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NATIONAL COMMUNICATIONS SYSTEM

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INVESTIGATION OF HIGH DEFINITION TELEVISION FOR APPLICATION TO TELECONFERENCING

FEBRUARY 1992

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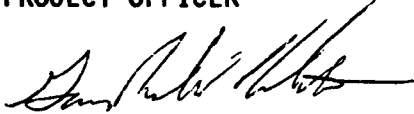
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INVESTIGATION OF HIGH DEFINITION TELEVISION
FOR APPLICATION TO TELECONFERENCING

FEBRUARY 1992

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FOREWORD

Among the responsibilities assigned to the Office of the Manager, National Communications System, is the management of the Federal Telecommunication Standards Program. Under this program, the NCS, with the assistance of the Federal Telecommunication Standards Committee identifies, develops, and coordinates proposed Federal Standards which either contribute to the interoperability of functionally similar Federal telecommunication systems or to the achievement of a compatible and efficient interface between computer and telecommunication systems. In developing and coordinating these standards, a considerable amount of effort is expended in initiating and pursuing joint standards development efforts with appropriate technical committees of the International Organization for Standardization, and the International Telegraph and Telephone Consultative Committee of the International Telecommunication Union. This Technical Information Bulletin presents an overview of an effort which is contributing to the development of compatible Federal, national, and international standards in the area of High Definition Television as applied to Video Teleconferencing. It has been prepared to inform interested Federal activities of the progress of these efforts. Any comments, inputs or statements of requirements which could assist in the advancement of this work are welcome and should be addressed to:

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**INVESTIGATION OF HIGH DEFINITION
TELEVISION FOR APPLICATION
TO TELECONFERENCING**

February, 1992

**FINAL REPORT
DCA100-91-C-0031**

**Submitted to:
NATIONAL COMMUNICATIONS SYSTEM
WASHINGTON, DC**

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1.0 INTRODUCTION

This document summarizes work performed by Delta Information Systems, Inc. (Delta) for the National Communications System (NCS), Office of Technology and Standards. The NCS is responsible for the management of the Federal Telecommunications Standards Program, which develops telecommunications standards, whose use is mandatory for all Federal departments and agencies.

This document is a final report for a Task Order on Contract DCA100-91-C-0031. The titles for the contract and Task Order are listed below.

- **Contract DCA100-91-C-0031**
Development of Federal Telecommunication Standards Relating to Digital Facsimile and Video Teleconferencing.
- **Task Order No. 6 (1991)**
Investigation of High Definition Television for Application to Government Teleconferencing.

In recent years, there has been considerable activity in the development of technology and standards related to High Definition Television (HDTV). According to the Advanced Television Systems Committee (ASTC), "the term HDTV refers to television systems with approximately twice the horizontal and vertical emitted resolution of standard NTSC. HDTV systems are wide aspect ratio systems and may include improvements from IDTV (Improved Definition Television) and EDTV (Extended Definition Television)". The purpose of this task was to investigate HDTV to determine its potential applicability to teleconferencing within the Government community. Work on this project was divided into four tasks -- (1) HDTV standards activity, (2) communications considerations, (3) TV compression technology, and (4) System Analysis.

HDTV STANDARDS ACTIVITY

One of the most fundamental and complex tasks facing the FCC is the selection of the scan format for the HDTV signal (e.g. number of scan lines, interlace vs. progressive scan, bandwidth, etc.). This decision will have a significant impact on the ease with which the signal is used for teleconferencing. For this reason, work on this project was directed toward a review of the HDTV

standards activity from this perspective. Particular attention has been given to the four all-digital proposals for the HDTV standard.

COMMUNICATION CONSIDERATIONS FOR TELECONFERENCING

In general, video teleconferencing requires a high transmission bit rate relative to other services such as voice and data. For that reason, the availability of teleconferencing for the government community is dependent upon the availability of ubiquitous, inexpensive, switched, communication channels operating at high bit rates. The purpose of this section is to, in very general terms, examine communication issues as they relate specifically to video teleconferencing. The discussion is divided into four parts; (1) teleconferencing communication today, (2) narrowband ISDN, (3) broadband ISDN, and (4) FTS-2000.

COMPRESSION OF THE HDTV SIGNAL

If teleconferencing is to be practical, the signal must be transmitted over switched communication channels having a bit rate low enough to be economical. The teleconferencing industry has been growing rapidly in recent years because compression is even more critical for HDTV then it is for the NTSC signal. The purpose of this task is to examine compression technology for the application of HDTV to teleconferencing. Work on this task was divided into three parts as listed below.

- Compression technology was reviewed in general to provide a broad background for further coding studies.
- Particular attention has been directed toward a discussion of the Discrete Cosine Transform (DCT) since this algorithm forms the basis for all four digital HDTV proposals.
- From a telecommunications perspective, it is very important that the HDTV system be as interoperable as possible with other standards such as the CCITT P x 64. These issues of scalability and interoperability are discussed and analyzed.

SYSTEM ANALYSIS

The purpose of this task is to integrate the results of the above three tasks into complete end-to-end communications services and to examine these potential services as they relate to the government requirements. Examples of government needs which are examined include the following.

- teleconferencing
- videophone
- command and control function
- training
- multimedia

Conclusions drawn from the work performed on this project are summarized in Section 6.0.

2.0 HDTV STANDARDS

In 1991, there was considerable effort and activity to develop standards for High Definition Television (HDTV). For convenience, this work has been divided into the international and domestic areas as described in Sections 2.1 and 2.2 respectively.

Before proceeding further, it is useful to define HDTV as it relates to other TV standard's work which are included under the broad category of "Advanced Television" (ATV). A number of ATV systems have been proposed which can be categorized as Improved Definition TV (IDTV) and Extended Definition TV (EDTV), and High Definition TV (HDTV). These terms are defined by the Advanced Television Systems Committee (ATSC), the standards group formed by the TV industry, as follows.

IDTV - IMPROVED DEFINITION TELEVISION - The term Improved Definition Television refers to improvements to NTSC television which remain within the general parameters of NTSC emission standards and, as such, would require little or no FCC action. Improvements may be made at the source and/or at the television receiver and may include improvements in encoding, filtering, ghost cancellation, and other parameters that may be transmitted and received as standard NTSC in a 4:3 aspect ratio.

EDTV - EXTENDED DEFINITION TELEVISION - The term Extended Definition Television refers to a number of different improvements that modify NTSC emissions but that are NTSC receiver-compatible (as either standard 4:3 or "letter-box" format). These changes may include one or more of the following:

1. Wide aspect ratio.
2. Extended picture definition at a level less than twice the horizontal and vertical emitted resolution of standard NTSC.
3. Any applicable improvements of IDTV.

For purposes of identification, EDTV transmitted as 4:3 is referred to as EDTV, and when transmitted in a wider aspect ratio, as EDTV-Wide.

HDTV - HIGH DEFINITION TELEVISION - The term High Definition Television refers to television systems with approximately twice the horizontal and vertical emitted resolution of standard NTSC. HDTV systems are wide aspect ratio systems and may include applicable improvements from IDTV and EDTV.

IDTV and EDTV feature compatibility with the existing NTSC system which automatically puts severe limitations on their achievable performance. HDTV eliminates this constraint and thus holds promise for future development.

2.1 International Standards

High Definition Television has existed in different forms for many years to serve primarily niche applications. Black and white versions were available more than 20 years ago for industrial applications requiring more than NTSC (National Television System Committee - the present 525 system in use in the U.S. today) resolution. Some of these systems employed over 1000 lines for full motion detail. In the computer display world high definition has been utilized for many years in mid and large sized computer systems for computer aided design, geographical map displays, etc. Many of these systems were in color but did not have to display full motion imagery, thus permitting a more primitive vector drawing method and fewer updates per second than present day raster scan television systems. However, no commercial broadcasting used these systems and standards were not set for video bandwidth, number of lines, interlace (if used), frames per second and color encoding technique.

In the mid-1970's Japan put together the first full color and full motion high definition system employing 1125 scanning lines, 2:1 interlace and 60 fields per second. This system employed a 5:3 aspect ratio (widescreen) format to better accommodate the televising of the present day motion picture film format.

The technology was refined and several steps accomplished toward bringing the Japanese HDTV system closer to the market. First, large screen CRT (cathode ray tube) displays up to 40 inches diagonal were perfected which permits viewing a receiver at close to normal consumer living room viewing distances. Second, video tape recorders - both analog and digital - were developed to save televised segments and programs after they had been made, for later editing and final presentation to the intended audience. Third, a method of reducing the bandwidth of color signals into a single 8 MHz. composite signal with little subjective quality

deteriorization. This encoding process is called MUSE, for Multiple Sub-Nyquist Encoding.

This Japanese HDTV transmission system has no direct channel compatibility with any existing TV systems of the world since it requires 8 MHz. bandwidth, and no direct receiver compatibility in that no existing receivers (NTSC, PAL or SECAM) can use a received signal to generate a TV picture with it. This incompatibility was not an oversight but instead intended to be a Direct Broadcast Satellite transmission system utilizing all new equipment, including the customers' receivers. The particular MUSE encoding method is a mostly analog compression system employing a considerable amount of digital storage of imagery. As such, one might call it a hybrid encoding system. The signal compression is achieved by extension of the interlace to 4:1, filtering out some little used signal components, employing reduction of color-difference signal bandwidths (a technique also used in the NTSC system) and a primitive, whole-screen, form of motion compensation. The compression is achieved by discarding little used components of the original signal, as opposed to the technique of redundancy reduction and statistical coding used today in modern compression systems (for example, the H.261 CCITT standard for Audio and Video Teleconferencing). The MUSE system has become fully operational in Japan after launch of a satellite made for this purpose with sufficient power to enable use of fairly small receiving antennas (dishes).

Outside of Japan interest in HDTV mounted during the 1980's to the point where other nations of the world began thinking in terms of HDTV broadcast transmission in their own countries. The United States generally favored the 1125/60/2:1 Japanese system but Europeans who employ 50 Hz. field rate systems (PAL and SECAM) saw it as too much of a departure from their present transmission systems. In 1986 the CCIR convened a Plenary Session in Dubrovnik, Yugoslavia to consider a world-wide standard for HDTV and the 1125/60/2:1 was submitted for approval. The French, who had done quite a bit of homework on the matter, led the European opposition to the Japanese system and although the U.S. did considerable lobbying for it, an impasse developed and no standard was accepted. The U.S. took the position that the 1125 line system could at least be adopted as a "studio standard", a basis for generating the basic TV program material, after which the result could be standards-converted to whatever technical line-rate, field-rate, etc. format was later determined for local broadcast. But this was not acceptable to the Europeans either. The session

ended with the action of assigning work to a study group which was to make its report and recommendation two years hence.

Rather than working out differences and coming toward a world-wide standard, since that time positions taken by various countries have either solidified or turned away from the Japanese system. The one thing that seemed to come out of the Dubrovnik session was a renaissance of HDTV development activity entered into by many organizations throughout the world including broadcasters, equipment suppliers and universities. A new technical focus on HDTV was in vogue and with it came proposals for a wide variety of high definition as well as not-so-high definition systems. Meanwhile, the standards activity in Europe has solidified around a system employing 1250 lines, 50 fields per second and 2:1 interlace. This system is considered to be more compatible with the existing PAL and SECAM system in that integer ratios exist in scanning parameters making down-converting of HDTV video signals to PAL signals much more convenient. It is estimated that both conventional resolution and HDTV will co-exist for some time and that new TV programs made for HDTV will also be converted and shown on existing PAL and SECAM broadcast channels. It is generally believed that the Europeans consider the HDTV situation in economic terms as well, thinking it easier to maintain production and manufacturing in Europe rather than in Asia if a 1250 line standard is adopted.

Plans for Direct Satellite Broadcasting of HDTV in Europe are in place with coordination coming under the Eureka project, a collaboration of some 17 countries in western Europe. In that a satellite system is envisioned the transmitted signal need not be limited to 6 MHz. or the present channelization for PAL which is slightly greater than for NTSC. This permits use of a compression system with less compression. In particular, one of the MAC (Multiplexed Analog Components) transmission systems of European origin will be used.

It is interesting to note the results of the "Report of the Moscow Group HDTV Evaluations" which were published July 5, 1990 by the EBU (European Broadcasting Union). These tests were set up and conducted over a two year period to compare the performance of the 1125 line system and the proposed 1250 line European system. Briefly, the results indicated that subjective viewing of images in the 1250 line format were slightly inferior to the 1125 line system despite having higher resolution. The reason for this is mostly the lower field scanning rate (50 HZ.) of the European system which exhibits more large area

flicker than a 60 Hz. system. Whereas this has always been a problem with 50 Hz TV systems, it is not as noticeable on smaller displays associated with lower definition. To solve this problem with a 50 Hz. system one must usually build progressive scanning into the receiver such that the effective field rate becomes 100 Hz.

In 1988, the F.C.C. working with a group, the Advanced Television Systems Committee (ATSC) sponsored by broadcasters, entertained proposals for a broadcast standard for the U.S. The submissions had rather little in common in that some were receiver compatible [NTSC receivers could receive the new HDTV signal and display it with reduced resolution - about the same as NTSC] with the present NTSC system while others were not, and some were channel compatible [occupied the same 6 MHz. as the present NTSC signals] while others were not. Systems were so different that three names were generated to try to categorize them. There were: 1) improved definition, 2) extended definition, and 3) high definition, which took into account roughly the amount of resolution increase that was to be had by a system. These same three categories also generally meant 1) requiring little or no FCC approval for adoption and provides a modest increase in resolution, along with improvements in picture quality, 2) required FCC approval for new signal components introduced into the basic NTSC signal to provide widescreen image aspect ratio and increased resolution and picture quality, and 3) a doubling of the present vertical and horizontal resolution, extending the image aspect ratio to 16:9, and requiring FCC approval. On top of all this, systems were categorized as 1) being contained in a single 6 MHz. channel, 2) using a second supplementary channel, usually less than 6 components, and 3) using two (usually adjacent) TV channels to transmit a HDTV signal wherein one of the two channels contained the information to provide higher definition.

At the last minute the Zenith Corporation submitted a proposal which had a revolutionary aspect to it, that of spectrum compatibility, wherein TV channels which previously were unusable in a particular geographic area due to interference with those already in use could now be used with little or no interference. This opened the door to simulcast systems wherein both a NTSC signal on one rf channel and a (6 MHz) compressed HDTV signal on a second rf channel (perhaps directly adjacent in channel number) could be employed without requiring any new rf bandwidth dedicated to television broadcast. This system would be like the radio AM and FM system of earlier days wherein the same programs were

broadcast on both AM (fairly low audio bandwidth) and FM (high fidelity). A significant technical advantage of this generic system is that a high definition TV system need not be attempted to be built on top of the present NTSC with its attendant distortion and artifact problems. Furthermore, the simulcast system need not be an analog system either in that no direct compatibility with the NTSC signal need be maintained.

Although Zenith's encoding method falls short of being a digital system, which many believe to be the ultimate category of choice, it did open up a category of systems in broadcasters eyes which adaptively select video signal components filtered in the frequency domain for transmission or for discarding based upon their importance (amplitude, in general). Although Zenith proposed it and is promoting it, much of the groundwork for it was done under M.I.T. auspices. With this initial acceptance of adaptability built into a consumer product and its assurance that it really wouldn't be expensive (the camel's nose was firmly under the tent, so to speak) relative to acceptance of other systems with similar signal dissection and selective transmission of components to attain sufficient compression to fit the result into a 6 MHz. rf channel.

Most of the technical effort towards transmission systems and an accompanying standard in the late 1980's was directed toward improved and enhanced TV systems. Ironically, Japan was at the forefront of these efforts, although by no means alone, and has been responsible for several technical initiatives toward achieving higher resolution imagery as well as the aspect ratio increase all within the framework of NTSC compatibility. Japan's interest is probably both geographic and economic. First, Japan uses the same NTSC system as employed in the U.S. and has a wide base of NTSC receivers and broadcasters. HDTV is seen as taking a long time, due to its initial high cost, to penetrate the Japanese consumer market and a market probably exists in Japan for improved or enhanced definition television in the interim. The economic reason may have been due to the perceived U.S. market which had been thought to be evolutionary, starting with improved definition television, migrating to enhanced definition and finally some time in the future achieving a high definition system. Japan certainly has wanted to be a large part of the consumer television market in the U.S.

The FCC changed all of this with its announcement that it would not consider improved and enhanced television proposals until (which may mean never) it had first considered full HDTV proposals using the simulcast mode. Somewhat

later the General Instruments Corp. proposed to the FCC that a new all-digital system of its own design be allowed to be tested along side those already proposed in the up-coming evaluation to be conducted by the ATTC (Advanced Television Test Center). This occurred right after the suggestion by Senator Markey, Committee Chairman of Telecommunications, suggested that the FCC study a digital type of transmission system as a possible HDTV standard. General Instrument was permitted to be included in the test program. The G.I. system is significant in that it embraces Discrete Cosine Transform (DCT) processing for signal compression, the same as had been used in the CCITT standard H.261. The proposal of this system may never have come about had not integrated circuits been developed for the DCT and associated motion compensation processing which have showed that this type of signal compression need not be too expensive for a consumer item. The development of these integrated circuit chips have no doubt been motivated in large part by the H.261 standard.

It is possible that an all-digital standard using the DCT may ultimately result for a HDTV transmission standard in the U.S., but that is not sure and also is not presently the front runner in the race. On the other hand, the reason for this may be that it has only recently entered the race.

Problems unique to the terrestrial broadcasting environment need to be overcome before a standard for fully digital transmission can be embraced, and it is this service which presently has all the standards activity. The problems center around HDTV reception in the present of "noise" (the weak signal condition) and "ghosts" (the multi-path transmission problem). Digital transmission of compressed imagery is generally very sensitive to transmission errors, and even with the usage of Forward Error Correction (FEC), conditions exist in the terrestrial broadcast environment which cause bursts of errors which are very difficult to handle. The digital compression systems usually employ interframe coding which takes advantage of the redundancy between successive image frames. To do this the system requires that a new image frame be constructed at the display based in part on the previous image frame, and so forth. When one image frame is corrupted by a noise burst the errors introduced propagate also to subsequent frames. Thus a digital transmission system has to be cleverly designed to minimize this unique problem, whereas an analog system, or a hybrid analog and digital system which has very limited subsequent usage of previous frame imagery does not exhibit the error propagation problem. The adoption of an all digital standard

for terrestrial broadcast may well hinge on the success of solving this particular technical problem.

