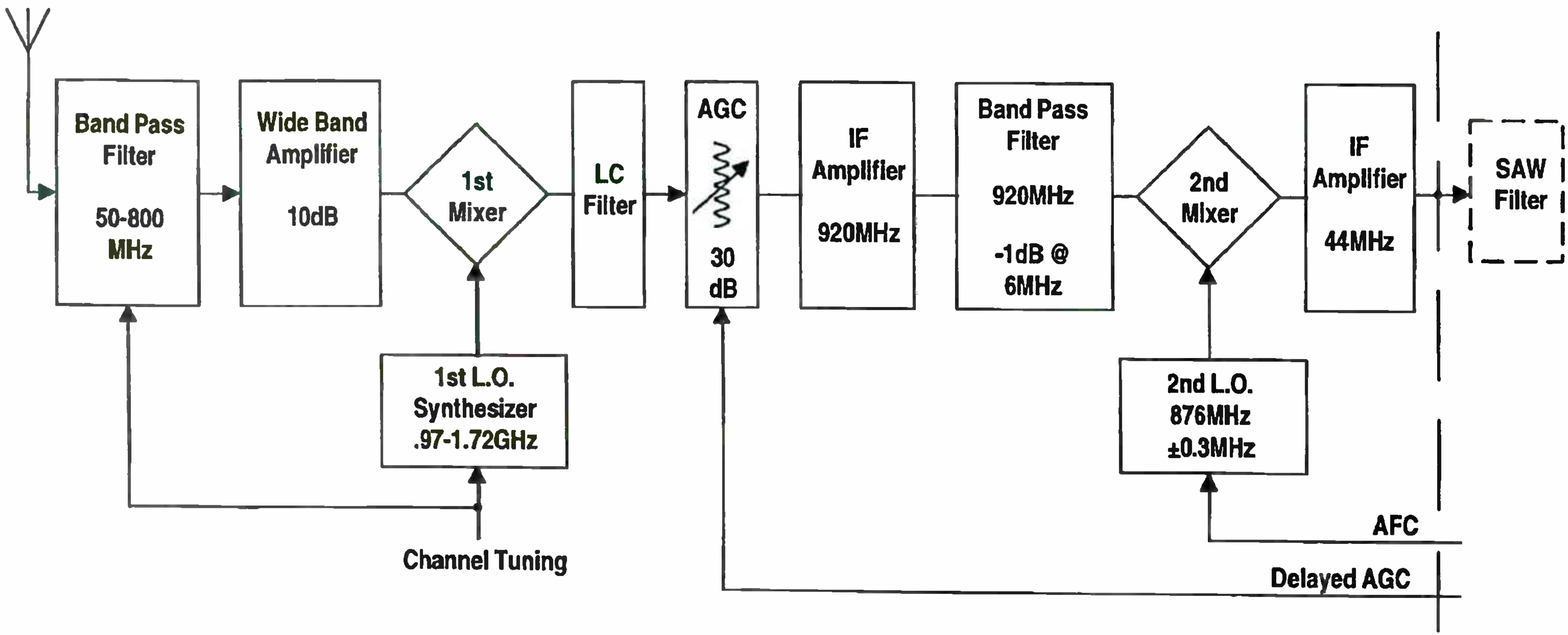


Figure 11

TUNER- IF - FPLL

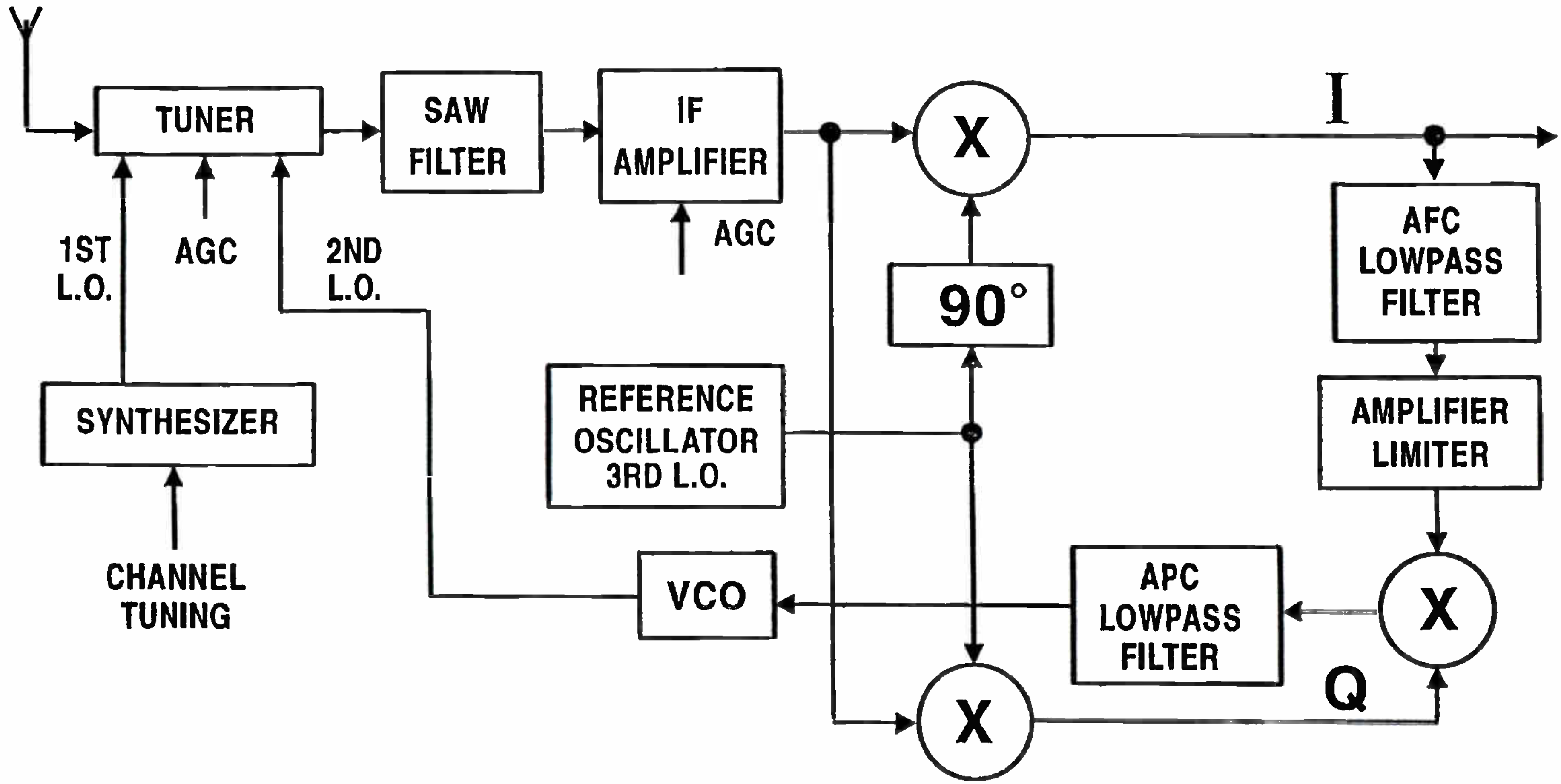


Figure 12

SEGMENT SYNC + SYMBOL CLOCK RECOVERY WITH AGC GENERATION

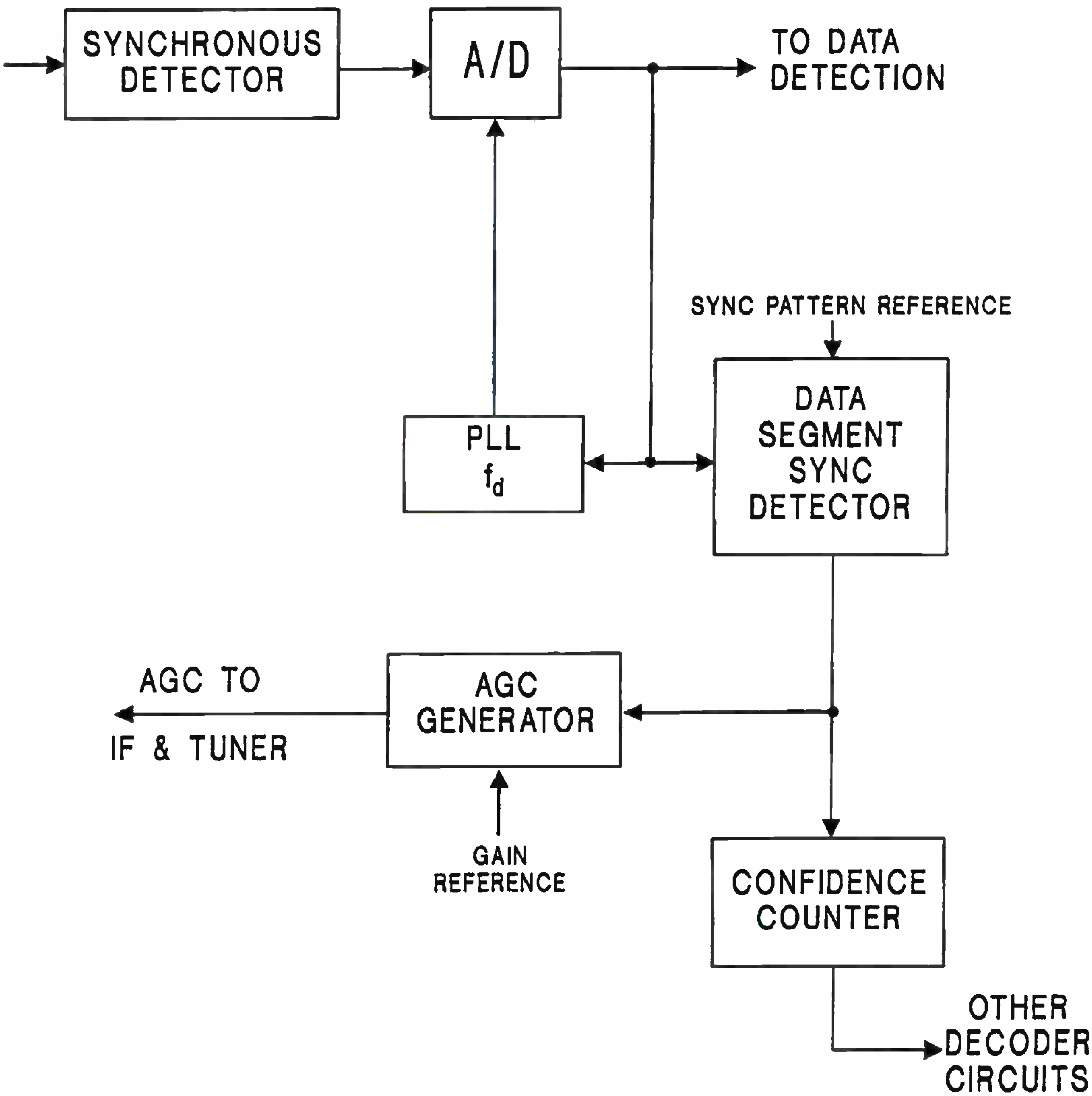