Another aspect of the simulcast method of transmission is a potential requirement, at least in the case of the Zenith proposal, that the scan rates for the HDTV signal be perfectly locked to those of the present NTSC signal, which are of course fixed. This of course places a significant constraint on the scanning and frame rate parameters used for the HDTV standard. The field rate must be related to 59.94 Hz., the NTSC rate, and not 60 Hz as in the 1125 line system. Similarly the number of lines must be related to the NTSC number of lines per field of 262.5 (525 per frame). The possibilities for HDTV are therefore 1050 (4 X 262.5) as well as some other multiples of 262.5. This requirement is dependent upon the encoding method and may well not occur for an all-digital system.

An Extraordinary Meeting of the CCIR Study Group 11 in May 1989 discovered a rather unchanged situation [1]. Standard candidates stood unmoved and unmovable. In the meantime, IWP 11/6 had produced a *form sheet* containing a collection of all relevant parameters and started to fill in the empty spaces with agreed upon figures, a process that gained some success and continued during the Extraordinary Meeting. Among those where agreement still did not exist were the:

- Loci of the primary colors
- Shape of the blanking period of the analog HDTV signal
- Synchronizing signal of the digital HDTV signal
- Number of active lines and, associated with it, the pixel geometry
- Scanning principle -- interlaced versus progressive
- Field rate

CCIR STATUS - CCIR Recommendation 709 gives details on *levels, blanking, synchronization* (the latter interrelated with the scanning principle and sample arrangement chosen), *sampling frequencies*, and the *spatial/temporal order of samples*. These parameters as well as statements on *quantization* and *dynamic range* are subject to further studies.

In the course of a general reorganization of the CCIR work, the IWP 11/6 was disbanded and a new Task Group (TG) 11/1, *High Definition Television for Studio and Program Exchange*, was established reporting directly to CCIR Study Group 11 that will further handle the topic of HDTV. In detail, its tasks are:

- Completion of outstanding parameter values in Recommendation 709
- Characteristics and parameters of interfaces within the HDTV studio
- Characteristics and parameters of interfaces with terrestrial broadcasting emission systems

The U.S. has been a major contributor to the proceedings of the CCIR in the HDTV area. For example, the U.S. contributed the document in Appendix A to the CCIR in September, 1991. This document, entitled "Interoperability Considerations for Digital HDTV", nicely summarizes the objectives of organizations such as the NCS to achieve a high degree of interoperability with other media such as telecommunications and computers.

2.2 U.S. Standards

2.2.1 Standards Organizations

FEDERAL COMMUNICATIONS COMMISSION

In 1941, the Federal Communications Commission (FCC) made the first spectrum allocation and adopted the first technical standard for the transmission of black and white television in the U.S. [2]. Forty-six years later, in response to a joint petition filed by the Association of Maximum Service Telecasters, Inc. and 57 other broadcast organizations and companies, the Commission began a formal proceeding to replace that standard. On July 16, 1987, the Commission adopted a Notice of Inquiry in the matter of Advanced Television (ATV) Systems and their impact on the existing television broadcast service.

This notice went beyond the basic proposal to accommodate a new technology and raised the idea of delivering ATV by over-the-air terrestrial means, that is, by individual broadcast stations. This was a bold challenge considering that ATV research had concentrated on satellite and satellite-cable delivery systems, and many parties did not believe that terrestrial broadcast of high definition pictures was practical. Nonetheless, in order to make the benefits of an ATV service available to all viewers, the Commission was of the view that existing television licensees should be given the opportunity to provide that service.

Following the issuance of the Notice, the Commission established the Advisory Committee on ATV Service (ACATS). This group operates under a broad

charter for the purpose of advising the Commission on the facts and circumstances regarding ATV systems, and recommending policies, standards, and regulations that would facilitate the orderly and timely introduction of ATV services in the U.S. In its activities, it may consider all technical, economic, legal, and regulatory issues.

The organizational structure of the ACATS is illustrated in Figure 2.1. The

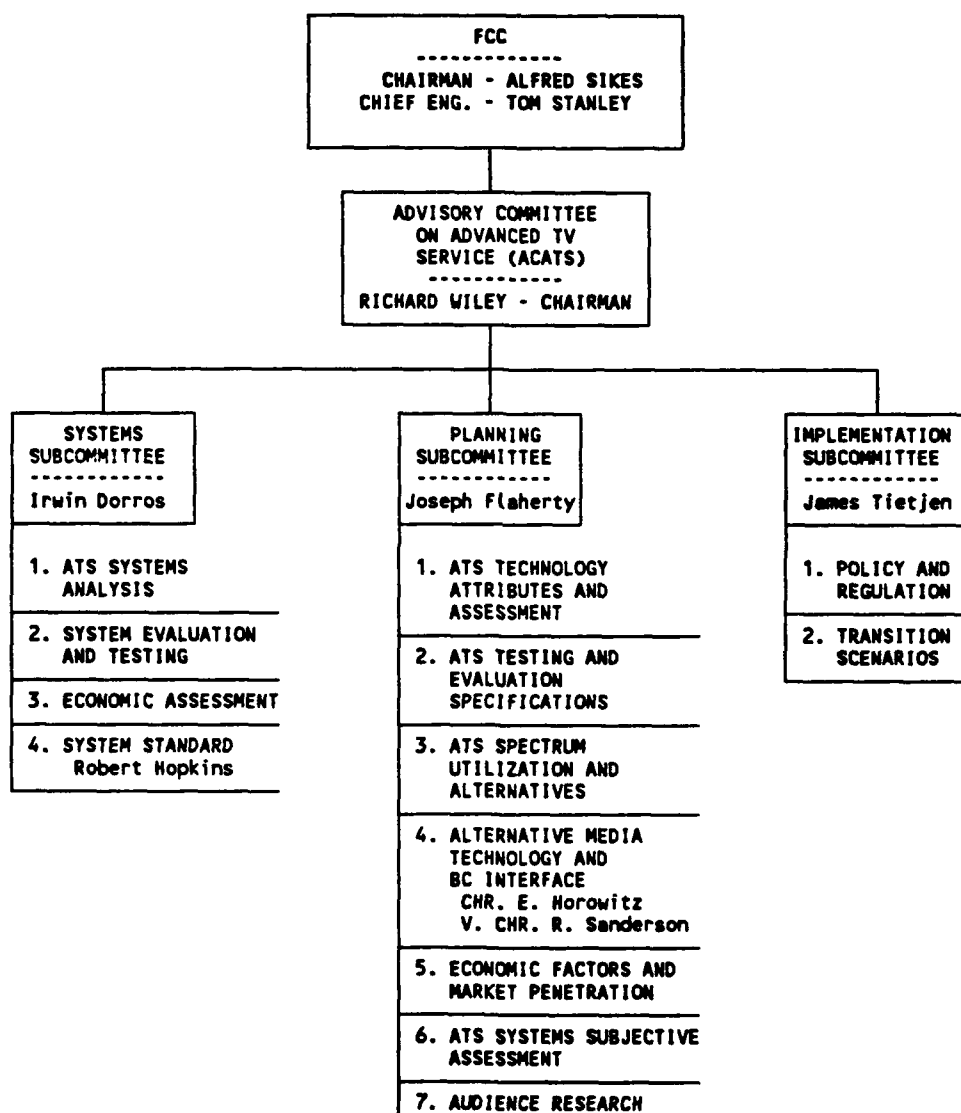


Figure 2.1
ACATS Organizational Structure

Planning Subcommittee has the task of defining the desirable characteristics of an ATV service and recommending planning factors for the establishment of that service. The Systems Subcommittee is hardware oriented; its task is to evaluate ATV systems under development, recommend a system (or specify the design of a system), and advise on the appropriate technical standards and spectrum requirements. Working Party 4 (Alternative Media Technology) is particularly important because its function is to insure that "inoperability" and "extensibility" are properly considered in the selection of the HDTV standard. Appendix B is a press release defining a recent activity of this working party.

The Implementation subcommittee deals with transition schemes and provides advice on official policies and regulations needed to implement an ATV service. At the present time, the members are concentrating on spectrum and implementation issues and the evaluation of six proposed ATV systems at the ATV Test Center. This test center is a facility established and funded by major elements of the television industry for the purpose of thoroughly measuring the attributes and performance of potential ATV systems under normal and adverse conditions. The results of the tests will be given to the Systems Subcommittee and will eventually be incorporated in the final report of the Advisory Committee to the Commission.

In a report adopted on August 24, 1990, the Commission made a critical technical decision and established a rough timetable for the completion of this proceeding and adoption of an ATV standard. The Commission stated that it intends to select a simulcast High Definition Television (HDTV) system for its ATV standard. *Simulcast* is a contraction of *simultaneous broadcast* and normally means the broadcast of one program over two channels in the same area at the same time. In this proceeding, simulcast has come to mean an independent ATV signal that can produce an advanced picture and be broadcast simultaneously with conventional television signals in the same area without causing interference. A licensee operating both a conventional and ATV station in the same market may or may not transmit the same program on both stations.

Consistent with this position, the Commission also said that it will give no further consideration to systems that use additional spectrum for special signals to augment conventional television transmissions. That approach relies on ATV receivers to produce an advanced, high-quality picture by combining the separate conventional and augmentation signals. Such a system would be less spectrum efficient and more difficult to implement than the independent, simulcast approach.

In light of the rapid changes that were taking place in this field, and not wishing to foreclose on any new ideas that might be emerging outside of the established process, the Commission indicated that it will conduct a special review early in 1992. It will be looking for any new ATV designs that offer important new benefits and are in a sufficiently concrete state of development to be considered with the existing systems then being studied by the Advisory Committee.

Finally, the Commission urged the Advisory Committee to complete its work and submit a final report with recommendations by Autumn 1992. That would enable the Commission to meet the goal of selecting an ATV standard by the middle of 1993.

ADVANCED TELEVISION SYSTEMS COMMITTEE

The Advanced Television Systems Committee (ATSC) was formed by the NAB, NTCA, IEEE, EIA, and SMPTE to further the work on HDTV. It is a free

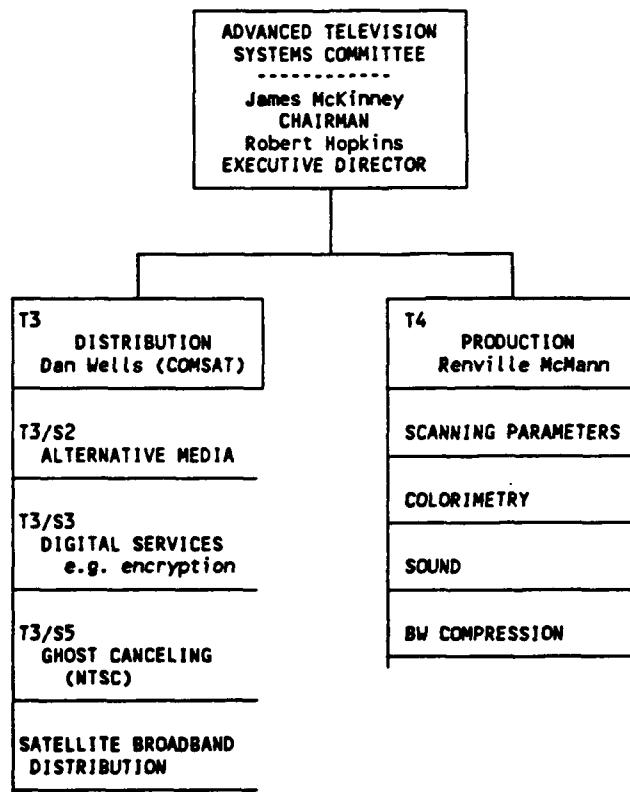


Figure 2.2
ATSC Organizational Structure

standing committee, and the organization is shown in Figure 2.2. Two subgroups representing key activity for telecommunications are (1) Alternative Media and (2) Bandwidth Compression.

SMPTE

The SMPTE organization has been a major contributor to the work in HDTV standards in recent years. One major contribution has been the development of the 240M standard (1125 lines, 60 fields/sec.) for an analog HDTV signal which is used extensively in TV production studios in the US and elsewhere. The key parameters for this standard are listed in Table 2-1.

Table 2-1
240M Key Parameters

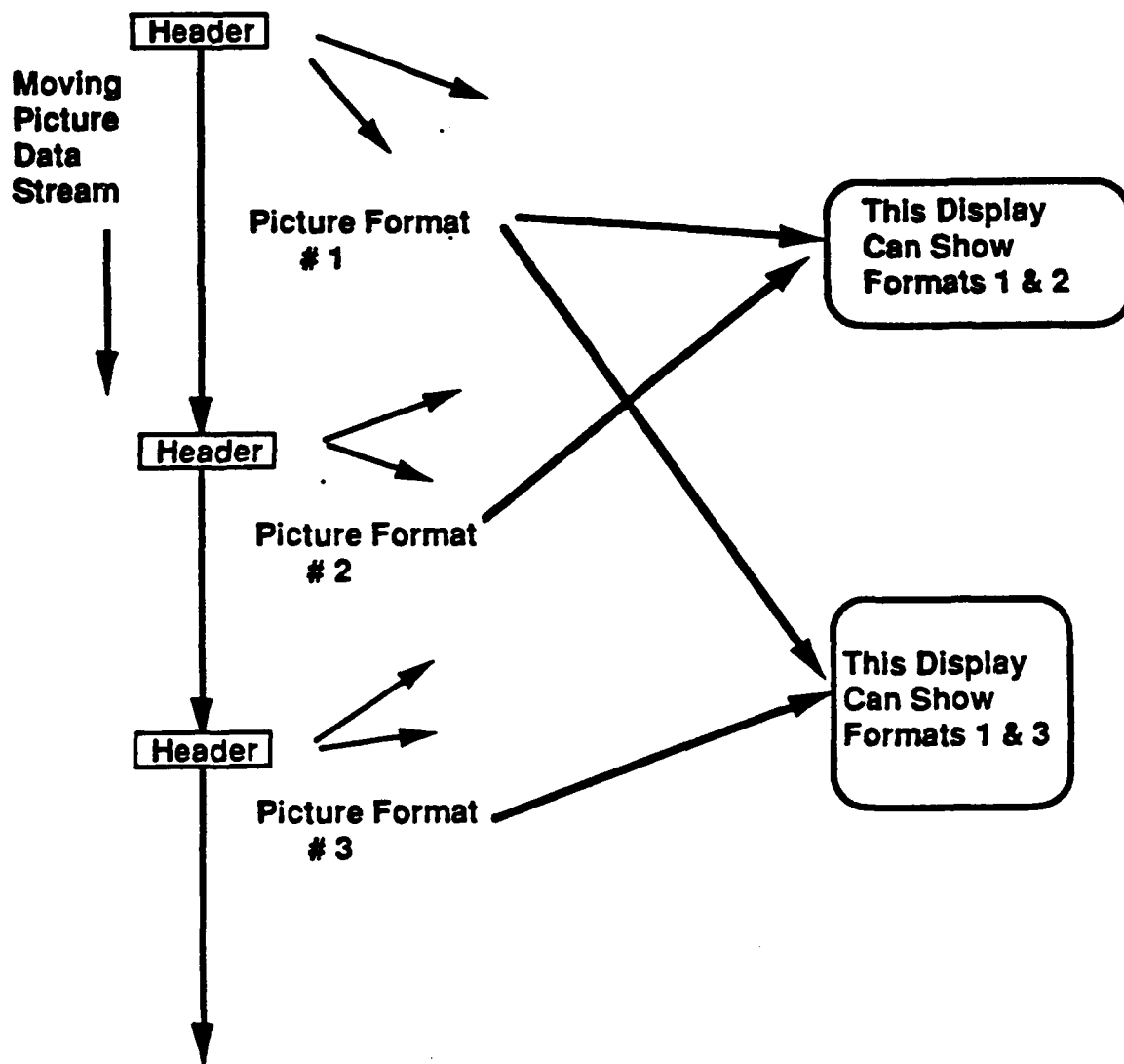
<u>IMAGE PARAMETERS</u>	<u>VALUE</u>
TOTAL NUMBER LINES	1125
NUMBER ACTIVE LINES	1035
FIELD RATE	60
INTERLACE	2:1
ASPECT RATIO	16:9
BANDWIDTH (MHz):	
LUMINANCE	Y = 30
COLOR DIFFERENCE #1	Pr = 15
COLOR DIFFERENCE #2	Pb = 15

Work is underway on the development of a digital version of 240M. In the digital system there will be 1920 samples/line.

The SMPTE is also working on a project to develop a Universal Header/Description for imagery which could have implications for HDTV. Specific goals for this project are listed below.

1. **Develop basis for new SMPTE standard for a header/descriptor suitable for digital images and other data.**
2. **Universality and interoperability -- usable across application and industries.**
3. **Extensibility -- accommodates essentially all technological advances with minimal impact on existing equipment.**
4. **Data Longevity -- permanent standard registration permits indefinite lifetime (centuries?).**
5. **Equipment Longevity -- header permits all equipment to ignore all unknown data packets efficiently.**
6. **Economy -- processing simplicity and data compactness.**

The goals of universality, extensibility, and interoperability are illustrated in Figure 2.3, 2.4, and 2.5 respectively.



**Figure 2.3. Universal Header:
Helpful in Interoperability and Extensibility**

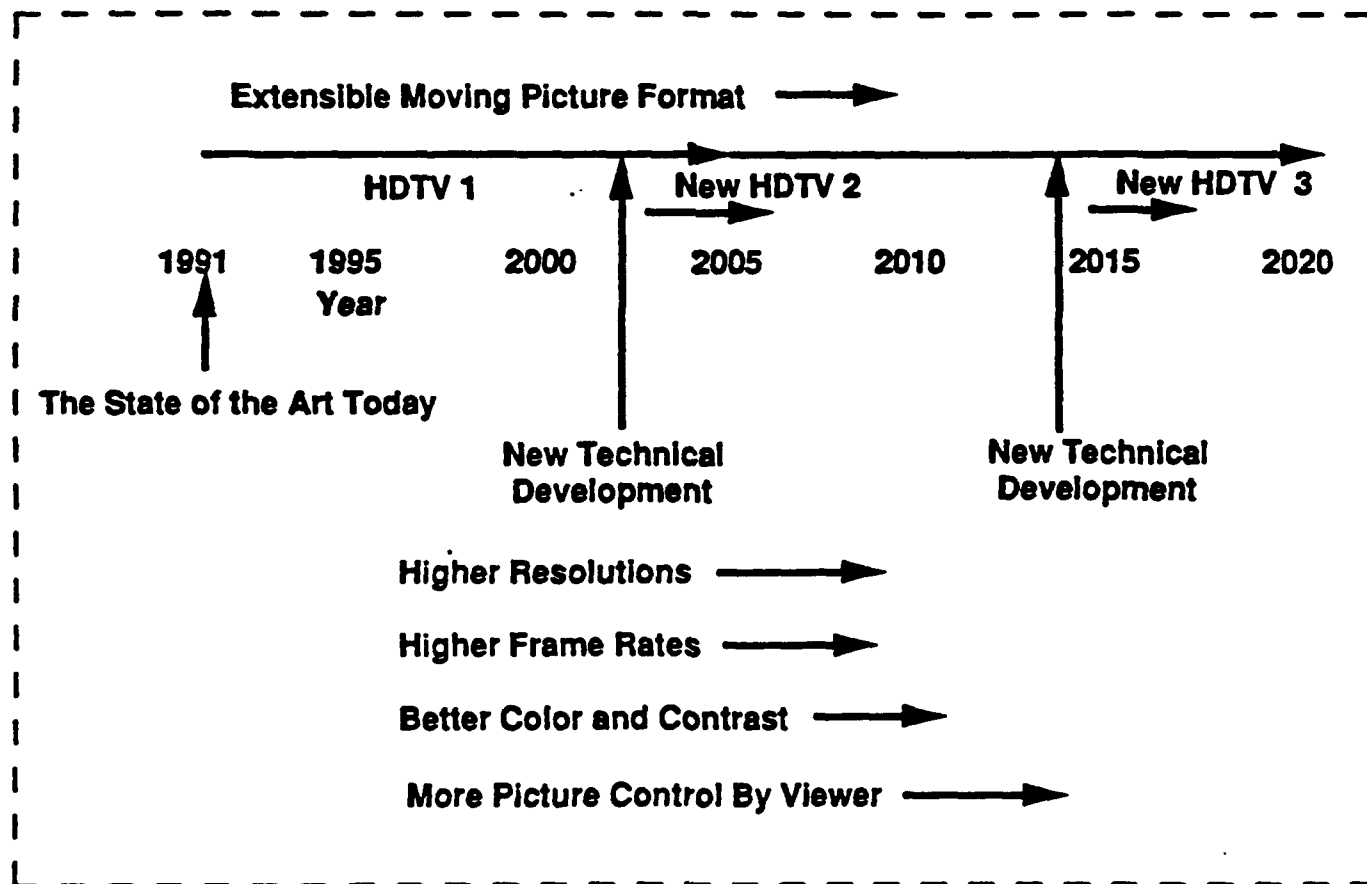


Figure 2.4. Extensibility: The Ability of a Format to Advance with Technology

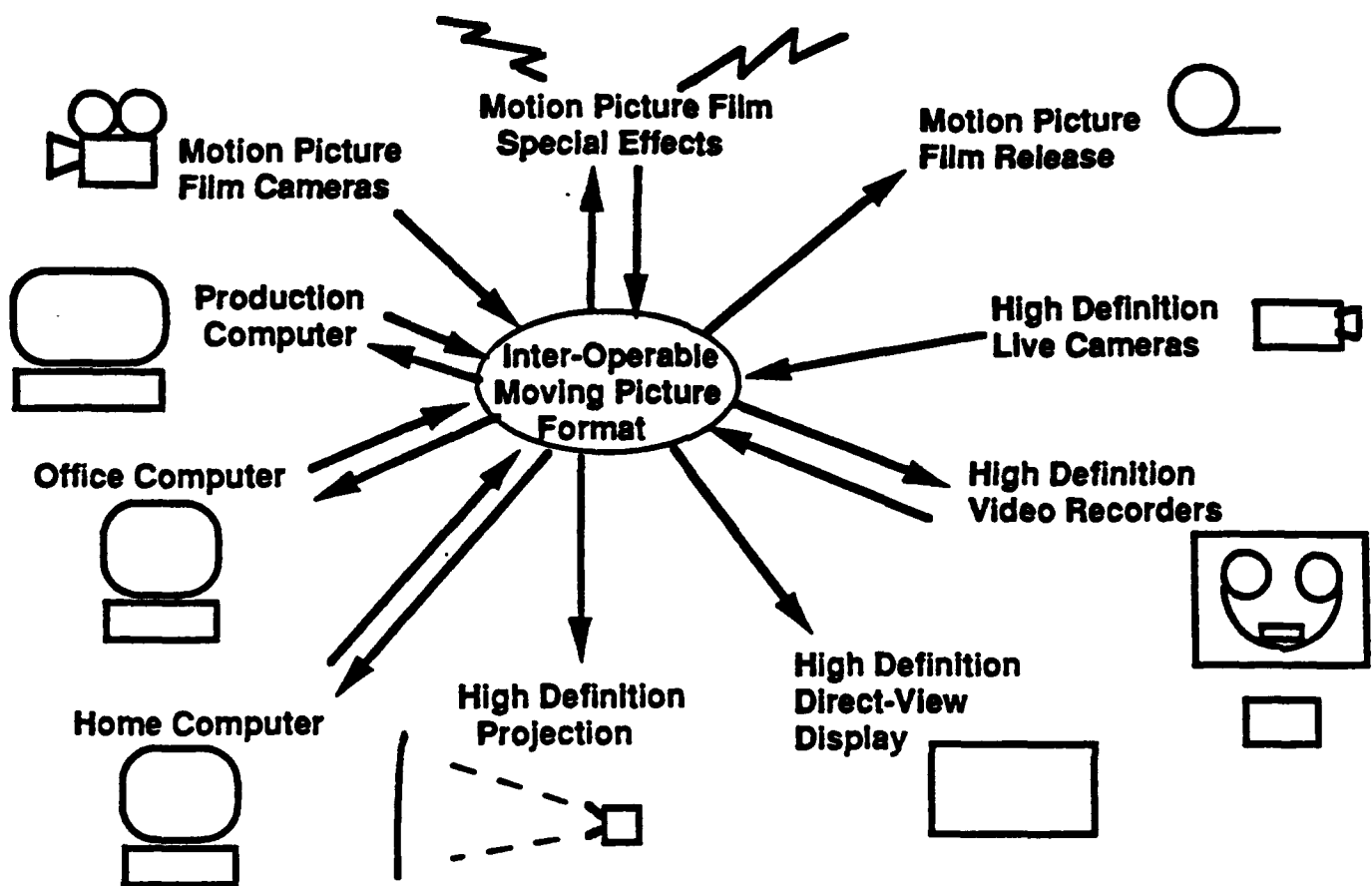


Figure 2.5. Inter-Operability: The Ability of Devices to Work Together

2.2.2 Proposed Digital HDTV Systems

Six systems have been proposed to the FCC/ACAT for the adoption as the new HDTV standard. The six are listed below along with the schedule for tests by the Advanced Television Test Center (ATTC) in Alexandria, VA.

<u>PROPONENT</u>	<u>NAME</u>	<u>TEST DATE</u>
Sarnoff Research		7/12 - 9/3 1991
NHK	Narrow Muse	9/1 - 10/24 1991
General Instrument, MIT	DigiCipher	11/14 - 1/7 1992
AT&T, Zenith	Digital Spectrum Compatible (DSC)	1/14 - 3/2 1992
NBC, Philips, Sarnoff, Thomson	Advanced Digital Television (MPEG + +)	3/9 - 4/22 1992
General Instrument, MIT	ATV Progressive	4/29 - 6/15 1992

The first two systems transmit the signal in an analog format while the last four are all-digital. There is a strong trend toward the adoption of one of the all-digital systems or some hybrid combination thereof. The technical characteristics of the four all-digital systems are summarized in Table 2-2. It is interesting to note that all systems employ the 8 x 8 DCT, interframe predictive coding, and motion compensation. This suggests a strong degree of interoperability with the P x 64 systems. Differentiating features include interlace, resolution, Variable Length Coding, layered coding, and packetized transmission.

The four digital systems are described in more detail in the sections below. This information was taken directly from the opening statements of the proponents documents submitted to the FCC/ACATS [3],[4],[5],[6].

NBC/PHILIPS/THOMSON/SARNOFF - Advanced Television Research Consortium,
Advanced Digital Television - Advanced Digital Television (ADTV) is a fully digital

Table 2-2
Comparison of the Four All-Digital
HDTV Proponents

PROPONENT(S)	GENERAL INSTRUMENT	AT&T,ZENITH	NBC,PHILIPS, SARNOFF, THOMSON	GEN. INST., MIT
SYSTEM NAME	DigiCipher	Digital Spectrum Compatible	Advanced Digital Television (MPEG + +)	ATV Progressive
SCAN LINES	1050	787.5	1050	787.5
INTERLACE	2:1	1:1	2:1	1:1
FRAME RATE	30	59.94	30	59.94
VISIBLE LINES (LUMINANCE)	960	720	960	720
PIXELS/LINE (LUMINANCE)	1408	1280	1440	1280
TYPE OF TRANSFORM	DCT 8 x 8 Pixels	DCT 8 x 8 Pixels	DCT 8 x 8 Pixels	DCT 8 x 8 Pixels
TYPE OF VLC	Two Dimensional Huffman	Depends on Quantization	Two Dimensional Huffman	
MOTION COMPENSATION	Yes	Yes	Yes	Yes
INTERFRAME PREDICTION	Yes	Yes	Yes	Yes
LAYERED CODING			Yes	
EXTENDED BLOCK STRUCTURE	Super Block (SB) 4 x 2 DCT; Macro - 11 x 1 SB			
INTRAFRAME UPDATE	0.37 sec.			
COMMUNICATION TECHNIQUE	FEC; Adaptive Equal, 16 QAM	RS FEC; Channel Equalization	Prioritized Cells QAM	DSB QM
COLOR SIGNALS	YUV	YUV	YBR	
PIXEL ASPECT RATIO		Square		Square

system that delivers high-definition television (HDTV) in a 6 MHz channel. The design of ADTV has been driven by the need of the terrestrial Broadcast, Cable Television, and Consumer Electronics industries to provide the American public with a quality simulcast HDTV service that is robust and reliable. To achieve its goals, ADTV has made significant improvements to proven digital compression and transmission techniques, and molded them into a single cohesive system.

There are three key elements in the ADTV system:

- First, ADTV's video compression, called MPEG ++, is based on a specific implementation of the MPEG (Moving Pictures Expert Group) compression approach. MPEG ++ upgrades the standard MPEG approach to HDTV performance level and incorporates a video data prioritization layer that allows the most important video data to be transmitted with the greatest reliability.
- Second, ADTV incorporates a Prioritized Data Transport (PDT) layer. PDT is a cell relay-based data transport layer that supports the prioritized delivery of video data, thus providing the feature of graceful service degradation under impaired channel conditions. PDT also offers service flexibility for a wide mixture of video, audio, and auxiliary data services, and compatibility with broadband ISDN (integrated services digital network).
- Third, ADTV applies spectral-shaping techniques to Quadrature Amplitude Modulation (QAM) to carefully minimize interference *from* and to any co-channel NTSC signals. The result is an extremely robust data transmission system, known as the Spectrally-Shaped QAM (SS-QAM).

Although submitted as a 1050-line system, ADTV is designed to provide flexible support of a wide range of services and future media formats. The initial hardware implementation will use the interlaced scan format for video source and display (1050/59.94/2:1), with a 16:9 aspect ratio and more than twice the NTSC resolution. Selection of this initial format was based on current camera and display technologies, and does not preclude a future adoption of other video formats, consistent with the evolution of studio equipment and production standards.

[MPEG is a committee within the International Standards Organization (ISO) that is currently working toward a standard for digital video storage applications.]

GENERAL INSTRUMENT - American Television Alliance, DigiCipher - General Instrument's DigiCipher System™ is an all digital HDTV system that can be transmitted over a single 6 MHz VHF or UHF channel. It provides full HDTV performance with virtually no visible transmission impairments due to noise, multipath, and interference. It offers high picture quality, while the complexity of the decoder is low. Furthermore, low transmitting power can be used, making it ideal for simulcast HDTV transmission using unused or taboo channels.

The DigiCipher HDTV System can also be used for cable and satellite transmission of HDTV. There is absolutely no satellite receive dish size penalty (compared to FM-NTSC) in the satellite delivery of DigiCipher HDTV. This is important not only for DBS, but for broadcast and cable since broadcast network and cable programming is typically delivered to affiliates via satellite.

To achieve the full HDTV performance in a single 6 MHz bandwidth, a highly efficient unique compression algorithm based on DCT transform coding is used. Through the extensive use of the computer simulation, the compression algorithm has been refined and optimized. Computer simulation results show excellent video quality for a variety of HDTV material. For error free transmission of the digital data, powerful error correction coding combined with adaptive equalization is used. At a carrier-to-noise ratio of above 19dB, essentially error-free reception can be achieved.

The DigiCipher HDTV System is an integrated system that can provide high definition digital video, CD-quality digital audio, data and text services over a single VHF or UHF channel. Bandwidth for conditional access capability that allows the encrypting of video, audio, and data services is also provided.

At the HDTV station, the encoder accepts one high definition video and four audio signals and transmits one 16-QAM modulated data stream. The control computer can supply program related information such as program name, remaining times, and program rating. At the consumer's home, the DigiCipher HDTV receiver receives the 16-QAM data stream and provides video, audio, data, and text to the subscriber. On screen display can be used to display the program related information.

The digital video encoder accepts YUV inputs with 16:9 aspect ratio and

1050-line interlaced (1050/2:1) at 59.94 field rate. The YUV signals are obtained from analog RGB inputs by RGB-to-YUV matrix, low pass filtering, and A/D conversion. The sampling frequency is 51.80 MHz for Y, U, and V. The digital video encoder implements the compression algorithm and generates video data stream. The digital audio encoder accepts four audio inputs and generates audio data stream. The data/text processor accepts four data channels at 9600 baud and generates a data stream. The control channel processor interfaces with the control computer and generates control data stream.

The multiplexer combines the various data streams into one data stream at 15.8 Mbits/sec. The FEC encoder adds error correction overhead bits and provides 19.42 Mbits/sec of data to the 16-QAM modulator. The symbol rate of the 16-QAM signal is 4.86 MHz.

The 16-QAM demodulator receives the IF signal from the VHF/UHF tuner and provides the demodulated data at 19.42 Mbits/sec. The demodulator has an adaptive equalizer to effectively combat multipath distortions common in VHF or UHF terrestrial transmission. The FEC decoder corrects virtually all random or burst errors and provides the error-free data to the Sync/Data selector. The Sync/Data Selector maintains overall synchronization and provides video, audio, data-text, and control data streams to appropriate processing blocks.

The control channel processor decodes the program related information. The user microprocessor receives commands from the remote control unit (RCU) and controls various functions of the decoder including the channel selection.

GENERAL INSTRUMENT/MIT - American Television Alliance, ATV-Progressive System - The ATVA-Progressive system is an all-digital advanced television system submitted by the Massachusetts Institute of Technology on behalf of the American Television Alliance. The ATVA-Progressive system has a number of important features. The system is channel-compatible and will fit within the channels now being used in terrestrial transmission. The system is very efficient in using the given channel spectrum. A high resolution video signal (720 x 1280 picture elements, 60 frames/sec, 16:9 aspect ratio) can be transmitted within a single 6 MHz channel. The system is resistant to channel impairments. Very high quality pictures are delivered to the home in the presence of substantial channel degradation including noise, ghosts, and frequency distortion. In part because of the high resistance to noise and interference, high picture quality is achieved at

low transmitter power, making feasible the use of taboo channels.

To achieve these features, the ATVA-Progressive system uses a variety of sophisticated modern signal processing methods. To achieve a high degree of data compression, motion compensation and transform/subband coding are used. In this method, a motion-compensated residual is represented using a transform/subband analysis, and only transform/subband coefficients with significant energy are transmitted. Motion compensation exploits the temporal redundancy of the video signal to reduce the energy of the signal to be encoded. Transform/subband analysis exploits the spatial redundancy of the motion-compensated residual signal. Elimination of low-energy transform/subband coefficients reduces the data rate requirement without significantly affecting the picture quality. The ATVA-Progressive system uses an all-digital format for terrestrial transmission. A single carrier with double-sideband suppressed-carrier quadrature modulation (DSB QM) is used.

ZENITH/AT&T - Digital Spectrum Compatible - The Digital Spectrum-Compatible HDTV System (DSC-HDTV) is very closely related to and is a natural extension of, the previous analog/digital (A/DSC-HDTV) system. A new effective system has been devised to reject NTSC co-channel interference into the DSC-HDTV channel which will result in an HDTV service area equal to that of an NTSC broadcast station while radiating at least 12 dB less power. The transmission signal now has a more noise-like character which has further reduced the visibility of interference into a NTSC channel. The simulcast feature in a 6 MHz band, the NTSC-like transmission signal timing and the low power are all retained. It is foreseen, as in the A/DSC-HDTV, that the current taboo channels can supply every current TV broadcast station with an extra channel for DSC-HDTV if certain interference conditions can be met.

The DSC-HDTV transmitter will not only operate with less power but also without aural transmitter, without a notch diplexer and with a smaller antenna. New algorithms have been developed to achieve the video compression needed for robust transmission without sacrificing image quality.

The reason for conversion from partially digital to all-digital transmission is found in the improved displayed picture throughout a station's service area. Digital transmission renders a picture essentially free of thermal noise or snow. Other picture impairments are avoided by forward error correction using a Reed-Solomon

code and by automatic ghost canceling/channel equalization.

There are a number of other all-digital benefits: more flexible and better bandwidth compression; noise free tape recording in studio and home; no noise, interference, or ghost accumulation from cascaded processing and/or cascaded transmission segments; synergy with other communication and computer equipment.