Figure 13

DATA SEGMENT SYNC

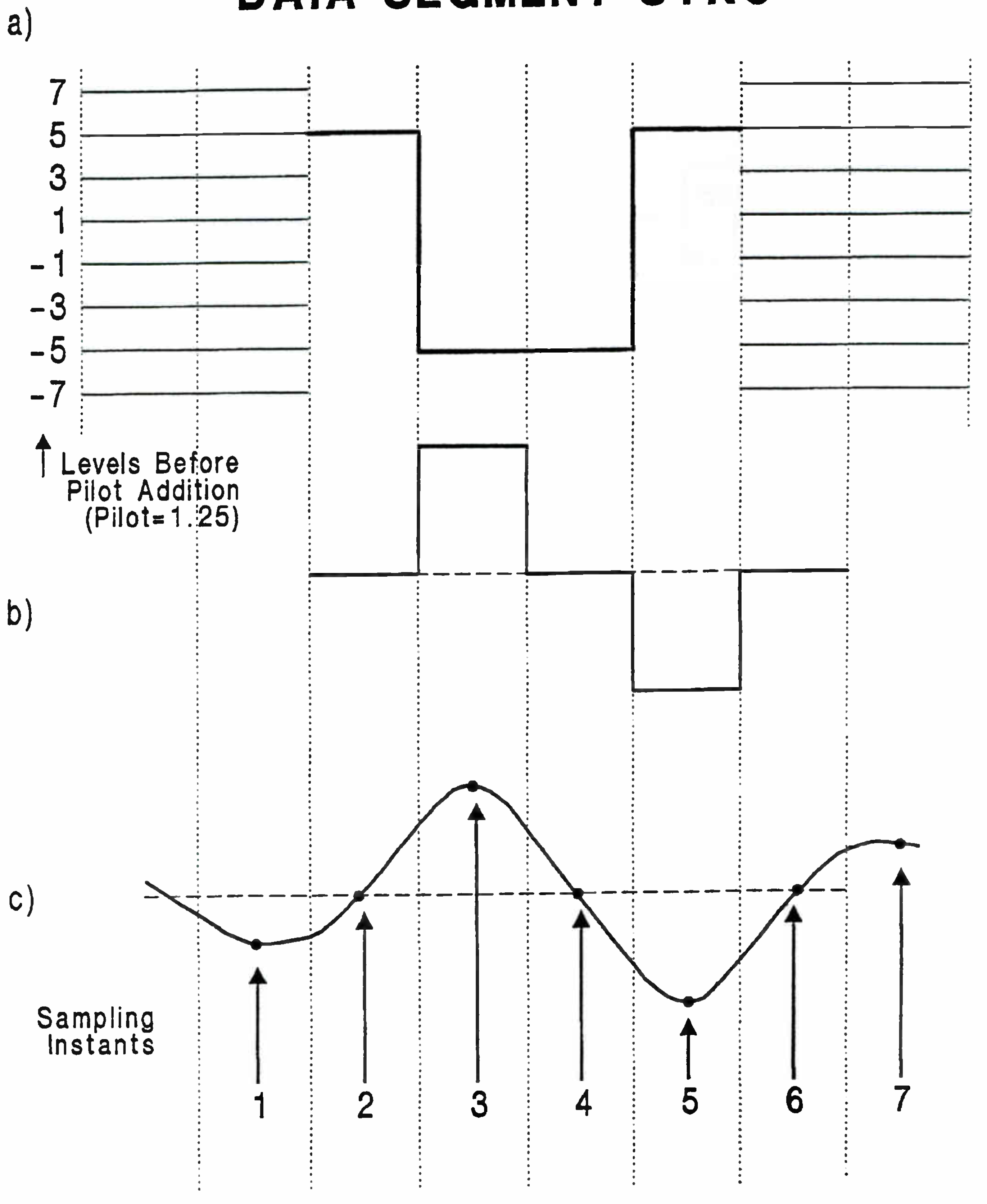


Figure 14

DATA FIELD SYNC RECOVERY

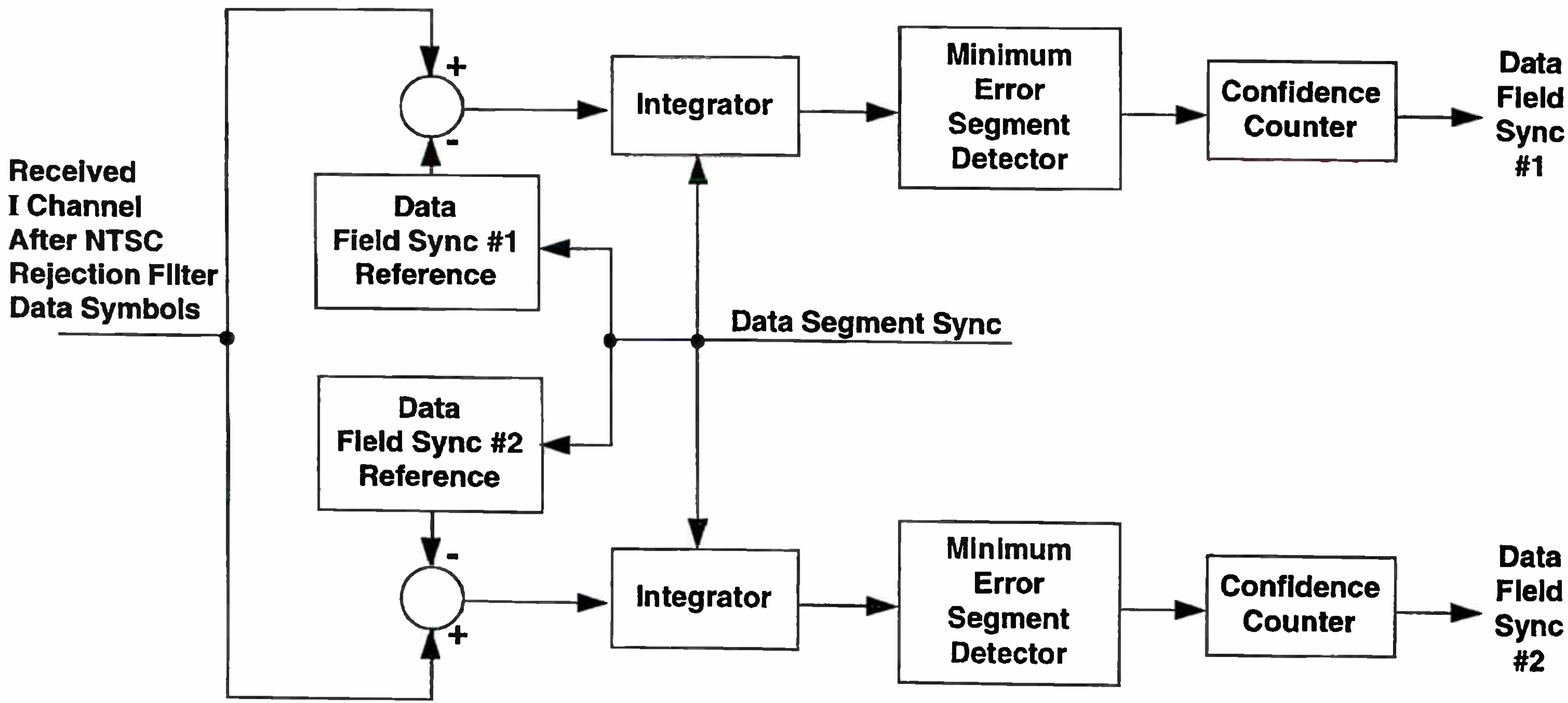


Figure 15