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3.0 TELECOMMUNICATIONS OVERVIEW

In general, video teleconferencing requires a high transmission bit rate relative to other services such as voice and data. For that reason, the availability of teleconferencing for the government community is dependent upon the availability of ubiquitous, inexpensive, switched, communication channels operating at high bit rates. The purpose of this section is to review the available telecommunication options which may be used for video teleconferencing today and in the future. The discussion, which focuses on the transmission of HDTV signals for teleconferencing, is divided into four parts: (1) teleconferencing communications today, (2) narrow band ISDN, (3) broadband ISDN, and (4) FTS-2000.

3.1 Teleconferencing Communications Today

Teleconferencing systems which are being installed today fall into two general categories -- narrowband (switched 56 Kbps), and wideband (384 Kbps, 768 Kbps, 1.544 mbps). Typical wideband services are implemented using either dedicated private T1 circuits or a switched service from AT&T (Accunet Reserved) or Sprint (Meeting Channel). In either case, a dedicated T1 type trunk circuit must be brought to the user's premises. In the case of a switched service, the user's access line is connected to the existing network implemented by AT&T (Accunet Reserved) or Sprint (Meeting Channel). As indicated above, the typical transmission bit rates employed over these wideband networks are 384 Kbps, 768 Kbps, or 1.544 mbps depending upon the quality of service required.

In the case of narrowband T/C systems, a typical T/C terminal would require two parallel switched 56 Kbps circuits be brought to the users premises. The terminal typically reallocates the total 112 Kbps capacity by assigning 32 Kbps to audio and 80 Kbps to video for example. The video quality at 80 Kbps is obviously reduced relative to that provided for wideband teleconferencing. Nevertheless, it has been found to be very effective for problem solving sessions and a wide range of teleconference applications.

One major stimulant to the teleconferencing business is the reduction in communication costs. In 1987, the cost of a switched 56 Kbps line was around \$75 per hour. Today, the cost is in the range of \$10 to \$25 per hour.

Representative rates for the wideband service, as of the date of this report, are provided in Table 3-1 for the Sprint Meeting Channel.

Both the narrowband and wideband teleconference network approaches described above are directly applicable to the transmission of HDTV signals for teleconferencing applications. For example, HDTV signals can be compressed by the CCITT standard algorithm for transmission at 1.544 mbps and provide very reasonable quality. In addition, it is possible to transmit HDTV signals over existing switched 56 Kbps networks for command and control applications (status board, computer graphics, etc.) where the data does not change rapidly.

The Department of Defense has established the Defense Communications Teleconference Network (DCTN) to provide teleconference services within that agency. This network provides switched service at a transmission bit rate of 1.544 mbps. Considerations are presently underway for reducing this bit rate to 768 or 384 Kbps.

3.2 Narrowband ISDN

3.2.1 Overview

ISDN is an emerging world-wide standard endorsed by the International Telephone and Telegraph consultative Committee (CCITT), the American National Standards Institute (ANSI), and virtually every other national telecommunications authority. An objective of ISDN is to provide the user with access to a variety of network services through a limited number of user-network interfaces. In most cases, this means access to voice, data and image via a single network connection. Inter-operability of switches and terminal equipment, another ISDN objective, will be assured by conformance to standards. Standardization will lead to significant reductions in equipment cost. ISDN will also contribute to reduced operating costs through its capability of continuously monitoring loop performance over an embedded maintenance channel.

In the information era of the 1990s, local telephone companies will provide a wide range of voice, data and image services. The demand for facsimile and emerging services such as video conferencing, high speed facsimile, and electronic mail are prime examples of the need for faster, more efficient communications services that will shape the telephone network of the 1990s. The major obstacle

Sprint Meeting Channel Rates

Location Key

A - Western United States
 B - Eastern United States
 C1 - Britain, Spain, France, Switzerland, Netherlands
 C2 - Germany, Norway, Finland, Sweden, Denmark,
 Belgium, Luxembourg, Austria, Italy, Greece
 D - Australia, Hong Kong, Singapore, Japan

E - Western Canada
 F - Eastern Canada
 G - South America
 H - U.S.S.R.
 I - Bermuda

Two-Way Rates, Within North America (Per Half Hour)

Chart I

	1.544 mbps	768 kbps	384 kbps	56/64 kbps
B to F	\$336	\$224	N/A	N/A
B to E	\$448	\$336	N/A	N/A
B to B	\$250	\$150	\$90	\$30
A to F	\$448	\$336	N/A	N/A
A to E	\$336	\$224	N/A	N/A
A to B	\$325	\$225	\$135	\$45
A to A	\$250	\$150	\$90	\$30

One-Way Rates, South America (Per Half Hour)

Chart II

	384 kbps
F to G	\$810
E to G	\$930
D to G	\$1,105
C2 to G	\$1,105
C1 to G	\$995
B to G	\$500
A to G	\$500

One-Way Rates, All Other International (Per Half Hour) *

Chart III

	1.544 mbps	768 kbps
D to F	\$1,036	\$924
D to E	\$924	\$812
C2 to F	\$896	\$784
C1 to F	\$784	\$672
C2 to E	\$1,008	\$896
C1 to E	\$896	\$784
C2 to D	\$1,200	\$1,100
C1 to D	\$1,100	\$1,000
B to I	\$325	\$225
B to H	\$1,080	\$982
B to D	\$840	\$602
B to C2	\$720	\$602
B to C1	\$620	\$416
A to I	\$325	\$225
A to H	\$1,080	\$982
A to D	\$840	\$602
A to C2	\$720	\$602
A to C1	\$620	\$412

* Minimum international transmission time is one hour. Foreign carrier charges are additional.

to providing these services by the present analog system is the local loop itself - it is difficult to accommodate improved service, such as Group IV facsimile, or adequately handle image and video transmission. The data service provided by the current public telephone network is limited by its analog architecture -- far too low for today's rapidly expanding information requirements. The Integrated Services Digital Network (ISDN) will transform the present telephone network into an end-to-end switched digital network providing 64 kbps, 128 kbps and higher bit rates.

The Regional Bell Holding/Operating Companies (RBOCs) project that by the end of 1994, they will have almost 65 million ISDN-capable access lines (57%) and 1832 ISDN-capable switches (19%). Independent Local Exchange carriers (LEC) are also deploying ISDN. LEC data is not available.

RBOCs are in the process of putting ISDN basic rate interface tariffs into place. For example, PACTEL recently issued basic rate ISDN tariffs of \$17.50 per month for B+D and \$29.50 per month for 2B+D. The federal Government strongly endorsed ISDN by awarding a 10-year, multibillion-dollar contract for FTS-2000 to AT&T and Sprint. Both proposals were based on ISDN architecture.

3.2.2 Benefits of ISDN

Digital telecommunications via the public telephone systems is opening a new world of information exchange. ISDN will provide the standard interface and medium for taking full advantage of digital communications. The modem will become obsolete.

Some of the benefits to be reaped by ISDN are:

- a. Decreased costs from dynamic bandwidth allocation which will replace trunk lines and T1;
- b. Faster data rates over the ISDN D-channel (i.e., packet switching);
- c. Improved access to corporate information from remote access;
- d. LAN-to-LAN applications operating at higher speeds without the requirement for dedicated lines;

- e. **Call-by-call service selection which will enable lower long distance charges;**
- f. **Call center ISDN applications that improve customer service through increased queuing efficiency and inventory control and superior network management due to call source/destination information.**

Narrowband ISDN uses existing network facilities, typified by the twisted pair local loop and 64 kbps digital switch. Basic rate service (2B + D) can be provided to a customer by replacing an analog telephone with an ISDN telephone and a network termination. The B channels (64 kbps) provide circuit-switched voice and data; the D channel (16 kbps) provides out-of-band signalling plus packet data. Primary rate ISDN consists of 23 B channels and one 64 kbps D channel for signalling. The PRI may also be allocated as high speed (H) channels at 384, 1472 or 1536 kbps.

Ubiquitous ISDN depends on the implementation of Common Channel Interoffice Signalling System Number 7 (SS7). SS7 is a packet switched signalling network operating in parallel with the traffic bearing network. The current, in-band, inter-switch signalling system (SS6) cannot preserve ISDN signalling between switches. Out-of-band signalling is necessary to connect ISDN switches and is essential to any service provided from a network database such as CLID and call-by-call routing. Custom Local Area Signalling Services (CLASS) for example, require SS7.

3.2.3 ISDN Capabilities/Applications

Basic Rate ISDN (BRI)

BRI provides a composite bandwidth of 144 kbps (2 x 64 kbps B channels + 1 x 16 kbps D channel) and full duplex transmission with time division multiplexing (TDM) into a single stream containing both user and signalling information.

Two interfaces are available with BRI, the S/T interface and the U interface. The 4-wire S/T interface, (CCITT I.430 and ANSI T1.605/1989 standards), is wired inside the premise and is manifested as a standard wall plug to link telephones, facsimiles and computers with the ISDN network. Each S/T interface

can accommodate a maximum of eight devices.

The 2-wire U interface connects local telephone lines to the customer premise (network termination) telephone lines bypassing the S interface.

Basic rate ISDN uses existing network facilities, typified by the twisted pair local loop and 64-kbps digital switch. Basic rate service (2B + D) can be provided to the customer by replacing an analog telephone with an ISDN telephone and a network termination. The B channels (64 kbps) provide circuit-switched voice and data and packet-switched data; the D channel (16 kbps) provides out-of-band signalling plus packet data. Basic rate ISDN allows applications such as Group IV facsimile, PC screen sharing, image transfer and limited motion videophone. (Primary rate (2B + D) provides up to 1.536 Mbps.)

ISDN basic rate can provide a number of new services with promising revenue potential, using the existing telephone plant. Examples include Group IV facsimile (fast facsimile service with laser printer quality), videotext (high-quality graphics), image transfer, telewriting (an electronic sketchpad) and video teleconferencing.

For geographically-dispersed, smaller business offices, ISDN could be an available alternative to private networks, especially when switches are connected through SS7. ISDN in combination with enhanced centrex services could allow local exchange carriers to compete effectively with PBX services and LANs.

An innovative new application capability is that of telemetry. The D channel packet switch capability can be used to transmit low-data-rate information while the phone is "on hook".

Primary Rate ISDN (PRI)

Primary rate ISDN (PRI) consists of 23 64-kbps B channels (30B channels in Europe) and one 64-kbps D channel for signalling. The PRI may also be allocated as high-speed (H) channels at 384, 1472 or 1536 kbps. The H channels are currently viewed as wideband circuits and could possibly be used to meet most demands of data communications users.

Primary rate channels can be used to multiplex lower data rate channels, provide high-data-rate wide area network connectivity and implement private branch exchange (PBX) connectivity.

PRI can also be used for LAN-to-LAN connectivity.

3.2.4 ISDN Status

ANSI standards for ISDN basic and primary rate transmission services are largely in place, and standards covering internetworking of ISDN and SS7, terminal adapters for non-ISDN equipment and supplementary services are nearing completion. At the same time, groups of the North American ISDN User's Forum are working to increase the effectiveness of these standards by resolving ambiguities that could lead to incompatibility among implementations. Major central office switch manufacturers (AT&T, Ericsson, Northern Telecom and Siemens) are now working to resolve incompatibility problems and expect to offer ANSI-Conformant N-ISDN switches by the end of 1991.

Presently, the majority of ISDN switch installations consist of a single switching node providing basic rate service within a local exchange area. A nationwide ISDN network will be possible only after the deployment of SS7, now under construction and partially deployed by the RBOCs and Inter-Exchange Carriers (IECs). So far, the IECs are further ahead, with the three largest (AT&T, MCI and US Sprint) already functional with SS7, and in the process of developing ISDN primary rate applications. MCI filed a PRI tariff that became effective on October 1, 1990. Service offerings include calling number identification and call-by-call service selection.

The RBOCs expect to have SS7 installed by 1992. (See description of National ISDN-1 below). Bellcore is coordinating nationwide SS7 interconnection which was targeted for completion by the end of 1992. Also, AT&T has announced plans for a "virtual SS7", scheduled to become available in 1991, that will give independent telephone companies full access to AT&T's SS7 network.

Figure 3.1 depicts the status of ISDN switch deployment by the Regional Bell Operating Companies in the United States for the period 1990 - 1994. When considering the RBOC strategy of deployment to metropolitan areas first and rural areas later, it can be seen from this graph that the majority of the access lines in the United States will be ISDN capable by 1994.

Figure 3.2 depicts the status of ISDN line deployment by the Regional Bell Operating Companies in the United States for the period 1990 - 1994. When considering the RBOC strategy of deployment to metropolitan areas first and rural areas later, it can be seen from this graph that the majority of the United States will be ISDN capable by 1994.

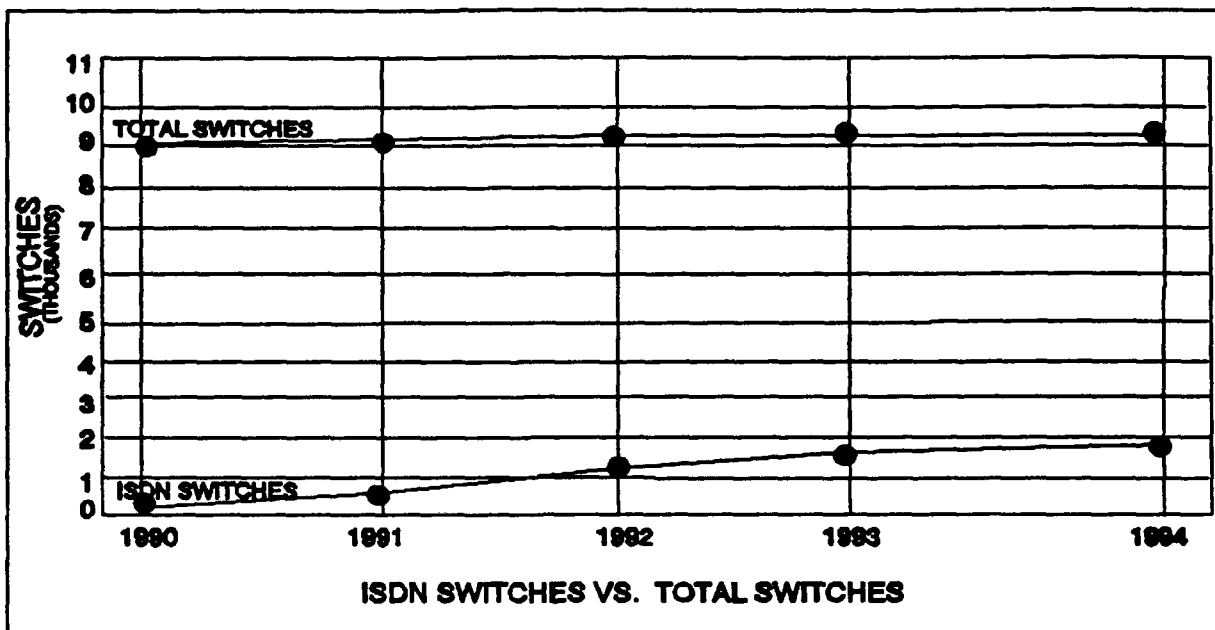


Figure 3.1. RBOC ISDN and non-ISDN Switches

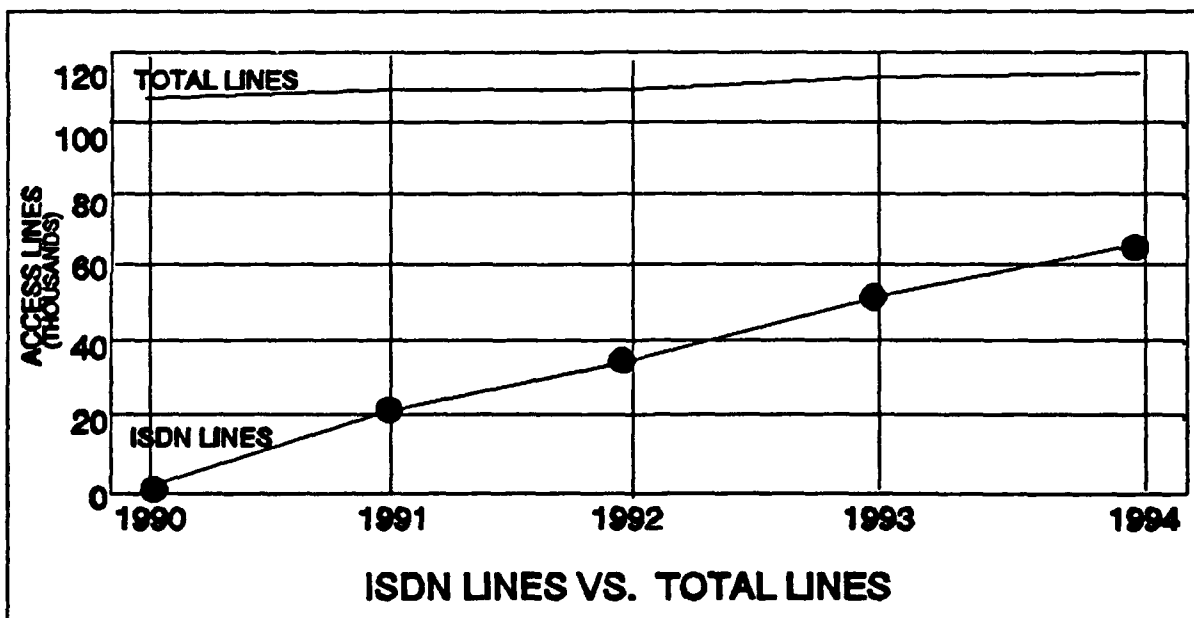


Figure 3.2. RBOC Total Access Lines vs. ISDN Access Lines

3.2.5 Offerings

BRI Offerings

All seven Regional Bell Operating companies have filed ISDN Basic Rate Interface (BRI) tariffs by the end of 1990.

The past two to three years have seen a number of ISDN trials and commercial offerings. This year will see an expansion in the number of ISDN lines and service offerings as general tariffs are formalized to replace special service contracts. PACTEL, for example, recently issued basic rate ISDN tariffs of \$17.50 per month for 1B+D and \$29.50 per month for 2B+D. As inter-switch SS7 connectivity becomes pervasive and equipment for applications becomes less expensive, ISDN will become more acceptable to the small business user.