NTSC INTERFERENCE REJECTION FILTER SPECTRUM OCCUPANCY

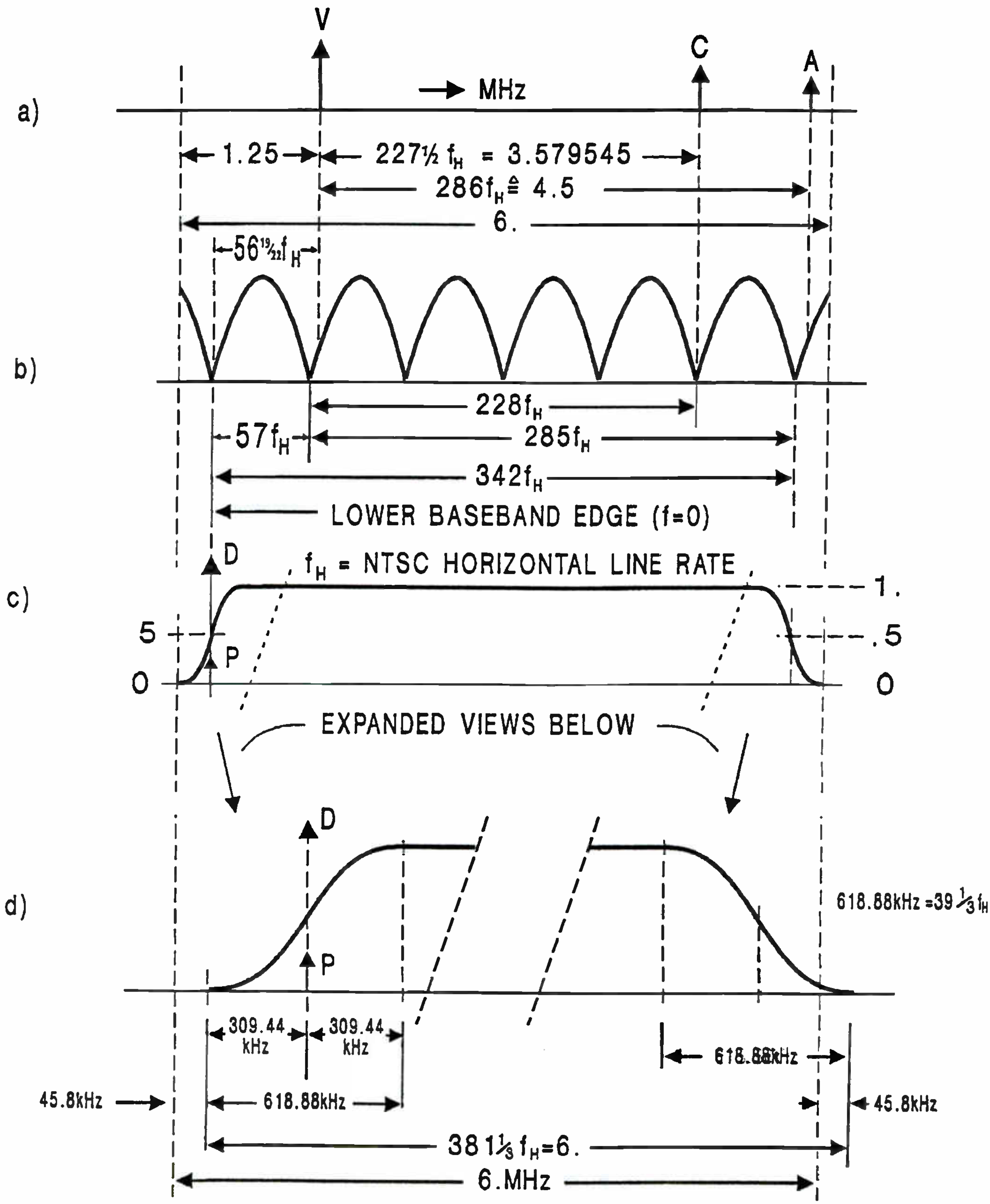


Figure 16

NTSC INTERFERENCE DETECTION

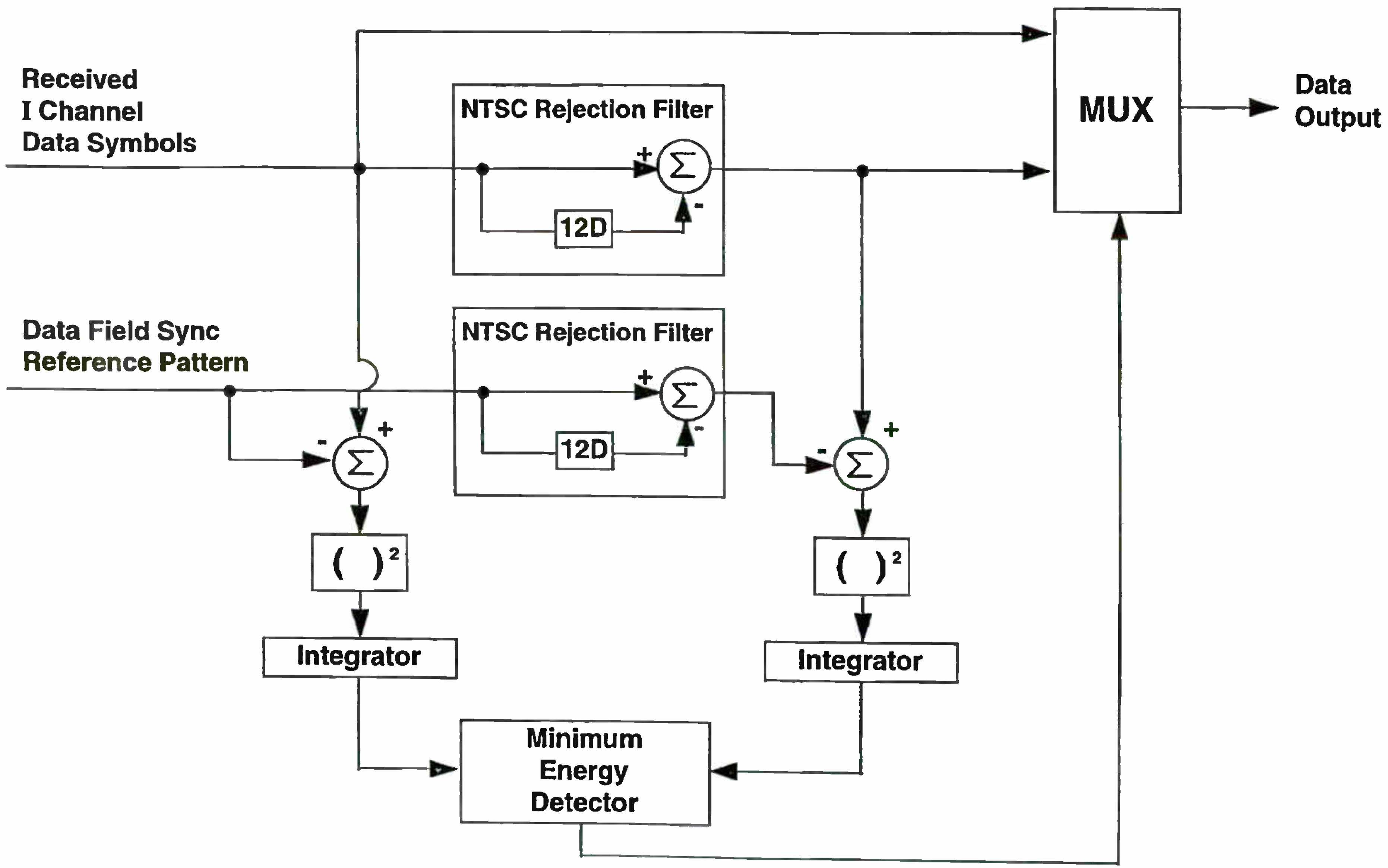


Figure 17

VSB EQUALIZER SYSTEM