PRI Offerings

RBOCs have begun to offer PRI access. Users are implementing and testing PBX access to support data transfer and special applications such as video conferencing and multi-media applications. Costs vary widely between exchange areas. Prices from \$300 to \$800 per month for PRI access have been found. Usage charges for long distance are additional.

3.2.6 ISDN Users Groups

Any major change to a very large, communications systems infrastructure such as the United States public telephone systems must overcome a tremendous inertia at rest. Further, a "catch 22" situation exists regarding the local and interexchange carriers making the financial investment to develop the necessary infrastructure prior to the availability of the market, the applications developers' investment prior to the availability and standardization of the infrastructure, and the development of the market (public need) without knowing the uses and services that the new system can provide.

ISDN User Groups are performing a significant service to the public. These groups are promoting user interest, informing application developers, and prodding the local exchange and interexchange carries to install the infrastructure.

ISDN Forum (NIU/Bellcore)

Working Together to Make National ISDN Happen. Organized by: Ameritech, Bell Atlantic, Bellcore, BellSouth, NYNEX, Pacific Bell and US West Communications. The ISDN Forum has been highly instrumental in providing ISDN implementation standardization within the RBOCs, the Independents, and the inter-exchange carriers. Bellcore has published several technical reports and special reports to support the standard implementation of ISDN in the United States. These reports are based on international and national standards and, in effect, provide specific guidance on subsets of the major standards for early, phased implementation. The objective is to get all carriers to implement common ISDN capabilities in step with each other in order to advance the pace of overall ISDN implementation. The ISDN Forum is the primary sponsor, along with NIU, of National ISDN-1 (described below).

North American ISDN Users Forum

The NIU Forum is sponsored by the United States National Institute of Standards and Technology (NIST). The precise relationship of the NIU Forum, NIST, and other business concerns is defined by the "cooperative Research and Development Agreement: The Consortium on ISDN Based Systems" which is available from the NIU Forum Secretariat. Although this forum focuses on the requirements of the ISDN users in North America, participation and membership is open to anyone.

The NIU Forum was established to achieve three principle objectives:

- a. promote an ISDN forum committed to providing users the opportunity to influence the developing ISDNs to reflect their needs;
- b. identify ISDN applications, develop implementation requirements, and facilitate their timely, harmonized, and interoperable introduction; and
- c. solicit, user, product provider, and service provider participation in this process.

The NIU Forum creates common North American functional and operational standards. A functional standard is limited to those functions that are executed to

achieve a capability that fulfills the applications requirements. An operational standard describes how this capability is provided to the application. Compliance with functional and operational standards enables applications to interwork and interoperate across communications links with conformant capability and behavior.

National ISDN

National ISDN-X is a program to describe an ISDN implementation plan over several years which will help harmonize service provider, application developer, systems integrators, and user initiatives.

National ISDN-1 is a set of service capabilities that have been developed for implementation in 1992 which will demonstrate the national connectivity of ISDN. ISDN-1 will enable the simultaneous voice and data service over the same telephone line. It is based on technical references (TRs) specified by Bellcore that will lay the groundwork for a national ISDN infrastructure. Bellcore issued the National ISDN-1 document SR-NWT-0001937, Issue 1, in February 1991 which describes the capabilities expected to be deployed on multiple vendor switches in 1992. Bellcore issued National ISDN-1 document SR-NWT-001953, issue 1, in June 1991, which provides Generic Guidelines for ISDN Terminal Equipment on Basic Access Interfaces.

National ISDN represents an effort by the telecommunications industry (primarily the RBOCs and Bellcore) and government agencies to implement a national network to demonstrate coast to coast ISDN connectivity. This demonstration would support various user tests of their ISDN systems on a national basis. The purpose is to publicize ISDN and to promote the development and implementation of user applications.

3.2.7 Conclusions

N-ISDN is being deployed rapidly. By the mid nineties N-ISDN will be ubiquitous in the more populated areas in North America. N-ISDN offers many opportunities for implementation of improved performance in communications systems, many new productivity enhancements, new applications and, in some cases, cost reductions.

3.3 Broadband ISDN

Integrated Service Data Networks (ISDN) are divided into two parts -- Narrowband and Broadband. N-ISDN operates at rates equal to, or less than, the primary rates (e.g. 1.544 mbps) while the B-ISDN operates at rates above the primary rates. The system parameters listed in Figure 3.3 show that a coding efficiency of 0.02 bits/pixel (a compression ratio of 400-to-1) would be required to transmit a typical HDTV signal at the primary rate. Since this level of compression is not economically achievable today, it is concluded that the B-ISDN will be required to transmit the HDTV signals to provide good quality for moving scenes. for this reason, it is particularly important to review and discuss the B-ISDN when considering the use of HDTV for teleconferencing.

■ Assumed HDTV Parameters

Luminance	pixels/line	1440
	lines/picture	960
Chroma	pixels/line	720
	lines/picture	480
Pixels/Picture	$= (1440 \times 960) + 2 (720 \times 480)$ $= 2.1 \times 10^6$	
Pixel Rate	$= 30 \times 2.1 \times 10^6 = 63 \times 10^6 \text{ pixels/sec}$	

■ Required Coding Efficiency for Transmission at Primary Rate

$$\frac{1.544 \times 10^6 \text{ bits/sec}}{63 \times 10^6 \text{ pixels/sec}} = 0.02 \text{ bits/pixel}$$

Figure 3.3
Transmission Bit Rate Requirements
for HDTV Signals

One backbone structure for the B-ISDN consists of a new Synchronous Digital Hierarchy (SDH) defined in CCITT Recommendations G.707, G.708, and G.709 [1-3]. These recommendations are concerned with bit rates for the synchronous digital hierarchy, details of the resulting network node interface, and a synchronous network multiplexing structure. Three digital Synchronous Transfer Modes (STM) have been so far specified:

- STM-1 : 155.52 Mb/s
- STM-4 : 622.08 Mb/s
- STM-16: 2488.32 Mb/s

The North American version of the standards uses a basic transport module of 51.84 Mb/s (Synchronous Transport Signal-Level 1, STS-1) and is described in a recent article by Ballart and Ching [4].

In addition to optical interfaces for the above synchronous bit rates (SONET - Synchronous Optical Network), CCITT Working Party XVIII/7 has agreed to standardize an STM-1 electrical interface for inclusion in Recommendation G.703. SONET is defined in the U.S. by the ANSI standards as listed below.

T1.105-1988	Optical Interface Rates & Formats
T1.106-1988	Optical Interface Specifications: Single-mode
T1.105-1990a	Addendum (in voting process)
T1.xxx-1990	Optical Interface Specification: Short Reach (in voting process)
T1.yyy-1990	SONET OAM&P specification (drafted, voting to begin 11/90)

Figure 3.4 compares the new synchronous digital hierarchy with the existing digital hierarchy.

The general structure and service capabilities of the ISDN are defined in a total of 77 CCITT Recommendations I.110 through I.605 which are contained in the CCITT Blue Books. Within this set, Recommendation I.121 provides the basic description of the B-ISDN. Much of the material in the following paragraphs is derived from this Recommendation.

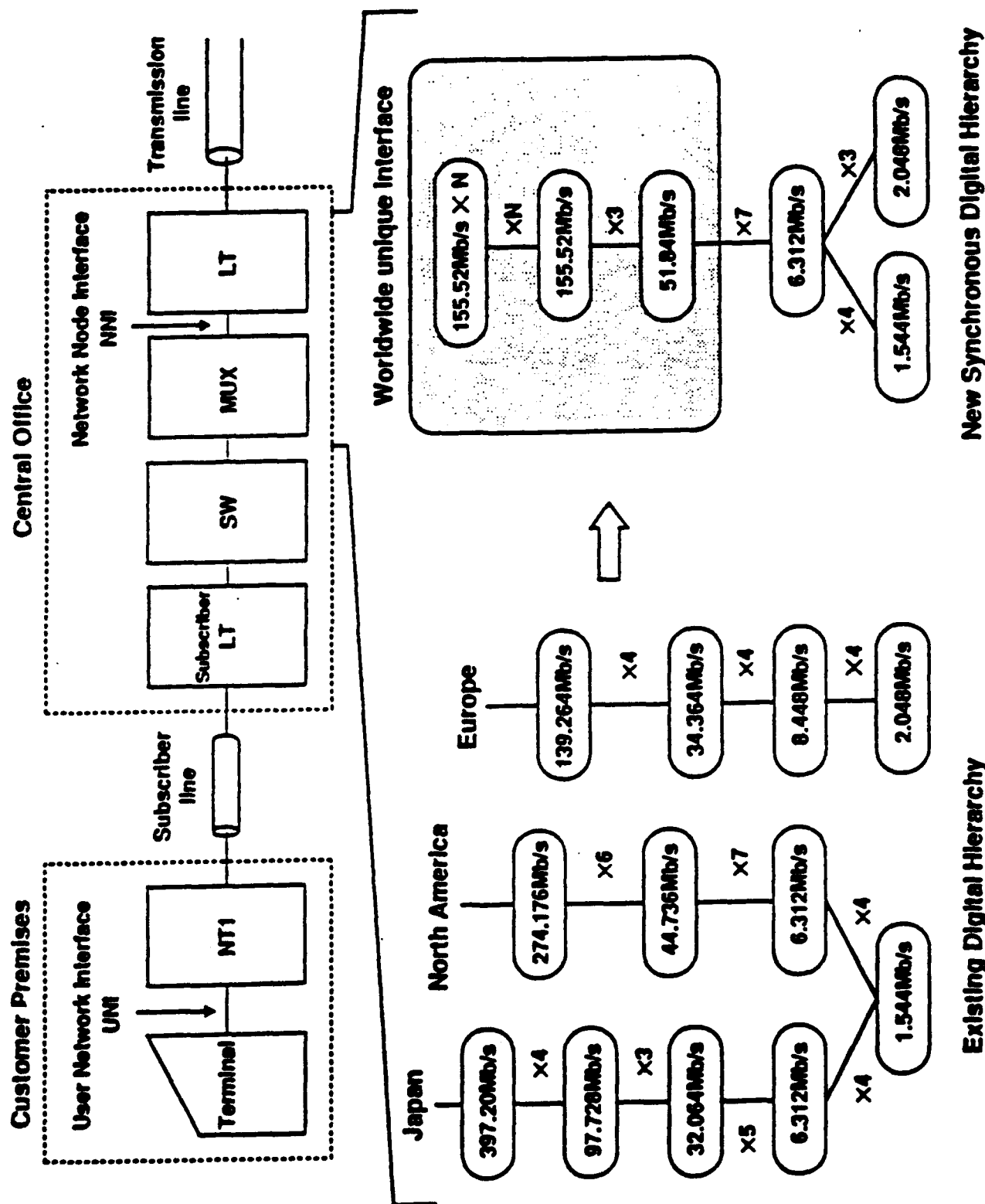


Figure 3.4. Comparison of New Synchronous Digital Hierarchy to Existing Digital Hierarchy

Principles of B-ISDN

The main feature of the ISDN concept is the support of a wide range of audio, video and data applications in the same network. A key element of service integration for an ISDN is the provision of a range of services using a limited set of connection types and multipurpose user-network interfaces. B-ISDNs support both switched and non-switched connections. Connections in a B-ISDN support both circuit-mode and packet-mode services.

B-ISDN will contain intelligence for the purpose of providing service features, maintenance and network management functions. A layered structure will be used for the specification of the access protocol to a B-ISDN.

Target Transfer Mode

Asynchronous transfer mode (ATM) is the target transfer mode solution for implementing a B-ISDN. It will influence the standardization of digital hierarchies and multiplexing structures, switching and interfaces for broadband signals.

ATM concerns a specific packet-oriented transfer mode using the asynchronous time division multiplexing technique: the multiplexed information flow is organized in fixed size blocks, called cells. A cell consists of a user information field and a header; the primary role of the header is to identify cells belonging to the same virtual channel on an asynchronous time division multiplex. Cells are assigned on demand, depending on the source activity and the available resources. Cell sequence integrity on a virtual channel is preserved by the ATM layer.

ATM is a connection-oriented technique. Header values are assigned to each section of a connection when required and released when no longer needed. The connections identified by the headers remain unchanged during the lifetime of a call. Signalling and user information are carried on separate virtual channels. ATM will offer a flexible transfer capability common to all services, including connectionless services.

Evolution Steps

B-ISDN will be based on the concepts developed for ISDN and may evolve by progressively incorporating additional functions and services (e.g. high quality video applications). The deployment of B-ISDN may require a period of time extending over one or more decades. Thus, arrangements must be developed for

the interworking of services on B-ISDN and services on other networks. In the early stages of the evolution of B-ISDN, some interim user-network arrangements {e.g. combinations of synchronous transfer mode (STM) and ATM techniques} may need to be adopted in certain countries to facilitate early penetration of digital service capabilities.

Service Aspects of B-ISDN

Two main service categories have been identified: interactive services and distribution services. The interactive services are subdivided into three classes of services, viz., the conversational services, the messaging services, and the retrieval services. The distribution services are represented by the class of distribution services without user individual presentation control and the class of distribution services with user individual presentation control.

Conversational Services

Conversational services in general provide the means for bidirectional dialogue communication with real-time (no store-and-forward) end-to-end information transfer from user to user or between user and host (e.g. for data processing). The flow of the user information may be bidirectional symmetric, bidirectional asymmetric and in some specific cases (e.g. such as video surveillance), the flow of information may be unidirectional. The information is generated by the sending user or users, and is dedicated to one or more individual communication partners at the receiving site. Examples of broadband conversational services are videotelephony, video conference and high speed data transmission.

Messaging Services

Messaging services offer user-to-user communication between individual users via storage units with store-and-forward, mailbox and/or message handling (e.g. information editing, processing and conversion) functions. Examples of broadband messaging services are message handling services and mail services for moving pictures (films), high resolution images and audio information.

Retrieval Services

The user of retrieval services can retrieve information stored in information

centers and in general provided for public use. This information will be sent to the user on his demand only. The information can be retrieved on an individual basis. Moreover, the time at which an information sequence is to start is under the control of the user. Examples are broadband retrieval services for film, high resolution image, audio information, and archive information.

Distribution Services Without User Individual Presentation Control

These services include broadcast services. They provide a continuous flow of information which is distributed from a central source to an unlimited number of authorized receivers connected to the network. The user can access this flow of information without the ability to determine at which instant the distribution of a string of information will be started. The user cannot control the start and order of the presentation of the broadcast information. Depending on the point of time of the user's access, the information will not be presented from its beginning. Examples are broadcast services for television and audio-programs.

Distribution Services With User Individual Presentation Control

Services of this class also distribute information from a central source to a large number of users. However, the information is provided as a sequence of information entities (e.g. frames) with cyclical repetition. So, the user has the ability of individual access to the cyclical distributed information and can control the start and order of presentation. Due to the cyclical repetition, the information entities selected by the user will always be presented from its beginning. One example of such a service is full channel broadcast videography.

Information Flows

Video service information can be characterized in many ways, including:

- The direction of information flow: video services may be bidirectional, e.g. videotelephony and videoconference, or essentially unidirectional, e.g. video distribution services for business and entertainment.
- The symmetry of information flow: messaging, retrieval and distribution services are characterized by asymmetrical information flows.
- The origin of the source material: how video signals enter the

network (e.g. direct from camera, from storage media, via satellite or other delivery mechanisms) can also provide a means of characterizing service information flows.

Switch Functionality

The switching infrastructure of a Broadband ISDN will probably support a variety of switched services. For example,

- Point-to-point switching, e.g. videotelephony
- Point-to-multipoint: bi-directional, e.g. videoconferencing, multimedia conferencing
- Point-to-multipoint unidirectional, e.g. broadcast distribution services, switched distribution services.
- Multipoint-to-point, e.g. televoting services, transfer of charging information to service providers.
- General and selective broadcast switching, e.g. switched and unswitched distribution services.

An ATM based B-ISDN will have the ability to support one-to-many call distribution through multicast switching and should result in significant simplification in multimedia terminal design and could also support the flexible deployment of multimedia and multipoint bridges.

Storage Requirements

Many video and image services may require the network to store some aspect of the service. For example, video and messaging services will require network resident storage facilities, as will many forms of database. The efficiency and economy of such services is strongly influenced by the ease of coding and decoding images for storage and the characteristics of the storage medium itself, e.g. compact disk, videotape, magnetic disk.

Storage related issues:

- Efficient image compression/decompression algorithms to achieve cost efficient use of available storage capacity.
- Image coding times which reflect the nature of the intended service, e.g. for deferred delivery services, such as video mail, acceptable

coding solutions may take the form of slow, but efficient coding and fast decoding.

- Others for further study.

Service Bit Rates

The B-ISDN will be based on ATM techniques which are well suited to supporting source traffic which is time varying. The establishment of virtual connections which involve the transfer of information only when required will mean that the resources of the network can be closely matched to the needs of the source traffic.

The following areas relating to service bit rates are still under investigation:

Maximum Service Bit Rate supported by the 155.52 Mbit/s Interface

The transfer capability of the 155.52 Mbit/s interface provides a payload capacity of 149.76 Mbit/s. Allowing for ATM cell overheads, the maximum service bit rate which can be supported is equal to or less than 135.631 Mbit/s. The actual maximum service bit rate is for further study. The actual value depends on the capacity required by signalling, operations and maintenance and ATM adaptation overheads. The granularity of the actual service bit rates offered by networks is for further study.

Maximum Service Bit Rate Supported by the 622.08 Mbit/s Interface

Agreement has been reached that the B-UNI (1.413, 1.432) at 622 Mbit/s should be based on a single ATM stream rather than a multiplexed structure of four 155 Mbit/s streams. The overhead structure of the UNI and the NNI at 622 Mbit/s is common and this results in an ATM cell transfer capacity of 599.040 Mbit/s.

Bit Rate Assurances

Parameters for constant and variable bit rates agreed at call set up time are assured for the duration of the call. No assurance is given concerning additional traffic above the level initially negotiated.

The Specification of Service Bit Rate Parameters

- specification of the bit rate of CBR services requires only a single parameter

- specification of the service bit rate of VBR services is expected to require multiple parametersaverage rates, burst length, etc.
- the time period over which the rate is specified is influenced by service timing and buffering constraints and the capabilities of the network interface.
- for CBR services there are two options to be addressed in specifying service bit rates;
 - the service bit rate is the actual bit rate, i.e. a user must generate traffic at the exact bit rate.
 - the service bit rate means a ceiling to be supported by the B-ISDN. A user can generate traffic at any bit rate less than the service bit rate. The user of a CBR service in this manner is for further study.
- the parameters selected must be of a form and nature which allows the network to exercise the option of statistically multiplexing VBR services, where appropriate, in a manner which does not violate the agreed QOS.

ATM Performance Parameters

- Cell Loss Ratio
- Cell Error Ratio
- Cell Transfer Delay
- Mean Cell Transfer Delay
- Cell Delay Variation

Work Plan

The timeframes of this work plan indicate when Recommendations are expected. The contents should reflect the development status of B-ISDN network capability and the stages in development of Recommendations relevant to video coding for the B-ISDN. Appendix C is a list of the international standards organizations which contribute to telecommunication work for B-ISDN and therefore HDTV.

1990-92

- **Communicative video services support on B-ISDN (point-to-point)**
- **Initial services**
 - **videotelephony and videoconferencing (using limited point-to-multipoint connections)**
 - **lower rate retrieval**
 - **point-to-point switched**
- **Initial emphasis on CBR mode**
- **Assessment of VBR vs CBR advantages/disadvantages**
- **AAL Type 1 SAR and CS to be specified**
- **Degree of compatibility with existing coding standards required**
- **SGXVIII Recommendation on network aspects of Integrated Video Services on B-ISDN**
- **Studies on layered coding for service interworking in IVS**
- **Traffic control and resource management**

1992-94

- **Lower rate (?) distribution services**
- **Support of VBR mode emerging, with use of CLP**
- **AAL Type 2 SAR specified**
- **AAL Type 2 CS specified**
- **Digital television distribution**
- **Enhanced multimedia video services**
- **Enhanced signalling and control capabilities**
- **Recommendations relating to service interworking in IVS and B-ISDN**
- **Architecture aspects**
- **Video coding for ATM**
 - **Communicative**
 - **Distribution**

1994 +

- **Wide range of interactive and distribution services**
- **Digital HDTV services - broadcast and non-broadcast applications**

3.4 FTS-2000

The FTS-2000 Project was initiated by the General Services Administration in the mid-1980's to provide standard common carrier telephone network services to Government agencies that would satisfy other than specialized information resources needs. Contracts were awarded in 1988 to two common carriers, AT&T and SPRINT for FTS-2000 services. The networks operated by these carriers are referred to as Network A and Network B, respectively. Individual agencies are assigned to either network in accordance with a plan to maintain a specific percentage traffic distribution between the networks at the time of the 1988 award.

The services provided to users have been expanded. Currently Switched Data Services (SDS) and Dedicated Transmission Services (DTS) are available. Compressed Video Transmission Services (CVTS) is now being provided on the networks. Its inclusion requires specific requests for the service from user agencies.

Earlier this year, ISDN Services were introduced in FTS-2000. It is currently available on Network A and should be available on Network B shortly. Currently, only Circuit switched voice and data services are available. ISDN Packet Service is not available on either network yet. It will be incorporated when user agencies request it.

ISDN PRI service can be provided to any user agency with an ISDN compatible PBX via a T-1 carrier to the PBX. Currently approved PBXs include AT&T Definity 75 and 85 models, NTI SL-1 and SL-100 models. The NEC 2400 PBX was expected to be approved in October 1991.

FTS-2000 does not provide ISDN BRI services. In addition, the full scope of ISDN services are not available through FTS-2000 even to users with fully capable ISDN PBX/switch facilities. Since Caller ID data is not passed along in FTS-2000, those inter-site capabilities that utilize this data will not be possible.

REFERENCES

- [1] CCITT Recommendation G.707, "Synchronous Digital Hierarchy Bit Rates," Study Group XVIII Plenary Assembly, Melbourne, Nov. 1988.
- [2] CCITT Recommendation G.708, "Network Node Interface for the Synchronous Digital Hierarchy," Study Group XVIII Plenary Assembly, Melbourne, Nov. 1988.
- [3] CCITT Recommendation G.709, "Synchronous Multiplexing Structure," Study Group XVIII Plenary Assembly, Melbourne, Nov. 1988.
- [4] R. Ballart and Y.C. Ching, "SONET: Now Its the Standard Optical Network," *IEEE Commun. Mag.*, pp.8-15, Mar. 1989.

4.0 VIDEO COMPRESSION

This section is provided to give some perspective of the many methods available for video compression at this time and to indicate directions which various applications are taking. Although much theory on video compression was developed prior to the 1970's and several systems built for special non-commercial applications it has been the last 15 to 20 years that have seen video compression become practical for commercial and indeed soon for consumer applications. A combination of new methods and groups of methods, and of practical implementations in the form of very large scale integrated circuits has made this possible.

One of the greatest positive influences to low-cost video compression and hence many new applications was not a technical advance but rather the CCITT (Consultative Committee for International Telephone and Telegraphy) H.261 technical Recommendation which has given Integrated Circuit vendors some technical specifications on which to base silicon designs. The advent of I.C.'s has in turn encouraged video compression product vendors incentive to design a variety of products not necessarily limited to the H.261 standard per se. It is readily apparent that several of these same I.C.'s can be used for higher resolution imagery systems than the originally intended H.261 standard.

Within the last 18 months video compression has been proposed for broadcast purposes and even High Definition Television transmission which is scheduled to become a consumer reality within compression at reasonable cost is now here the technology is still advancing at a rapid pace. This creates a dilemma specifically in the HDTV arena where a U.S. terrestrial transmission standard is scheduled to be finalized in 1993. Some argue that a standard should be delayed while newer, and maybe better, video compression technology is evaluated. A feeling is generally held that any standard adopted will be used for tens of years in the future and should therefore be the best one possible. The background for this, of course, is that the present NTSC standard was adopted 37 years ago.

Video compression is a form of data compression and will be discussed here in several forms. One distinction to be kept in mind is that of lossless versus lossy compression. Lossless compression requires the delivery of a decoded output picture which is exactly the same as the input to the encoder (typically 8 bits/pixels). Lossy compression permits the output to be degraded relative to the

input. Usually only lossless compression is considered for data handled by computers in that any loss of data can't be tolerated. As a result, the lossless compression methods usually yield compression factors of perhaps two to four. The nature of the data, however, has a great deal to do with what compression factor is possible.

Although lossless compression can be performed upon video data to achieve compression factors of two to four, much higher compression ratios are possible if some loss is permitted. A key objective for lossy coding techniques is to degrade the output picture in a way that is not visible to the eye. The majority of practical video compression involves lossy compression methods, often in combination with other lossless methods. Examples of lossless compression methods are Run-length coding and Variable Length (usually Huffman) coding. Lossy methods involving a single image at a time include Transform Coding, Vector Quantization, Subband Coding and others. Motion Image lossy methods include predictive coding, motion compensated prediction, transform techniques and skipped or interpolated frames.

Figure 4.1 is a functional block diagram of a generic video compression system. It shows that the compressor is typically implemented in three sequential functions -- pixel transformation, quantization, and variable length coding. Basically, the Pixel Transformer performs measurements on the input uncompressed pixels. It computes prediction error, or transform coefficients, converts the pixels to bit planes, filters the signal into sub-bands, correlates the pixels with pre-stored VQ patterns or correlates the pixels with the image itself (Fractals). The Pixel Transformer performs measurements on a single frame (intraframe) or on multiple frames (interframe). Predictive and bit plane coding techniques can be either lossless or lossy. The other techniques are fundamentally lossy.

The output of the typical Pixel Transformation process is quite accurate -- 8-bit to 12-bit precision. In lossless compressors, this accuracy is preserved in the quantization process. In lossy systems, the quantizer reduces the accuracy of the transformer output in a way that is as acceptable to the eye as possible. The quantization can be employed on either a scalar or vector basis.

The final step in the compression process is to encode the quantizer output with a Variable Length Code (sometimes called Entropy coding). VLC is a technique whereby each event is assigned a code that may have a different number of bits. In order to obtain compression, short codes are assigned to

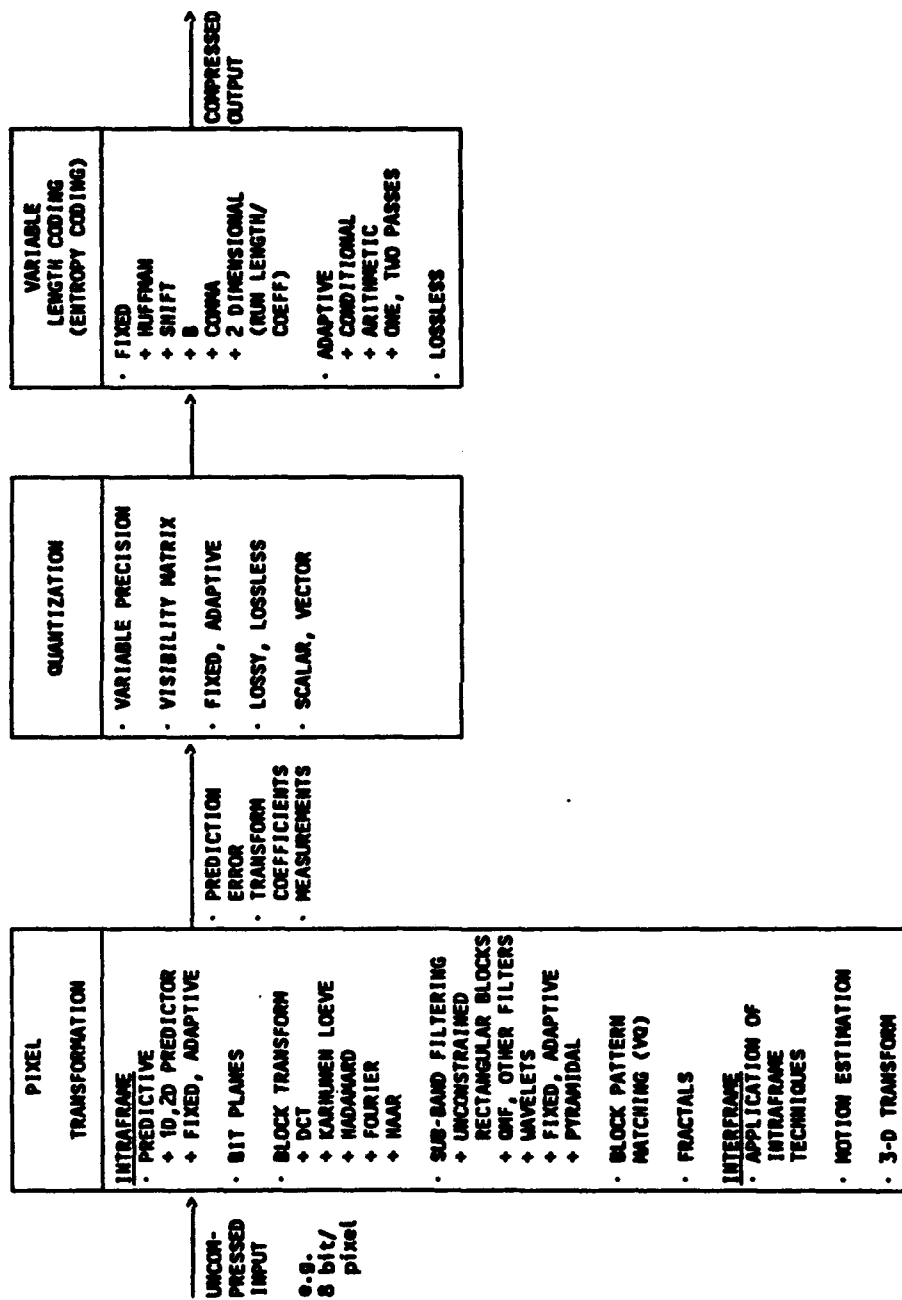


Figure 4.1. Functional Block Diagram of a Generic Video Compression System

frequently occurring events, and long codes are assigned to infrequent events. The expectation is that the average code length will be less than the fixed code length that would otherwise be required. If all events are equally likely, or nearly so, then VLC will not provide compression.

A major advantage of VLC is that it does not degrade the signal quality in any way (lossless). That is, the reconstituted signal will exactly match the input signal, so that if the signal is adequately described by a series of events, using VLC's to communicate them to the decoder will not change the events. Therefore, the system is transparent to the VLC used.

The disadvantage of VLC's is that they only provide compression in an average sense. Therefore, sometimes the code could be longer for a specific section of signal. This characteristic gives rise to the need for a buffer to match the variable rate of bit generation with the fixed bit rate of the communication channel, and a control strategy to prevent long-term overflows or underflows of the buffer.

The various Intraframe methods which include Transforms, Pyramidal, Subband and Vector Quantization, are not mutually exclusive; however, there are important distinctions which set them apart into separate classes. Fractals, on the other hand, are distinctly separate from the others. The decomposition by Fractals involves expressing an image in terms of mathematical functions unique to that image whereas the transforms, etc. use predetermined basis functions (different bases for different transforms but independent of the images) resulting in numerical coefficient outputs to represent images.

The two principal data reduction methods used in the transform coefficient domain are coefficient quantization and zonal coding, with adaptive methods yielding better compression performance than static ones. The methods can be used singly or in combination.

The video compression problem is usually that of a video source which generates data, after taking into account some data reduction methods such as frame to frame differencing, over a wide range of bits per second, and adjusting this rate to fit into a constant rate communication channel. The best compressors adapt to the adjustment process to minimize the perceived distortion when the source data rate is high.

This report does not review all of the techniques in detail since that review is readily available in the literature (see bibliography). However, aspects of the

various methods and differences between them will be discussed which are important to present and future HDTV applications.

The purpose of an Intraframe transform is to compact the video signal into a rather small region of the transform domain and to generate a set of resulting output coefficients which has a "frequency" characteristic to them. The first characteristic results in having to actually transmit a number of outputs much smaller than there were video picture elements at the input. The second aspect permits the operations of "quantization" and "zonal coding".

Quantization refers to the process of expressing the value of a coefficient to be transmitted with a smaller number of bits than were generated by the transformer, or were taken to represent the original picture elements themselves. When the quantization level is high, very few bits are used for each coefficient and a certain amount of image distortion results. However, this distortion is small compared with the distortion which would result had the quantization been applied to the picture elements themselves. Also, a large number of coefficients, when quantized, have amplitudes of zero and need not be transmitted at all.

Zonal coding refers to the process of selecting a specific zone, based upon frequency in the transform domain, of the resulting transform coefficients for transmission. The zone always includes at least the lowest frequency component and usually several other low frequency ones. The high frequency ones, which are discarded when additional compression is required to fit the data onto a fixed rate transmission channel, correspond to finer picture detail which may not be perceived as missing when the video signal is experiencing considerable motion.

Both of these processes are "lossy" in that the decoder cannot recreate the coefficients exactly as they were prior to these operations. In contrast the transform operation itself is exactly reversible, at least mathematically, prior to any quantization operation. The idea is to introduce a small, controlled amount of distortion in trade for a vastly reduced amount of data to transmit.

For Interframe data reduction, a Selective Replenishment process can be used either with or without Motion Compensation to take advantage of the redundancy often present in motion imagery. The idea here is that it requires far less data to be transmitted to instruct a decoder to use previously transmitted pieces of imagery to form a new image than to repeat the transmission of the image data. Another form of this temporal redundancy reduction is through use of a transform operating on data from several adjacent images frames. For stationary

images, the transform data contains only the single lowest frequency component (in the temporal direction). A three dimensional transform can also be employed to simultaneously perform Intra- and Inter- frame compression.

Transforms including the Haar, Walsh-Hadamard and the Fourier predate 1970 and were used for image compression in the sixties and seventies. The Haar and Walsh- have the desirable property that multiplications are not explicitly required, but on the other hand tend to show artifacts more quickly when subjected to quantization and zonal coding. The Discrete Cosine Transform (DCT) was developed/invented in 1974 and is by all measures the most optimal of the blocked transforms. A disadvantage relative to the two other cited transforms is that its calculation requires non-trivial multiplications. The first "fast" DCT algorithm was developed in 1977 which minimized the number of calculations and multiplications. Several other "fast" DCT algorithms followed later. By 1989, a two-dimensional DCT was available on an integrated circuit chip obviating most all previous calculation advantages of the other blocked transforms.

The 1980's saw the development of other methods of Intraframe transform-like compression which are enough different from the DCT to warrant significant attention. These are the Pyramidal transforms, the subband coding functions, and the image vector quantizer. Since these have been gaining attention more recently, some additional details will herein be included. Four descriptors are useful to show the differences and similarities between each of them and from the DCT:

1. Blocked vs. non-blocked basis function
2. Hierarchical arrangement of basis functions
3. The transformation preserves the number of data points
4. Exactly Invertible

A blocked transformation is one where all the basis functions and hence output coefficients are derived from the same sub-set of picture elements in a block whereas a non-blocked transformation employs basis functions which have staggered footprints wherein coefficients may partially overlap each other but do not necessarily completely overlap.

A hierarchical arrangement of basis functions means that there are multiple groups of the basis functions differing in the size of the footprint on the pictured

elements. Often the different groups are related in footprint size to each other by factors of four.

The smallest footprint coefficients correspond to the fine detail of the image and the largest footprint coefficients to the more global changes in the image. There are often four to six groups in the hierarchy of a practical configuration. Viewed in the frequency domain, the smallest footprint coefficients occupy the upper one-half (in each of the two spatial dimensions separately) of the frequency space. The next lower frequency group occupies one-quarter of frequency space in each direction. The last group contains the "d.c." terms and the lowest frequency components. A point is often made concerning the human visual system that spatial detail is perceived with progressively less acuity on a logarithmic basis. This implies that coefficient quantization can be performed progressively more roughly on each higher frequency group. Zonal coding is also practical in these hierarchical organizations.

Preservation of the number of data points means that the transformation process does not create any more coefficients than there were original picture elements.

The exactly invertible process is one which, in the absence of quantization, mathematically re-creates the same picture element amplitudes by application of the inverse transform to the coefficients resulting from the forward transform.