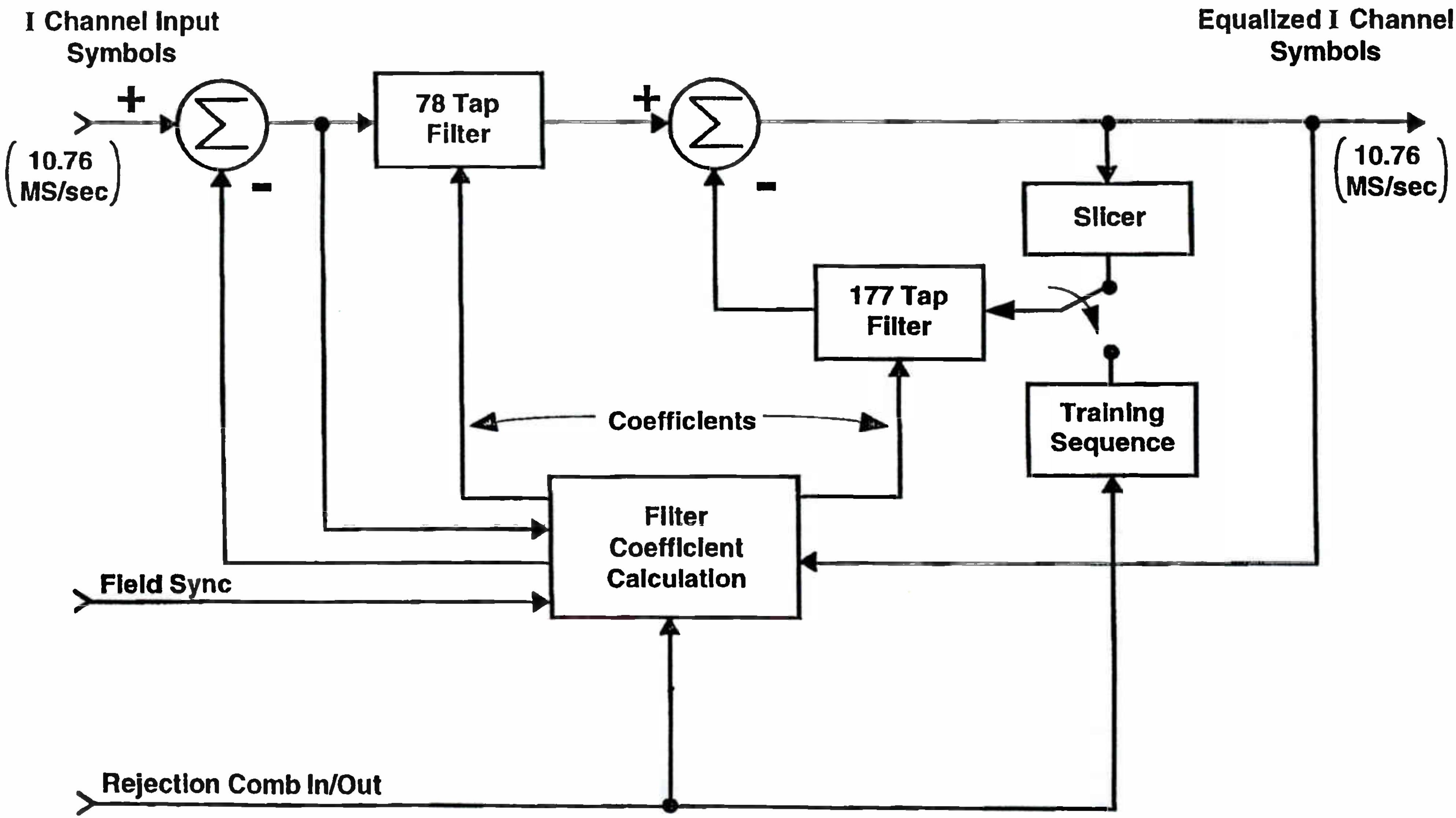


Figure 18

PHASE TRACKING LOOP

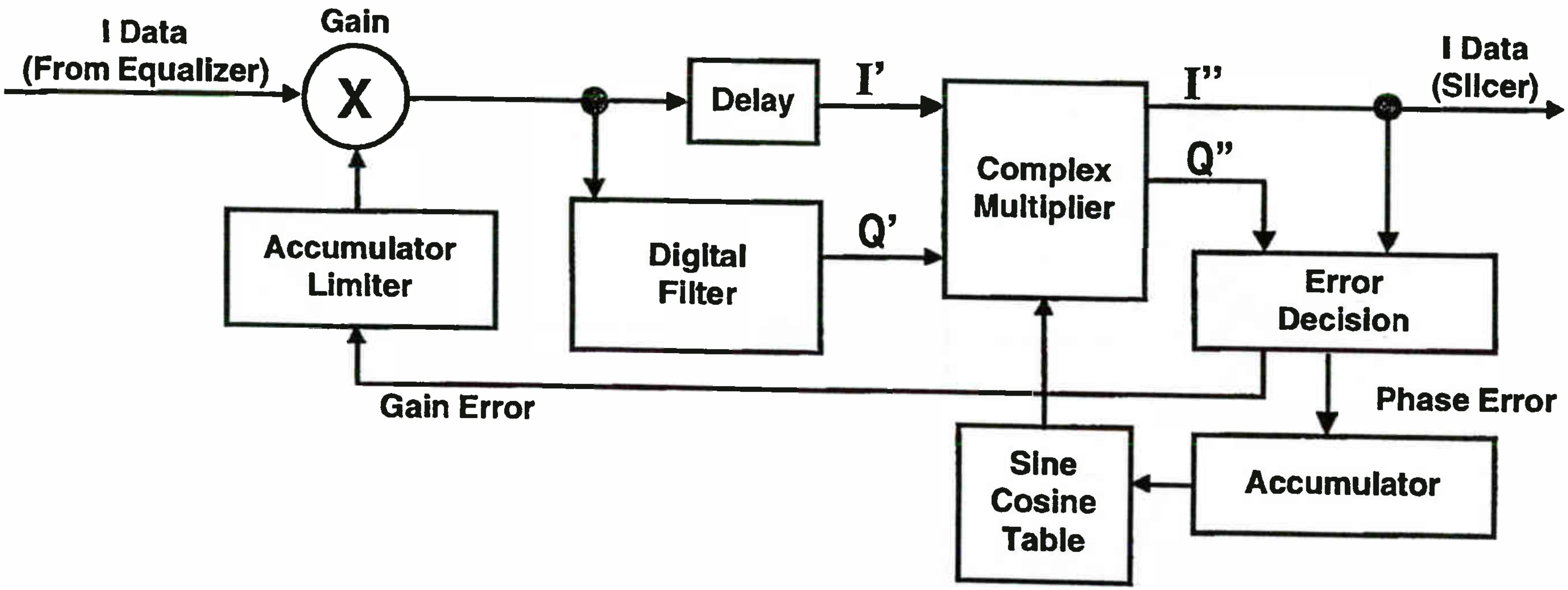


Figure 19

TRELLIS CODE DE-INTERLEAVER

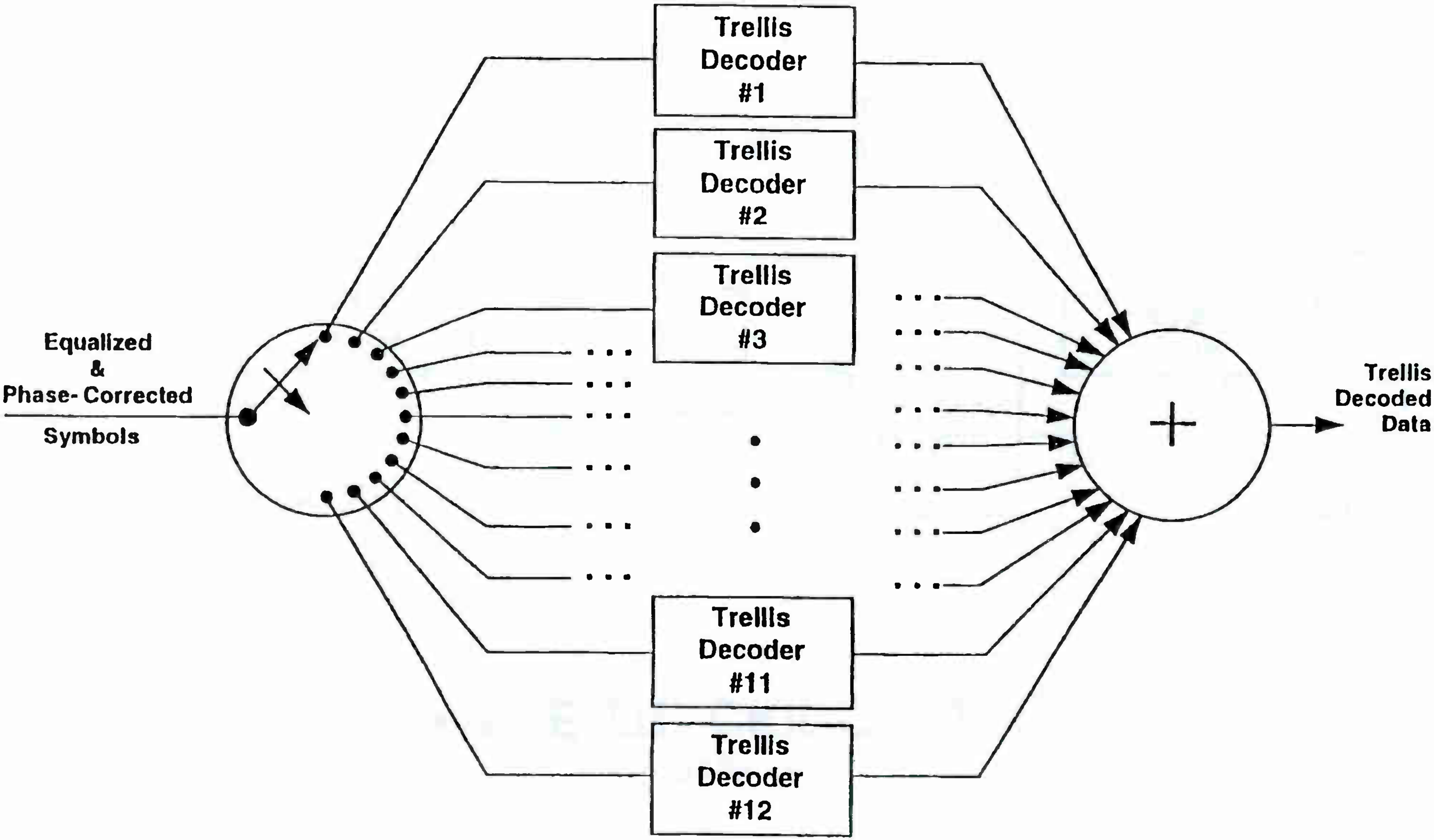


Figure 20

TRELLIS DECODING WITH AND WITHOUT NTSC REJECTION FILTER

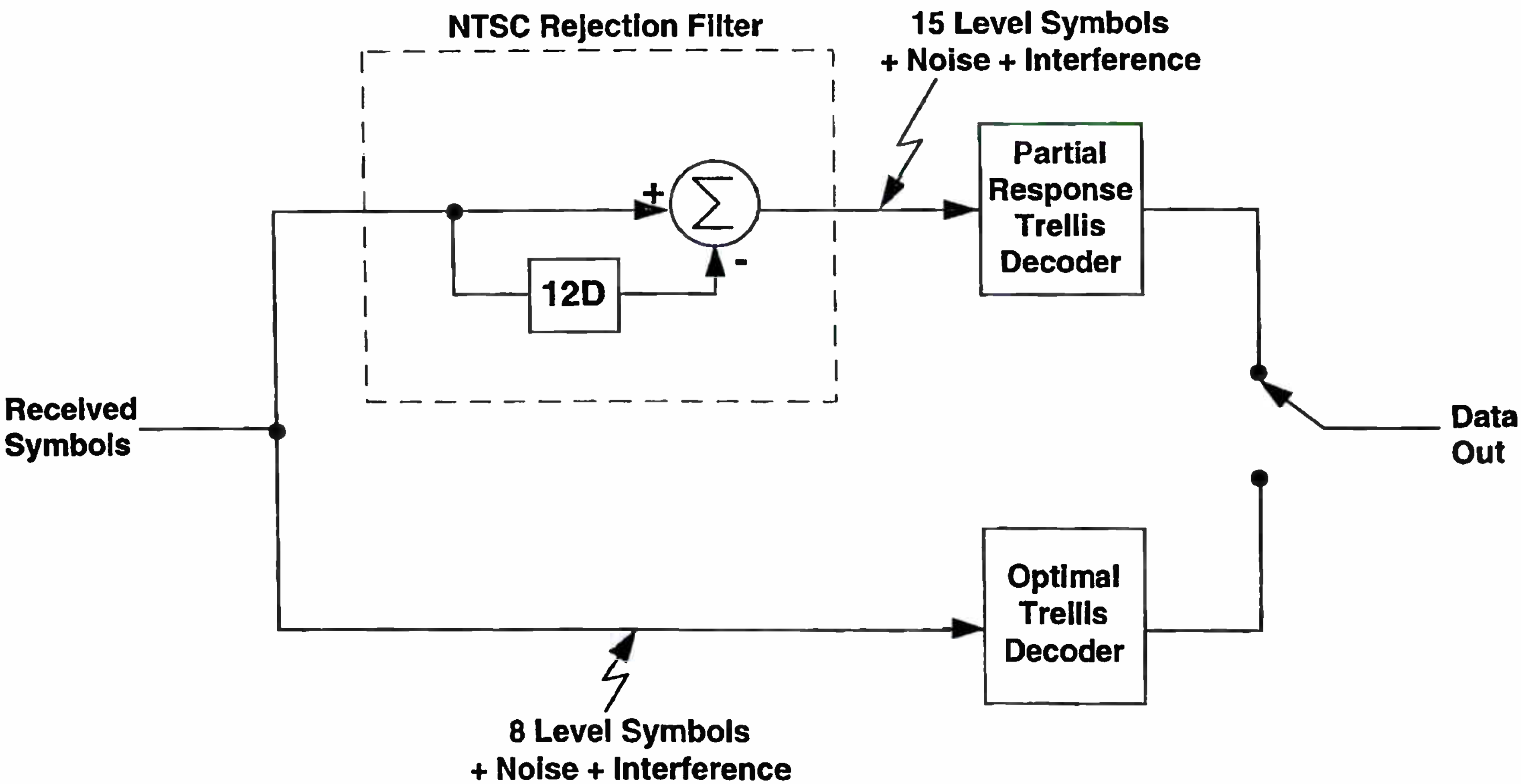


Figure 21

16-VSB AND NTSC CHANNEL OCCUPANCY

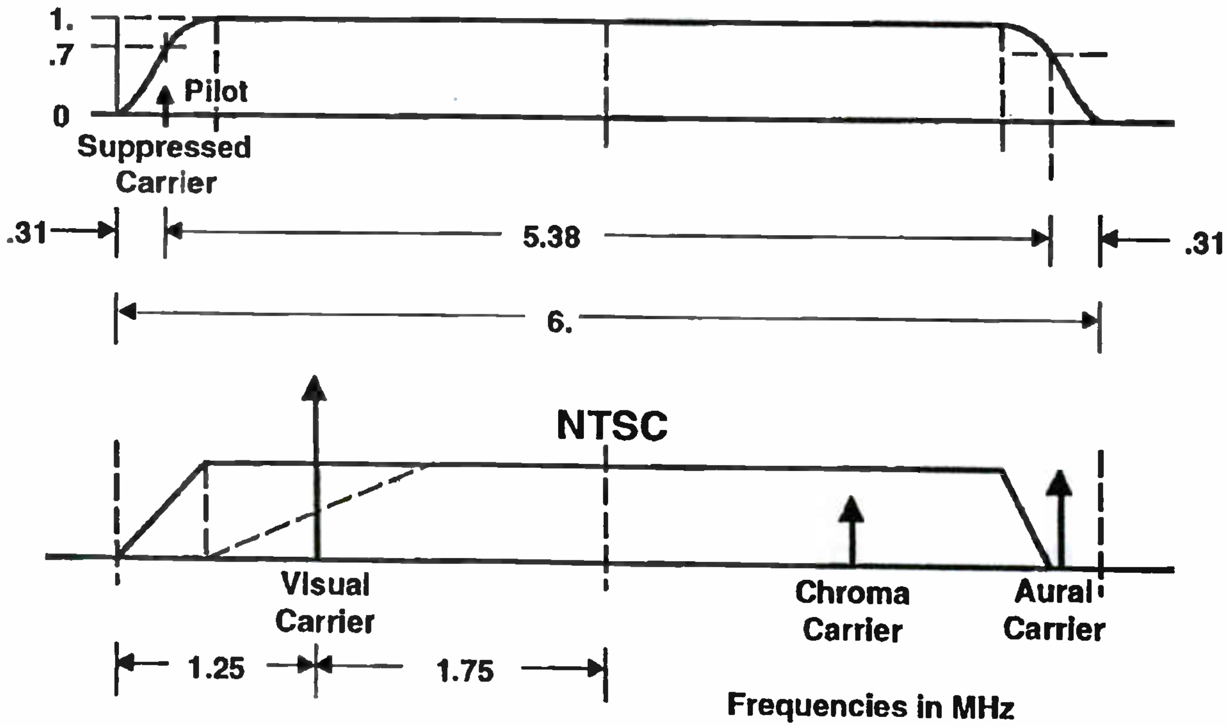


Figure 22