Table 4-1 below shows the types of image compression versus the four

Table 4-1. Types of Image Compression vs. Four Characteristics

TYPE	BLOCKED	HIERARCHICAL	PRESERVES PTS.	INVERTIBLE
DCT ET AL	Y	N	Y	Y
SUBBAND 1	N*	N	Y	Y
SUBBAND 2	N	Y	Y	Y
LAPLACIAN PYRAMID	N	Y	N	Y
VECTOR QUANT. 1	Y	N	NA	N
VECTOR QUANT. 2	Y	Y	NA	N

- * The first type of subband coding can be made to be a blocked in the extreme case. This technically allows the DCT (and other blocked transforms) to be called special cases of Subband Coding.

aforementioned characteristics.

The Vector Quantization (VQ) method differs from the others in that it is not mathematically invertible. It is useful to view this method as a transform method wherein the basis functions of the transform are optimized by computer examination of a training set of images. The method of coefficient quantization is simultaneously built into the VQ method by predetermining the number of coefficient amplitudes to be generated by the processor. This view of VQ places it in the same context as the methods with the exception of the built-in quantization procedure.

VQ may also be viewed more closely to how it is actually used in practice. The VQ operation (the non-hierarchical version) is performed on a small block, perhaps 8 to 20 picture elements, such that a Look-up Table of all the pre-stored patterns of these elements is investigated to determine the one which most closely fits the inciting picture block. The address in a computer-like memory for this pattern is transmitted to the decoder which uses the address in a duplicate Look-up Table to obtain the pattern of picture elements.

In a practical system, the number of addresses may be 64,000. The VQ system is determined as "unsymmetrical" in that the process at the encoder for determining the best pattern is far more complex than the decoder process of simply looking-up the result in the table. Its use favors the one-encoder, many-decoders scenario. VQ systems and the combination of VQ in hybrid systems with Interframe methods is less advanced. A major proponent of VQ, Scientific Atlanta, has designed and built a near-broadcast quality system which demonstrates its commercial viability, especially in satellite systems which are more prone to data errors than most terrestrial systems.

Many configurations employing VQ can be implemented just as many configurations of transform-based systems are implemented, although the adaptive and variable quantization aspects of the standardized H.261 system have not been realized. VQ can be organized in a hierarchical fashion which offers a different quantization possible for each group of the hierarchy corresponding to the reduced visual acuity for smaller image detail. Of course, the DCT can (and has) been organized in that way in one configuration by the PictureTel Corp.

Subband Coding is a mathematical transformation process which converts the image into a set of output coefficients which can then be operated upon with quantization and Zonal coding techniques, similar to the processing of DCT

coefficients. It is easiest to view Subband Coding as a bank of filters. Although it is also possible to view the DCT as a bank of equal bandwidth filters, Subband Coding does not have the equal bandwidth constraint as does the DCT. In the above Table, Subband Coding 1 has equal bandwidth filters whereas Subband Coding 2 has unequal bandwidth filters. The latter type is organized in a hierarchy such that each succeeding filter has one half the bandwidth (in a given spatial direction) that of the previous filter. The term "Wavelet Coding" has also been used for subband coding due to the appearance of the finite duration signal reconstructed from each of the (non-zero) filter coefficients.

What has made Subband Coding practical for images has been the evolution of the Perfect Reconstruction Filter whose technology generally evolved during the 1980's. These remarkable filters provide selective filtering in the frequency domain similar to one low-pass filter and a succession of bandpass filters. Although the frequency responses of these filters have overlapping skirts, the information carried by each of the filters may be exactly reconstructed without any interference from data in other filter bands.

The shape of the filters can be selected within constraints imposed by the perfect reconstruction criteria. It is desirable to make these filters have mutually exclusive frequency regions as much as possible for video compression to limit the occurrence of "aliasing". Sharpness of each filter in the bank tends to limit the effects of quantization and zonal coding to just the band(s) in which employed. Considering the desirability of using different amounts of these techniques in the different bands, it is desirable to have the filters as selective as possible. However, higher selectivity is paid for by an increased number of filter taps and hence implementation expense.

Pyramidal Transforms are generally considered to be a class of Perfect Reconstruction filters in which the number of "coefficient" outputs exceeds the number of input picture elements. This of course is a source of inefficiency in that additional data must be transmitted or stored. There is, however, some performance advantage to these filters in that aliasing is easier to control. In general, the so-called pyramidal transforms have not been used for commercial video compressors.

As early as 1979, it was argued forcefully that the development of quality video compression systems ought to proceed, not with the development of newer transform methods, but with energies directed toward adaptive methods of

coefficient processing and presumably other cooperative processing methods. That advice was not universally taken, as evidenced by all of the above transform work, although it can be argued that the advice is still valid. A first concern in 1979 was that the computation of the DCT was accomplished only by expensive hardware requiring high speed digital multipliers. The above mentioned "Pyramid Transform" (the subband coding method) was designed which required no multiplications, other than trivial ones by powers of two accomplished by binary shifts. With the advent of the integrated circuit DCT this advantage was virtually erased.

Other reasons have been more recently proposed for using subband transformations rather than the DCT which are not necessarily of a compression performance nature. The argument is that a hierarchical structure of image data permits a decoder to decode a lower resolution image (or set of images to create motion) without having to decode the incoming data relating to the unneeded higher resolution components. One of three or four different resolutions could be decoded. Another argument suggests this is a solution looking for a problem, but the hierarchical coding methods nonetheless offer this feature. One suggested use of this technique is for progressive transmission of still images over low bandwidth transmission media. A browse feature is offered to a remote viewer who receives an image progressively, not in the window-shade presentation of earlier freeze-frame systems, but with low frequency information first and quickly followed by progressively more detail until the image reaches full resolution. The viewer can see something of the entire image almost immediately and either continue to receive the image if of interest, or abort that image in favor of a different one.

The subband image coding advocates are at this time are high spirited and quite inventive and starting to influence development of some commercial systems. It is likely that one of the HDTV terrestrial transmission proponents (M.I.T.) will include in its proposal a subband coding intraframe compression approach by the time its hardware system is tested in mid-1992.

Decomposition of images by Fractals is the newest method of all. A transformation method was defined and patented within the last two years which permits an arbitrary image to be coded mathematically with Fractal equations. The equations are used by the decoder to provide imagery whose resolution can be as high as desired. This system is very unsymmetrical in that the encoding procedure is far more complex than the decoding procedure. The technology is claimed to

provide significantly greater compression efficiency than any other techniques. (This of course has previously been claimed for other methods as they were being developed.) A commercial system for satellite delivery of motion pictures using Fractal technology is now in the planning phase and may become a reality in a few years time.

Many video compression systems are hybrid in nature in that one method is employed for Intraframe compression and a second method employed for Interframe compression. The one which has gained most acceptance in that it is the foundation for the CCITT H.261 and the MPEG standards uses Motion Compensation for the Interframe compression as a prediction of the new frame. The difference between the actual new image and the predicted one, on a picture element to picture element basis, is then transform coded and sent to the decoder. The Transform used is the DCT. This system also has a second mode wherein only Intraframe transform coding is used. This mode is employed where no meaningful image data exists from the previous frame to be used for the new image. This system was suggested in 1979 (by Jain and Jain) and was popularized by inclusion in the above mentioned standards. A system of this type could be built with other types of Intraframe coding substituting for the DCT. This same architecture is the foundation for at least four of the six proposals for an HDTV terrestrial transmission standard presently before the Federal Communications Commission.

It is clear now that video compression is a major topic of both academic and commercial interest for HDTV. The number of hardware (and software) vendors of various systems is increasing rapidly and several permutations of various methods are being designed and fielded. Pioneering in this field is still very much alive and will timely remain so for a few more years before a small number of systems gain pre-eminence, and long enduring standards are in place. The development activity of new systems for commercial broadcast purposes is mostly in the U.S. where possible delivery channels are not as closely regulated by governmental bodies. It is difficult to predict the future usage and relative status of the several now competing compression methods. It is, however, comforting to have so many attractive video compression methods to apply to video storage and transmission problems and the integrated circuit technology available to implement them in reasonable cost hardware and software.

Layered Coding

In the future, there will be requirements for a wide range of video applications (e.g. communicative real-time video, video retrieval or store-and-forward video, distributive services), using signal formats covering a wide range from videophone resolution to HDTV and at a range of qualities for any given signal format. It is generally acknowledged that it is important to achieve a maximum degree of "interworking" of these services.

"Simulcast" is one approach to achieve interworking which is accomplished by the simultaneous transmission of the same video signal in different formats. "Layered coding" is a more promising means of facilitating interworking between video services.

In layered coding systems, a hierarchical representation of the video signal is defined. Coders transmit a baseband signal (e.g. H.261) which provides a basic quality service. Incremental signals, which can be used along with the baseband to recover a high quality signal (e.g. HDTV), are also transmitted. Receiving terminals utilize the baseband and an appropriate number of incremental signals to recover the video signal to the quality which they are capable of displaying. Transmitting terminals provide the number of signals which is commensurate with their input signal quality. Note that "embedded bitstream" and "syntactic extension" techniques are also versions of layered coding.

Layered coding seems suited to the widest application range. "Flexible layering" in which any number of layers can be used in any particular application, appears to provide broad interworking capability with few restrictions. It is recognized that to provide easy interworking or conversions between services, and to use common display components on a terminal device intended to access multiple video services, the definition of a family of picture formats is important. Picture formats represent an important area that will influence video coding and it is being studied actively in the SGXV Experts Group.

In ATM system, a layered coding approach appears attractive as a means of minimizing the effect of cell loss, particularly if it occurs in bursts. This requires separation of the video information into high and low priority components and appropriate setting of the cell loss priority (CLP) bit in the ATM cell header.

Some advantages of a layered coding system are:

- Ease of extension to future video systems (e.g. from HDTV to super HDTV with e.g. 4000 x 4000 pixels).
- Ease of compatibility among various video services.

Some of the disadvantages of layered coding are:

- Possible reduced coding efficiency when compared to non-layered systems if motion compensation is applied.
- Possible increase of the complexity of encoder and decoder.

Constant Bit Rate (CBR) and Variable Bit Rate (VBR) Coding

Restrictions of traditional circuit switched networks have meant that all commercial digital video codecs operate at a constant bit rate, despite the inherently varying information content of a motion video sequence (being dependent on changing image complexity, degree of motion, frequency of scene changes, etc.). The internally varying rate in these codecs is smoothed by buffering and dynamic control of codec parameters (sensitivity, quantizer stepsize, etc.) to ensure that the buffer neither empties nor overflows. Such codecs operate in a fixed rate, but variable quality, mode.

ATM Networks will support VBR coded video, allowing the transmitted bit rate to dynamically reflect the information content of the changing video signal, limited by the maximum channel capacity and parameters agreed with the network management system.

A VBR codec can therefore (usually) maintain a fixed quality, variable bit rate mode of operation. The possible advantages of this are:

- Because data is not transmitted when the information content is low, and because high rates are only used when necessary, VBR codecs are expected to deliver a given overall quality at a lower average rate than a CBR codec.
- The reduction in buffer size and easing of constraints on rate control means that there could be savings in codec complexity and cost.

- **Reduced buffering may mean that end-to-end delays will be reduced; this is an important consideration for communicative services such as videotelephony and videoconferencing.**

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This book contains excellent descriptions and comparative information on blocked transforms as well as background theory, human visual response, "fast" algorithms for implementation and error and noise effects.

2. William K. Pratt, Editor, "Image Transmission Techniques", Academic Press, London, New York, 1979.

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This paper describes original work on the Motion Compensated Prediction, DCT Transform Coded video compression system. The whole December 1981 of this periodical is a special issue on Picture Communication Systems.

4. J.W. Woods, Editor, "Subband Image Coding", Kluwer Academic Publishers, Boston, 1991.

This book describes the theory and several applications of single and motion image subband coding, written in part by the people who have developed the methods. This volume contains in one place much technical information on the subject collected from the last decade's worth of work.

5. G. Held and T. Marshall, "Data Compression", 2nd Ed. John Wiley and Sons, Chichester, New York, 1983, 1987, 1988.

This volume describes several data compression techniques, some of which are used for video coding. It provides background information not usually covered in papers dedicated to video coding.

6. Several issues of the IEEE Proceedings dedicated to Image Processing, Pattern Recognition and Image Coding:

July, 1972	October, 1972
April, 1975	May, 1979
March, 1980	May, 1991
April, 1985	April, 1990

Also, a Special Section on Multi-Dimensional Signal Processing in Images, from the IEEE Trans. on Acoustic, Speech and Signal Processing.

5.0 SYSTEM ANALYSIS

In Sections 2.0, 3.0, and 4.0, we examined the three basic elements of a comprehensive HDTV system for teleconferencing within the U.S. Government -- the HDTV signal, communication issues, and data compression. The purpose of this task is to integrate the results of these three tasks into complete end-to-end communications services and to examine these potential services as they relate to the government requirements. Examples of government needs which could be fulfilled by point-to-point telecommunication HDTV services are:

- teleconferencing
- videophone
- command and control function
- training
- multimedia

Each of these potential government applications is discussed below.

5.1 Teleconferencing

Table 5-1 is a summary of the formats for the standard and proposed HDTV signals. The pixel rate for these standards has been computed and listed in the next to last column. Transmission bit rates are also estimated for typical services and listed in the last column. The video transmission bit rates for the five systems are listed below.

Videophone H.261	80 Kbps (two 56 Kbps channels less 32 Kbps for audio)
Teleconferencing H.261	320 Kbps (384 Kbps channel less 64 Kbps for audio)
MPEG 1	1.5 mbps
MPEG 2	5 mbps
Typical HDTV proposal	20 mbps

Table 5-1. Formats of Standards and Proposed TV Signals

TV STANDARD/ PROPOSAL	SCANNING FORMAT			DIGITAL FORMAT			PIXELS/ FRAME	PIXELS/ SEC. MBPS	TRANS- MISSION BIT RATE
	SCAN LINES	INTER- LACE	FRAME RATE	LUMINANCE VISIBLE LINES	CHROMA VISIBLE LINES	PIXELS/ LINE			
ANALOG STANDARDS									
1. NTSC	525	2:1	30	480	480	---	---	---	---
2. PAL	625	2:1	50	576	576	---	---	---	---
3. MUSE (JAPAN)	1125	2:1	30	1080	---	---	---	---	---
4. EUREKA (EUROPE)	1250	2:1	50	1000	---	---	---	---	---
5. SMPTE 240M	1125	2:1	30	1035	1920	960			
DIGITAL STANDARDS									
6. CCIR 601	---	---	30	480	480	360	.69M	20.7	
7. CCITT H.261 (CIF)	---	1:1	submul- tiples of 30 (30/10)	144	72	88	38K	1.1/1.37	80KBPS
8. CCITT H.261 (CIF)	---	1:1	submul- tiples of 30 (30/10)	288	144	176	152K	4.6/1.5	320KBPS
9. MPEG 1	---	1:1	30	240	120	176	127K	3.8	1.5MBPS
10. MPEG 2/H.26X	---	2:1	30	480	240	360	518K	15.5	5MBPS
DIGITAL HDTV PROPOSALS									
11. ATT/ZENITH	767.5	1:1	60	720	360	640	1.38	82.8	
12. GI/DIGICIPHER	1050	2:1	30	960	480	362	1.7	51	
13. SARNOFF CONSORTIUM	1050	2:1	30	960	480	720	2.07	62.2	20 MBPS
14. GI/MIT		1:1	60	720	1280				

A plot of transmission rate as a function of pixel rate is provided in Figure 5.1. It is interesting to note that all three entertainment standards employ a compression of approximately 0.35 bits/pixel while the two teleconferencing standards require only 0.13 bits/pixel.

One interesting question is: What transmission bit rate would be required to deliver the HDTV signal with teleconferencing quality? A tentative answer is 8.5 Mbps as represented by point A on Figure 5.1.

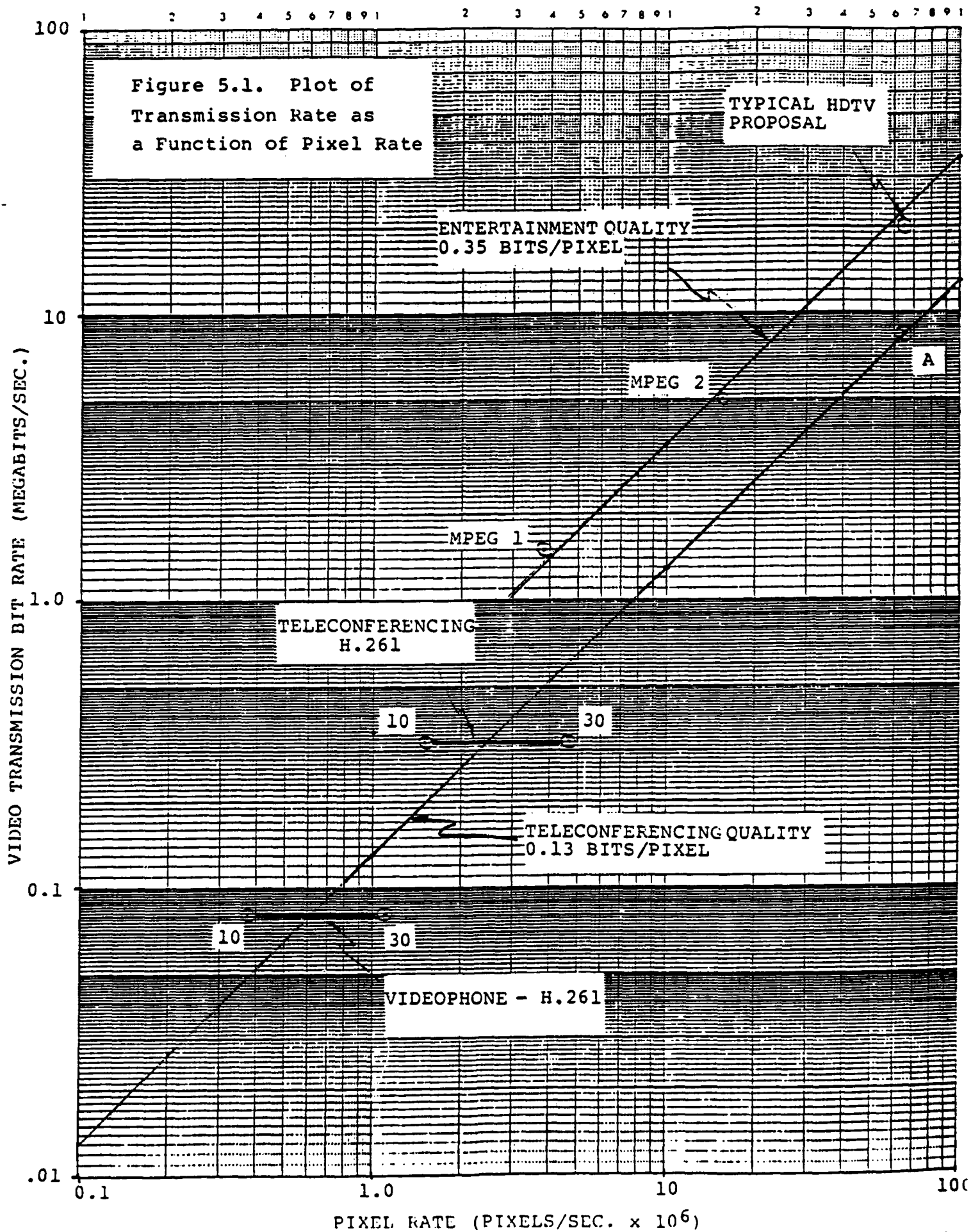
5.2 Videophone

There are two potential perceptions of a videophone service. The first refers to the transmission of a "head-and-shoulders" or "talking head" picture from one desk to another. This application requires very low spatial resolution (e.g. QCIF; 352 x 144 pixels) but good temporal resolution. The second application calls for high spatial resolution to view still pictures (e.g. charts, graphics) but very low temporal resolution. Obviously, the talking head application is not well matched to HDTV, but the graphics application is well suited. It may be feasible to view the HDTV videophone as a multi-media work station with windows. The basic presentation is a high resolution graphic image with a very small "talking head" window "superimposed".