CUMULATIVE DISTRIBUTION FUNCTION OF 16-VSB PEAK-TO-AVERAGE POWER RATIO

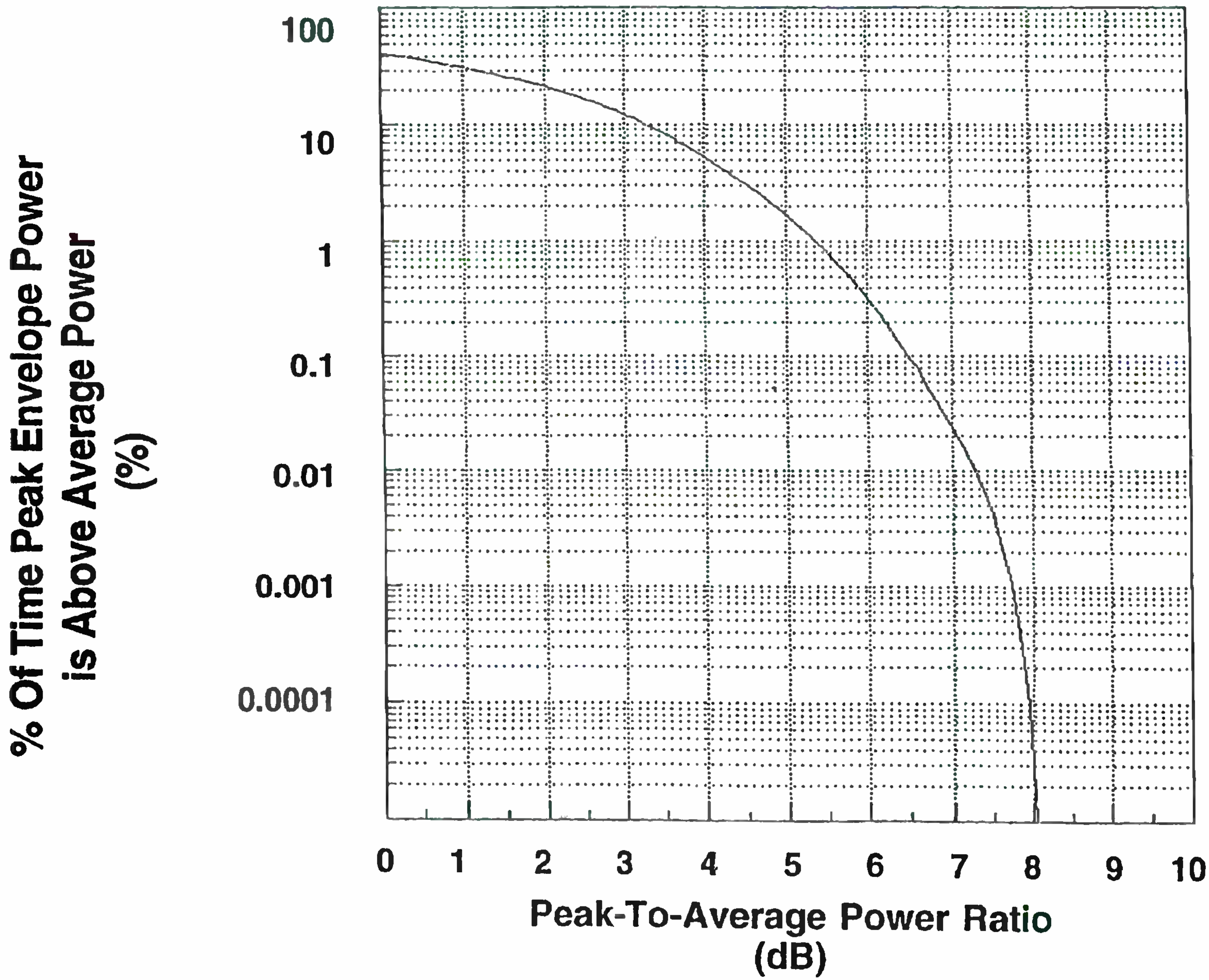


Figure 23

16-VSB ERROR PROBABILITY

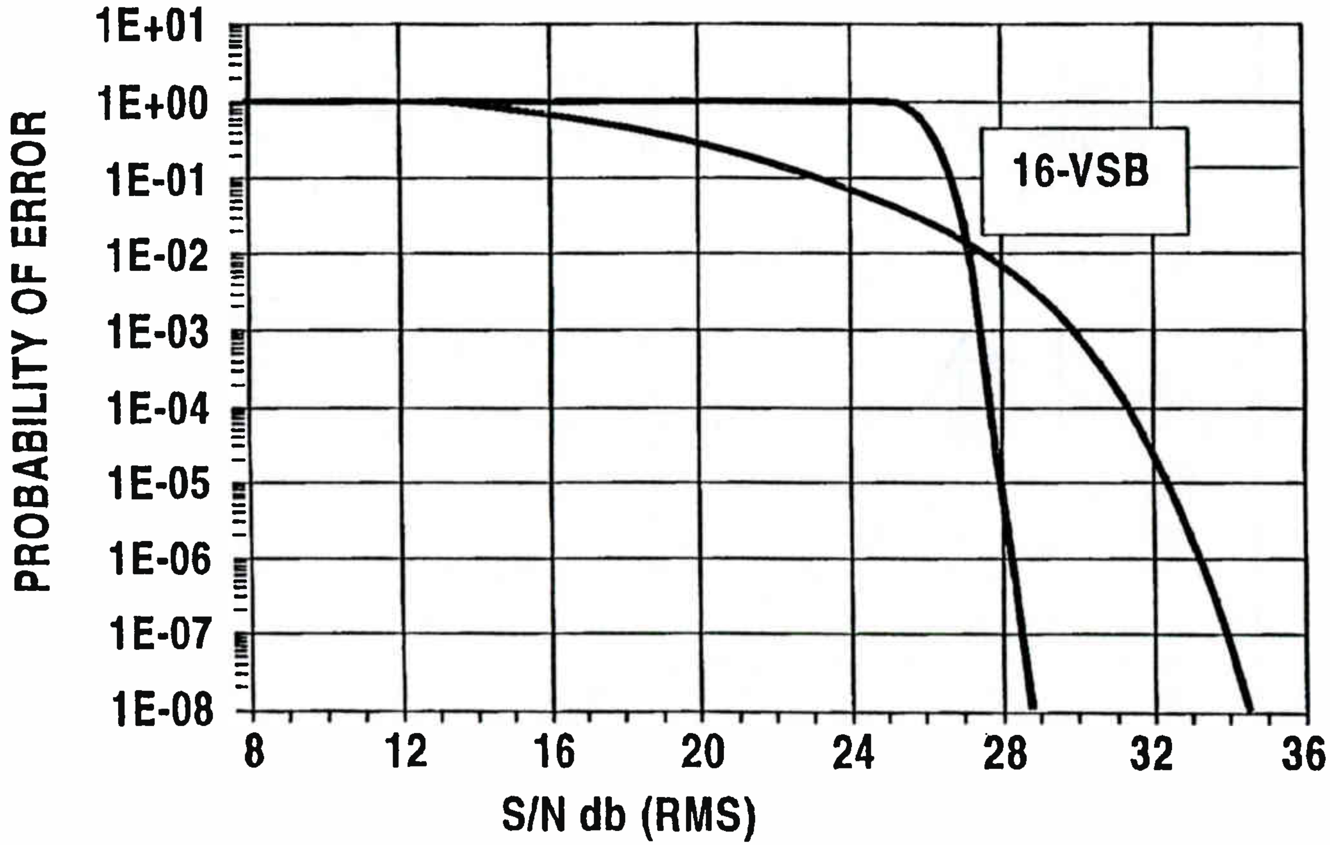


Figure 24

VSB DATA SEGMENT (Cable)