5.3 Command and Control Function

A typical command and control function involves the transmission and display (desk monitor or large display board) of a map with superimposed text and vector lines. The HDTV format is obviously ideal for this application. It is useful to note that in many situations only a small part of the image changes at one time. In such a case, the P x 64 coding technique could readily transmit the HDTV signal using existing telecommunications or N-ISDN channels. Assume an HDTV frame requires 10^6 bits for definition (0.5 bits/pixel). If the channel rate were 384 Kbps, an image can be completely updated in approximately 3 seconds. A channel rate of only 64 Kbps may be well adapted for a display which changes at a slow rate. Obviously, the HDTV pictures for this C³ application could be originated in a computer or by an HDTV camera.

With virtually no changes, the H.261 Recommendation could handle HDTV



signals. The SMPTE 240M is a representative HDTV signal and has a resolution of 1920 x 1035 pixels. This corresponds to 11 x 22 Groups of Blocks (GOB) as viewed from an H.261 perspective (vs. 2 x 6 for CIF and 1 x 3 for QCIF). Such an HDTV picture could be completely updated in one second over a T-1 circuit using H.261.

The only limitation in the H.261 format as far as image size is concerned, is the GOB number. H.261 allows for GOB numbers from 1 to 15, although only 1 to 12 are used for CIF. For the 240M format, $11 \times 22 = 242$ GOBs are needed. However, according to H.261, every GOB header must appear. Therefore, there would be no problem if GOBs were numbered module 16 (excluding zero, which is used for the Picture Start Code). Only a massive error burst, involving thousands of bits, could cause any confusion about which GOB is being coded.

Using the full H.261 algorithm (interframe coding plus motion compensation) should give 0.1 bits per pixel for most images. For the 240M format, this should permit updating at a 7.5 frames/sec. rate, using the full T.1 data rate. The 0.1 bits/pixel coding rate is justified because of the high degree of correlation of adjacent pixels at such a high resolution.

For low bit rates, most of the capacity is used up by overhead, mainly in the form of GOB headers. For a switched 56 Kbps channel carrying only video, there is a net video bit rate of 52,275 bits per second, after taking out FAS, BAS, and FEC bits. Each frame requires 6,324 bits of header data, so the maximum frame

rate is $\frac{52,275}{6,324} = 8.27 \text{ frames/sec.}$ even if there are no changes. However,

if the image is a highly detailed status chart, changes involving only a small fraction of the image would be updated virtually instantaneously. Initializing the status chart, or changing from one to another, would take about 20 seconds at 0.5 bits per pixel for intraframe coding.

5.4 Training

HDTV offers many opportunities in the educational field where an HD display is required. Using an aspect ratio of 16:9, it becomes easy to include subtitles, data, and other information as explanatory material on either the right or left side of the screen, or to divide the picture into two halves and display them side by side.

For expansion of applications, enriching the availability of peripheral equipment is important, and for industrial education computer graphics can be employed to visualize an imaginary phenomenon.

There are a wide range of possible training applications which could use HDTV very advantageously. However, many of those requirements can be well satisfied by analog HDTV tape recorders and display configurations. No transmission is required. In the case of medical training, it may again be feasible to employ analog transmission from satellite in a broadcast mode. However, the cost for the analog HDTV channel would probably be prohibitive. Consequently, compression would be the key technology to permit the use of HDTV for training to be practical and economical.

There are also cases where the government wishes to interactively train personnel using a high resolution display. In this case, it would be advantageous to operate with a digital HDTV picture so that small parts may be altered interactively depending upon the training mission.

There are many training situations where the imagery is a series of still pictures as opposed to full motion. In this case, as in prior applications, HDTV could be digitized effectively with either the JPEG or H.261 algorithms for narrowband storage or transmission from a remote location.

5.5 Multimedia

Multimedia services involve the integrated delivery and control of multiple information types within a single service, e.g. video, audio, imagery, and text. Multimedia communication will become increasingly important as multimedia applications become more prominent in workstation and personal communications environments. The basic framework for a desktop multimedia workstation is a display having a high resolution comparable to the HDTV formats. There is therefore a strong linkage between the multimedia workstation environment and HDTV.

Many of the functions previously associated with the multiplexing of components of a multimedia connection, and embodied in end-to-end signalling and framing systems, such as Recommendation H.221, are provided by basic ATM functionality. The impact of this development is seen in the range of multiplexing options available for combining multiple media for transport and switching.

5.6 Summary

In summary, the applications of HDTV to government point-to-point communications fall into three broad categories as distinguished by the type of coded involved. The three categories are discussed below.

- Existing standards: JPEG, P x 64 - Existing codec standards (JPEG, P x 64) may be used effectively for transmitting HDTV signals for all applications listed above. Applicable communication channels include existing telecommunications (switched 56 Kbps, T1), N-ISDN, and synchronous B-ISDN.
- High Bit Rate Synchronous (e.g. DS3; 45 mbps) - High quality T/C and other applications can be provided within a local area via DS3 (45 mbps) channels using the HDTV signal. This codec is not yet standardized nor developed.
- Variable Bit Rate Codec for ATM - In the future, the HDTV signal will be efficiently transmitted via the B-ISDN operating in the ATM. A new codec operating at a variable bit rate is in the process of being standardized for this application. Work is now underway in the T1Y1.1 and CCITT (H.26x) to accomplish this objective.

6.0 CONCLUSIONS

The following conclusions are drawn from the work performed on this project.

- There is a general trend toward the adoption of a domestic standard for HDTV transmission based upon all digital technology. At the present time, there are four proponents of all digital systems as listed below.

PROPONENT TEAM	SYSTEM	SCAN FORMAT	LUMINANCE PIXELS	CHROMA PIXELS
AT&T, ZENITH	SPECTRUM COMPATIBLE	787.5/1:1	720 X 1280	360 X 640
GENERAL INSTRUMENT, MIT	DIGICIPHER	1050/2:1	960 X 1408	480 X 352
SARNOFF, NBC, PHILIPS, THOMSON	ADVANCED COMPATIBLE TV	1050/2:1	960 X 1440	480 X 720
GI/MIT	ATV PROGRESSIVE	787.5/1:1	720 X 1280	360 X 640

All four proposed systems employ DCT coding (8 x 8 pixels) and motion compensation which is similar to the coding technique employed in CCITT Recommendation H.261. This trend toward all-digital transmission is obviously a very favorable development for the community interested in teleconferencing.

- There is little progress towards an agreement on international standards for HDTV. The Japanese are moving forward with the MUSE system (1125 lines; 60 fields/sec.) while the Europeans are proceeding vigorously with the EUREKA system (1250 lines; 50 fields/sec.). However, there is considerable concern in these foreign countries that the U.S. all-digital standard will supersede their approaches.
- The communications infrastructure is well positioned to exploit the HDTV technology for teleconferencing and videophone applications. At the present time switched 56 Kbps channels and switched 1.544 mbps channels are available. As the narrowband ISDN becomes more available, the 56 Kbps

channels will be replaced with 64 Kbps and a switched 384 Kbps service will become available. In the longer term, the Broadband ISDN will become available having an interface at 152 mbps.

- The key issue from a TV compression perspective is the work to insure that the HDTV standard is interoperable, scalable and extensible with other telecommunication standards such as P x 64.
- It is anticipated that the HDTV system will have a significant impact within the Government community. Examples of important applications are teleconferencing, command/control, and desktop multimedia workstations.

APPENDIX A

Documents
CCIR Study Groups
Period 1990-1994

Document USTG/11-1/1
Document 11-1/
06 September 1991
Original: English

Received:

Subject: Question 47/11
draft new Question [AAB/11]
draft new Question [AAC/11]

United States of America

INTEROPERABILITY CONSIDERATIONS FOR DIGITAL HDTV

1. Benefits of interoperability

The future benefits of video and other image technologies will be greatly enhanced if universal interchange of all kinds of images and image sequences can be implemented and managed economically. The ultimate beneficiary is the consumer who will have, at the consumer's option, image information of any kind in a form chosen by the consumer, instantly available, at an affordable price.

Rapid advances in digital semiconductors, digital communications, and digital processing algorithms will make it possible to tailor the video technology to specific applications (in terms of picture quality, price, format, and performance). Such a diversity in the video marketplace will be a positive development only if it is easy to move among different formats, applications, industries, and media. The key idea is to remove impediments to interworking among multiple formats so that market forces can guide the developments of products and services.

2. Potential drawbacks of interoperability

There are economic costs associated with the inclusion of interoperability in a video architecture. For example, building the desired flexibility into a terrestrial broadcast HDTV system could increase the cost of TV receivers. Not only will obvious additional costs be incurred due to additional connectors and interfaces, but the system architecture may need to be made more complex.

3. Types of interoperability

The potential for image interoperability exists at several levels or dimensions: among different signal formats, among different transport media, among different industries, among different applications, among different epochs, and among different geopolitical entities.

Examples include film to HDTV to NTSC to ultra-high definition images for different formats. Terrestrial broadcast, satellite, fiber optic networks and coaxial cable represent different media that will need to interconnect.

Education to medical imaging to entertainment television is an example of different industries that can usefully exchange image information. CAD-CAM, image databases, computer art, and entertainment TV are different applications of image technology with a need for interoperability. Education, history, and entertainment can benefit from interoperability among different epochs using archived as well as current images. Transfer of image data between the United States, Japan, Europe, Africa and the rest of the world is the geopolitical dimension.

4. Importance of digital representation for interoperability

Digital representation of signals is the key element in achieving interoperability for images and video. The digital nature of the signal means that all systems that process the signal have identical material to process. The ease of storing, transporting, and processing digital data is matched by the growing speed, power, and economy of semiconductors.

Once in digital form, signals can be filtered and processed in a predictable and reproducible way so that conversions among formats can be implemented using functions selected based on mathematical theories such as sampling, interpolation, and prediction.

5. Industries with interests in high-resolution images

While the traditional entertainment television industry has been built around a single dominant format (NTSC in the United States), the non-entertainment industries have generated a number of formats for still and moving images. In some cases the non-entertainment applications have used NTSC, although there has not always been a good match of NTSC capabilities to requirements.

Many now seem to believe that technology has reached a point where many new standards are about to be set, and the opportunity should not be missed to harmonize the various image and video standards. The advocates of harmonization are mainly in the computer, computer graphics, and telecommunications industries, and academia.

The advantages seen by those industries from interoperability is the ability to share a common technology with the large consumer television market thereby increasing the size of the market for non-entertainment image and video applications.

6. Specific attributes relevant to interoperability

Certain specific attributes of image-related systems contribute to interoperability and are described below.

6.1 Digital representation

The importance of a digital form for signal representation has been discussed above. This is the major attribute required for interoperability.

6.2 *Progressive scanning*

Progressive scanning in a raster-based sequence of images simplifies, to some extent, the filtering and interpolation used to convert among formats with different numbers of scan lines, different numbers of samples per line, and different temporal sampling (i.e., picture rate).

6.3 *Square pixels*

For computer graphics, equal geometric spacing among horizontal samples on a line and among samples displaced vertically is desirable for simple rendering of objects that may be transformed after creation.

6.4 *Provision for headers and descriptors within data*

An important area of agreement among advocates of interoperability and harmonization of images and video is the desirability of headers and descriptors imbedded within the stream of image data. The purpose of the headers and descriptors is to identify reliably and unambiguously the form of the data. The headers could include information on how the images or image sequences were originated, processed, and compressed. The headers could use a small fraction of data capacity to make image data streams self-identifying.

7. *Interoperability in proposed digital HDTV systems*

7.1 *Digital representation*

DigiCipher, DSC-HDTV, ADTV, and ATVA-P use digital processing and digital broadcasting and therefore meet the major requirement for interoperability, digital signal representation.

7.2 *Progressive scanning*

DSC-HDTV and ATVA-P use progressive scanning.

7.3 *Square pixels*

DSC-HDTV and ATVA-P use square pixels.

7.4 *Provision for headers and descriptors within data*

DigiCipher, DSC-HDTV, ADTV, and ATVA-P provide for auxiliary data in digital form. Part of that auxiliary data channel capacity could be used for headers and descriptors to make the data streams self-identifying.

8. *Summary*

Digital HDTV systems have the potential for facilitating interoperability among high resolution image systems. Selection of an advanced television system that incorporates attributes needed for interoperability will harmonize interchange of still and moving images from diverse sources. Carefully designed headers and descriptors imbedded within the digital HDTV data can be used to identify the image data to current and

future receivers, and can be incorporated within the protocols of telecommunications networks used for both entertainment and non-entertainment services.

REFERENCES

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APPENDIX B

PRESS RELEASE

Richard E. Wiley, Chairman of the FCC's Advisory Committee on Advanced Television Service, announced the appointment of Dr. Robert L. Sanderson of Eastman Kodak Company as a new Vice Chairman of the Alternative Media Interface Working Party to direct a review of "interoperability" and "extensibility" with regard to advanced video systems. The Working Party is chaired by Edward Horowitz of Viacom, and the Planning Subcommittee by Joseph Flaherty of CBS.

Interoperability (i.e., the ability to display different video and image formats) and extensibility (i.e., the ability to extend to technological advancement in the future) are two of the factors previously identified by the Advisory Committee to be considered in the establishment of a new video standard.

The Working Party already has developed a specification defining a "multiport" receiver. It will now focus, for example, on the degree to which high definition television systems can be useful to other applications and industries, such as telecommunications, publishing, graphics, medicine, education, computers, and science. The Working Party will also consider how future improvements and applications of high resolution video systems can be most effectively accommodated.

Mr. Wiley stressed that the review would be conducted within the established timetable of the entire Advisory Committee. Specifically, no delays in the Committee's final report to the FCC are contemplated as a result of Dr. Sanderson's efforts.

This new Working Party initiative will require input by experts from a variety of disciplines and industries. Anyone who may be interested in participating should contact either Dr. Sanderson at (716) 253-5362 or Mr. Horowitz at (212) 258-6324.

APPENDIX C

STANDARDS ORGANIZATIONS RELATED TO HDTV TELECOMMUNICATIONS

1.0 CCITT SGXVIII

Responsible for, among other tasks:

- **Recommendations on all network aspects of the B-ISDN, including the network architecture, transport techniques, User-Network Interface, access and inter-exchange signalling and ATM Adaptation Layer specifications;**
- **identifying network wide impact on B-ISDN service support, including the coordination across ITU bodies necessary to maximize commonality between communicative and distribution video services;**
- **establishing the framework for video service support in the B-ISDN;**
- **general aspects of quality of service and network performance in digital networks including ISDNs;**
- **providing coordination across different groups on Integrated Video Services in B-ISDN.**

The following areas of responsibility are SGXVIII's understanding of the terms of reference for the other groups interested in video services on B-ISDN.

2.0 CCITT SGXV

Responsible for, among other tasks:

- **Recommendations on video coding algorithms necessary to support a range of different quality communicative video services on the B-ISDN;**
- **Recommendations for transmission systems and equipment utilized in the B-ISDN;**
- **Recommendations for audiovisual system aspects;**
- **cooperatively assess compatibility between video coding algorithms used for the support of communicative and distribution video services with CMTT.**

3.0 CCITT SGXI

Responsible for, among other tasks:

- **Recommendations on Stage 2 and Stage 3 service descriptions for the Stage 1 service descriptions as provided by SGI.**

4.0 CCITT SGVIII

Responsible for, among other tasks:

- **Recommendations on coding for still image communication.**

5.0 CCITT SGI

Responsible for, among other tasks:

- **Recommendations on the service attributes including end-to-end service quality of all communicative services supported on the B-ISDN;**
- **Stage 1 service descriptions;**
- **cooperatively assess compatibility of end-to-end performance levels of television and communicative services on the B-ISDN with CCIR SG11.**

6.0 CMTT

Responsible for, among other tasks:

- **Recommendations on the bit rate reduction coding, packaging and transmission of television and sound programme signals in all portions of the telecommunications network; this includes contribution, primary distribution and secondary distribution signals;**
- **cooperatively assess compatibility between video coding algorithms used for the support of communicative and distributive video services with CCITT SGXV.**

7.0 CCIR SG11

Responsible for, among other tasks:

- **Recommendations defining and assessing the subjective and objective performance of digital television coding schemes proposed by the CMTT and supported on the B-ISDN proposed by CCITT SGXVIII;**
- **cooperatively assess compatibility of end-to-end performance levels of television and communicative service on the B-ISDN with CCITT SGI.**

8.0 IEC/ISO

Responsible for, among other tasks:

- **Through the Moving Picture Experts Group (MPEG), develop standards for storage and retrieval of moving images and sound for Digital Storage Media (DSM);**
- **development of standards for display devices.**

END