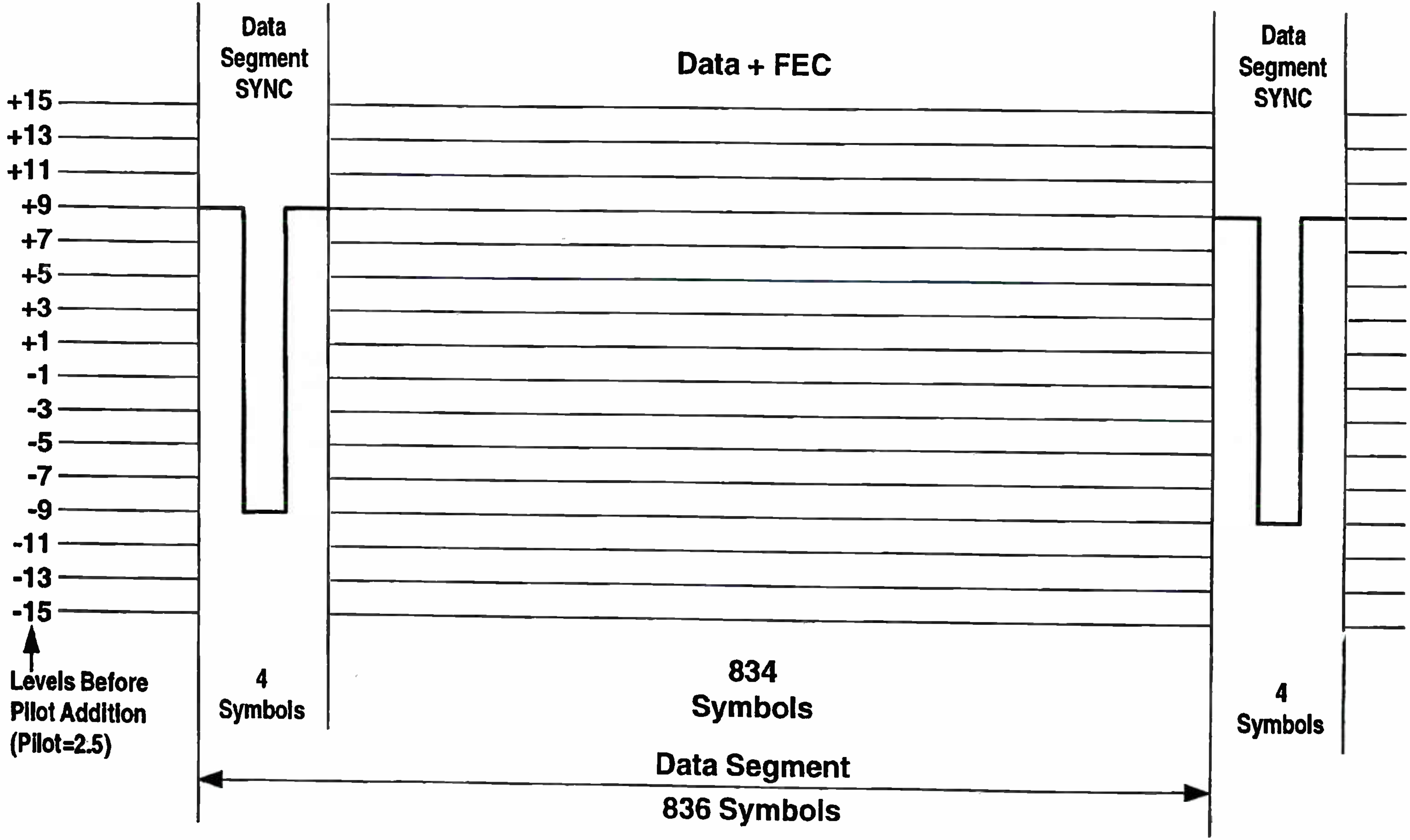


Figure 25

16-VSB TRANSMITTER

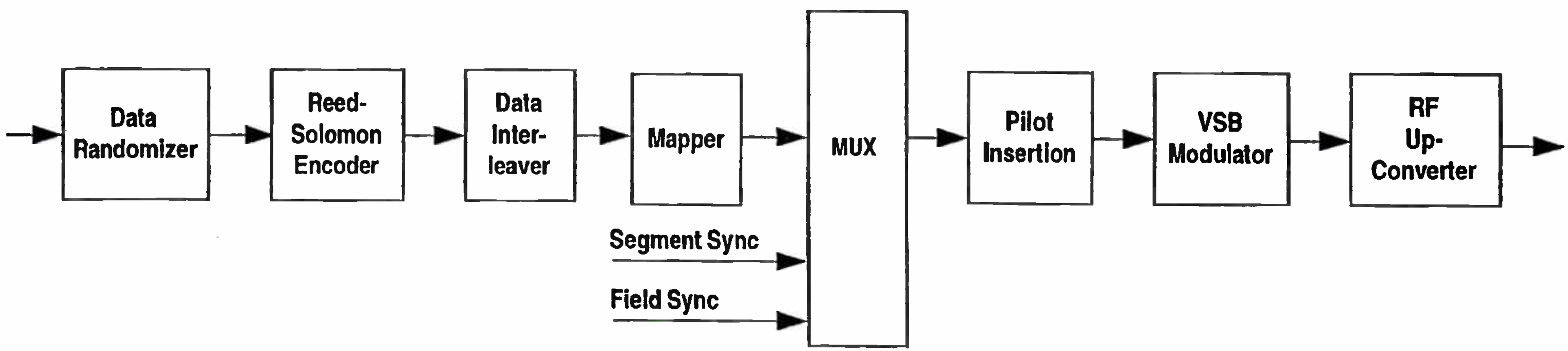
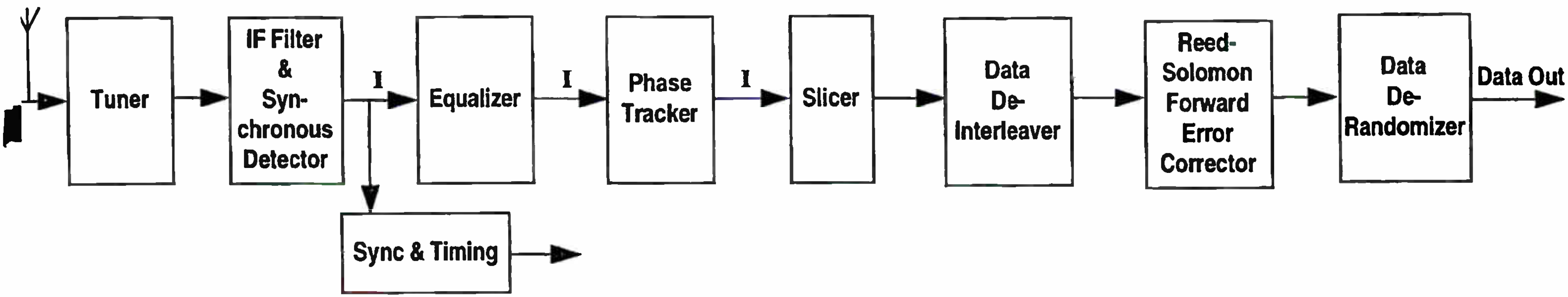


Figure 26

16-VSB RECEIVER



Chapter VII

GRAND ALLIANCE SYSTEM SUMMARY

System Summary

The following tables document features supported in the system description of the Grand Alliance HDTV specification. While the initial hardware prototype may not completely exercise all of the capabilities of the system standard, it is anticipated that future implementations will benefit from the flexibility provided by the standard.

SPECIFICATION TABLE

Video Parameter	Format 1	Format 2
Active Pixels	1280 (H) x 720 (V)	1920 (H) x 1080 (V)
Total Samples	1600 (H) x 787.5 (V)	2200 (H) x 1125 (V)
Frame Rate	60 Hz Progressive	60 Hz Interlaced / 30 Hz Progressive / 24 Hz Progressive
Chrominance Sampling	4:2:0	
Aspect Ratio	16:9	
Data Rate	Selected Fixed Rate, 10-45 Mbits / sec / or Variable	
Colorimetry	SMPTE 240 M	
Picture Coding Types	Intra Coded (I) / Predictive Coded (P) / Bidirectionally Predictive Coded (B)	
Video Refresh	I-Picture / Progressive	
Picture Structure	Frame / Field	
Coefficient Scan Pattern	Zig-Zag / Alternate Zig Zag	
DCT Modes	Frame	Frame / Field (60 Hz only)
Motion Compensation Modes	Frame	Frame / Field (60 Hz only) / Dual Prime (60 Hz only)
P-Frame Motion Vector Range	Horizontal: Unlimited by syntax Vertical: -128,+127.5	
B-Frame Motion Vector Range (forward and backward)	Horizontal: Unlimited by syntax Vertical: -128,+127.5	
Motion Vector Precision	1/2 Pixel	
DC Coefficient Precision	8 / 9/ 10 bits	
Rate Control	Modified TM5 with Forward Analyzer	
Film Mode Processing	Automated 3:2 Pulldown Detection and Coding	
Maximum VBV Buffer Size	8 Mbits	
Intra / Inter Quantization	Downloadable Matrices (Scene Dependent)	
VLC Coding	Separate Intra and Inter Runlength / Amplitude Codebooks	
Error Concealment	Motion Compensated Frame Holding (Slice Level)	

Transmission Parameter	Terrestrial Mode	High Data Rate Cable Mode
Channel Bandwidth	6 MHz	6 MHz
Excess Bandwidth	11.5%	11.5%
Symbol Rate	10.76 MSPS	10.76 MSPS
Bits per Symbol	3	4
Trellis FEC	2/3 rate	None
Reed-Solomon FEC	T=10 (208, 188)	T=10 (208, 188)
Segment Length	836 Symbols	836 Symbols
Segment Sync	4 Symbols per segment	4 Symbols per segment
Frame Sync	1 per 313 segments	1 per 313 segments
Payload Data Rate	19.3 Mb/s	38.6 Mb/s
NTSC Co-Channel Rejection	NTSC Rejection Filter in receiver	N/A
Pilot Power Contribution	0.3 dB	0.3 dB
C/N Threshold	14.9 dB	28.3 dB

SPECIFICATION TABLE

Audio Parameter	
Number of Channels	5.1
Audio Bandwidth	20-20KHz
Sampling Frequency	48 KHz
Dynamic Range	100 dB
Compressed Data Rate	384 KBits / sec.

Transport Parameter	
Multiplex Technique	MPEG-2 Systems Layer
Packet Size	188 Bytes
Packet Header	4 Bytes including sync
Number of services	
Conditional Access	Payload scrambled on service basis
Error Handling	4-bit continuity counter
Prioritization	1 bit / packet
System Multiplex	Multiple program capability described in PSI stream

Chapter VIII

PROTOTYPE HARDWARE IMPLEMENTATION

1 Overview

To demonstrate and verify the performance of the proposed standard, the member companies of the Grand Alliance, have combined their efforts to produce a hardware prototype for testing at the Advanced Television Test Center later this year. This prototype implementation will demonstrate excellent picture and sound quality, a flexible, packet-based transport, and transmission performance superior to previously tested systems. It is impractical, however, to incorporate all the flexibility and extensibility permitted by the standard into any one implementation. This chapter outlines the prototype design considerations and elements of the standard that will not be fully exercised in the initial implementation.

2 Video Compression Subsystem

The Video Compression subsystem implements a high level, main profile MPEG-2 compression algorithm. The MPEG-2 syntax offers an extremely powerful and flexible toolkit of compression techniques to cover a variety of multimedia communication capabilities. The prototype hardware will demonstrate excellent picture quality at the available bit rates, however it will not utilize all of the features available in the MPEG-2 syntax. Additionally, there are some features of the MPEG-2 algorithm that will be demonstrated with some restricted capabilities.

2.1 Picture Formats Supported:

The prototype hardware will be capable of compressing pictures in two input formats: 1920 x 1080 interlaced pictures at 60 fields per second and 1280 x 720 progressive pictures at 60 frames per second. The sequences will be coded in the format in which they are received. The compression decoder will be capable of recognizing, decoding and displaying pictures coded in either the progressive or interlace input format. Additionally, the decoder will have the capability to convert interlaced coded formats to a progressive display format. A similar capability will demonstrate conversion from progressive to interlaced formats.

The prototype will employ lookahead circuitry to detect and identify sequences that were originally sourced at 24 or 30 Hz progressive formats, and will take advantage of this redundancy by only coding the necessary frames. the hardware is not capable of directly receiving pictures whose input format and timebase are 24 frames per second. Information will be passed in the coded bitstream to allow the decoder to reconstruct and display the sequences at the appropriate input format.

2.2 Motion Compensation:

While the MPEG-2 syntax allows very extensive motion search range, the prototype will restrict its search range to +/- 128 pixels horizontally and +/- 32 pixels vertically for forward Predicted (P) frames, and +/- 64 pixels horizontally and +/- 32 pixels vertically in each temporal direction for Bi-directional predicted (B) frames. The motion vectors will be calculated and coded with 1/2 pixel accuracy. The prototype hardware will not implement concealment motion vectors.

The specification of the GA compression system includes the adaptive use of field, frame and dual prime prediction techniques for interlaced video. Since the dual prime technique is primarily useful for low-delay applications, it will not be included in the prototype hardware, in the interest of time.

2.3 Coding:

The prototype implements high-level, main profile of the MPEG-2 video coding standard. While the standard allows for variable bit-rate applications, the prototype will employ a selectable constant bit rate algorithm. The rate will be set through comminication with a computer console, and will allow the ability to show different allocations for video and data services.

Each video frame is coded in one of three modes: Intra-coding (MPEG I-picures), Predictive coding (MPEG P-pictures) or bi-directional predictive coding (MPEG B-pictures). The period of I-pictures (Group-of-pictures size, N) and the distance in number of frames between two anchor (I or P) pictures (known as M), can be programmed.

In the above mentioned mode of operation, I-pictures provide refresh. The codec will allow an alternate way of refreshing, which is achieved through progressive refresh. In this mode, parts of P-pictures (I-slices) are refreshed progressively. I-frames will be sent periodically when using progressive refresh mode to facilitate editing.

The chroma samples will be subsampled by a factor of two horizontally and vertically (4:2:0), relative to the luma sampling grid. The prototype will only use the standard coefficient scan pattern, not the alternate zig-zag. The DC DCT coefficients may be represented with a precision of either 8, 9, or 10 bits in the GA system. However the prototype will only support the use of 8 or 9 bits.

The GA specification includes the option of using field-structure pictures when processing interlaced video, however this option will not be used in the prototype. The maximum number of coded bits per frame will be 8 Mbits.

In addition to the features listed above, there are a number of extensions to the syntax supported in the GA system that are not implemented in the prototype. Pan and scan and picture display extensions will not be implemented. Also absent from the prototype will be spatial and temporal scalability extensions, that are not part of the main profile.

3 Audio Compression Subsystem

The prototype implementation of the audio compression subsystem is expected to include all features in the system description. Further details of the Audio compression prototype forthcoming.

4 Transport Subsystem

The transport subsystem is responsible for multiplexing the variety of services into a single bit stream for transmission. This subsystem also has the responsibility for managing and delivering synchronization information between the encoders and decoders. The GA transport system provides an extrordinary degree of flexibility. The transport bit stream could describe a large number of programs, each potentially consisting of a large number of individual services. This flexibility is intended for a broad scope of services and to allow for future growth.

The prototype hardware will implement all the MPEG-2 syntax elements that would allow this high degree of flexibility, but obviously will be limited in its ability to simultaneously exercise this flexibility. Nominally, the prototype will carry a single program of HDTV services. The prototype will, however, be capable of delivering two independent programs simultaneously. Each program can be comprised of up to five independent services (typically, one video, two audio, and two ancillary data services). The ancillary data services can be either synchronous or asynchronous with the video and audio services. For an 18.8 Mbps transmitted data stream, the transport decoder will be able to recognize all five of these services, but will simultaneously decode only three services (one video, one audio, and one data).

Prototype Hardware Implementation

The capability to assign a higher priority to certain transport packets is a capability of the system that will not be used in the prototype hardware. The splice countdown function will also be absent in this first hardware implementation, however the system will allow for the discontinuity flag to signal the decoder of switching that has occured in the compressed bit stream domain.

To demonstrate the ability of the transport layer to support conditional access, the prototype will include the capability to scramble the payload of individual packets on a "service by service" basis. Packets whose payloads include adaptation headers, will not be scrambled. As a demonstration, this scrambling circuitry is based on the DES electronic codebook encryption algorithm. Key management for the decryption of these services can also be demonstrated as an example implementation.

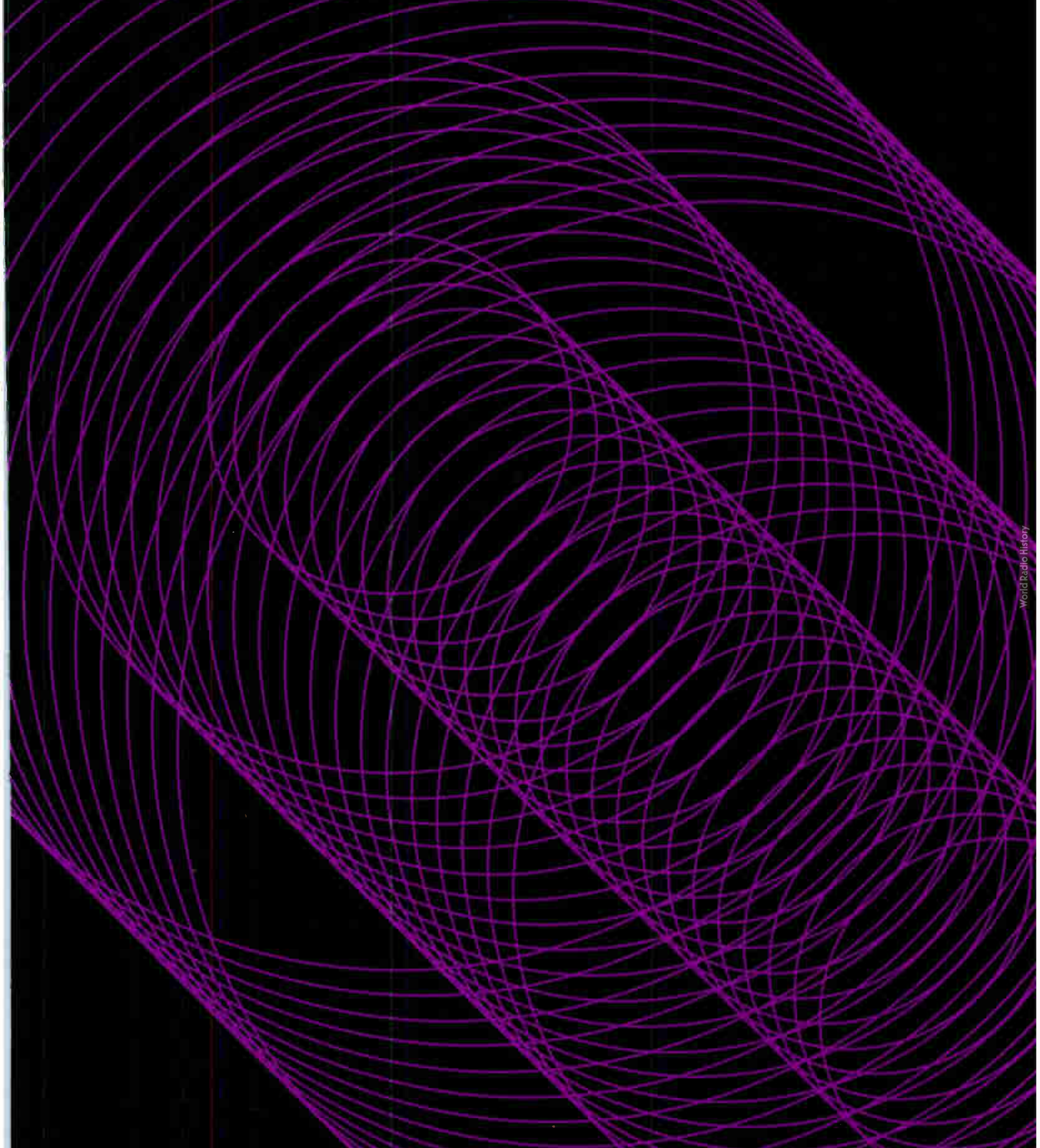
5 Transmission Subsystem

The prototype implementation of the transmission subsystem is expected to include all features in the system description. Further details of the Transmission prototype will be forthcoming.

6 Projected Prototype Performance

The following table details the typical performance anticipated for the prototype implementation. The information for this table originates from a number of sources, including the recently completed transmission bakeoff. Further additions and refinements to this table are expected as the prototype approaches completion.

1994 PROCEEDINGS



48TH ANNUAL BROADCASTING ENGINEERING CONFERENCE PROCEEDINGS

NAB BROADCASTERS

Projected Typical Prototype Performance

Performance Parameters	
A/V Noise Threshold (C/N at TOV 6 MHz)	~15.00
Cochannel Interference (D/U, weak desired signal) NTSC into ATV (@ TOV ATV into ATV (@ TOV, worst case) ATV into NTSC (@grade 3)	 ~ 2.5 ~ 16.0 ~ 33
Adjacent Channel Interference (D/U, weak desired signal) ATV into ATV Lower (@ TOV) ATV into ATV Upper (@ TOV) NTSC into ATV Lower (@ TOV) NTSC into ATV Upper (@ TOV) ATV into NTSC Lower (@ grade 3) ATV into NTSC Upper (@ grade 3)	 ~ -40 ~ -40 ~ -45 ~ -45 ~ -15 ~ -15
Taboo Channel Interference, ATV into NTSC (D/U, weak desired signal) n-2 Taboo @ grade 3 n+2 Taboo @ grade 3 n+4 Taboo @ grade 3 n+14 Taboo @ grade 3 n+15 Taboo @ grade 3	 ~ -28 ~ -33 ~ -25 ~ -35 ~ -25
Phase Noise (@ 20 KHz) (dBc/Hz) Burst Errors at 20 µS Width (rep rate in Hz) Residual FM (kHz) Pull-In Range (kHz) Impulse Noise (µS)	~ -70 ~ 1500 ~ 8 +/-100 ~ 150
System Attributes Net data rate after FEC (Mbps) Peak to Average Ratio (99.9%) (dB) Cable Interoperability (Mbps)	 18.8 6.2 37.55
Subjective Picture Quality	TBD
Audio Quality	TBD
Resolution	TBD

Chapter IX

REFERENCED DOCUMENTS

Referenced Documents

The following documents are referenced for their detailed account of the syntatical elements of the Grand Alliance HDTV standard.

1) Generic Coding of Moving Pictures and Associated Audio
ISO/IEC 13818-1 Systems Committee Draft
November 1993

2) Generic Coding of Moving Pictures and Associated Audio
ISO/IEC 13818-2 Video Committee Draft
November 1993

3) Dolby AC-3 Audio Compression Algorithm Description
February 1994

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