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INTERNETWORKING: DISTANCE LEARNING "TO SEA" VIA DESKTOP VIDEOCONFERENCING TOOLS AND IP MULTICAST PROTOCOLS

by

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13. ABSTRACT

While deployed at sea, sailors are traditionally provided much of their education at sea through correspondence and pace courses. But with recent developments in the Internet and videoconferencing, it is now feasible to deliver real-time educational material anywhere, even to a ship at sea. This thesis investigates the current status of networked desktop videoconferencing technology, and its use in support of Joint Vision 2010, with respect to Distance Learning. It provides an analysis of videoconferencing protocols, standards, and applications, as well as a videoconferencing pilot project. The objective of the analysis is to determine the viability and economical benefits of using videoconferencing technology and collaboration tools, from the desktop, as a means for simultaneously delivering synchronous and asynchronous distance learning material from an academic location to multiple students at remote locations. The results show that desktop videoconferencing technology, via IP based networks in the Defense Information Infrastructure, is a viable tool that can add numerous economical benefits, such as a decreased spending for travel and eliminating the need to rely on large, room-based systems.

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INTERNETWORKING: DISTANCE LEARNING "TO SEA" VIA DESKTOP VIDEOCONFERENCING TOOLS AND IP MULTICAST PROTOCOLS

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ABSTRACT

While deployed at sea, sailors are traditionally provided much of their education at sea through correspondence and pace courses. But with recent developments in the Internet and videoconferencing, it is now feasible to deliver real-time educational material anywhere, even to a ship at sea. This thesis investigates the current status of networked desktop videoconferencing technology, and its use in support of Joint Vision 2010, with respect to Distance Learning. It provides an analysis of videoconferencing protocols, standards, and applications, as well as a videoconferencing pilot project. The objective of the analysis is to determine the viability and economical benefits of using videoconferencing technology and collaboration tools, from the desktop, as a means for simultaneously delivering synchronous and asynchronous distance learning material from an academic location to multiple students at remote locations. The results show that desktop videoconferencing technology, via IP based networks in the Defense Information Infrastructure, is a viable tool that can add numerous economical benefits, such as a decreased spending for travel and eliminating the need to rely on large, room-based systems.

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I. INTRODUCTION

A. INTRODUCTION

This thesis investigates the current status of networked desktop videoconferencing technology, and its use in support of Joint Vision 2010, with respect to Distance Learning. It provides an analysis of videoconferencing protocols, standards, and applications, as well as a videoconferencing pilot project. It also follows work from the thesis "*Internetworking: Economical Storage and Retrieval of Digital Audio and Video for distance learning*,[Tiddy, 96].

B. MOTIVATION

DoD has implemented various videoconferencing systems in order to make distance learning more available, but there are still major obstacles.

The current systems that have been put into place are usually based upon a model using a dedicated room or roll-about system, with proprietary hardware and software. Also, users are still required to travel to the room-based systems in order to participate in the training sessions. Surveys of room videoconferencing system users have identified desired features such as shared drawing area, the ability to connect to multiple sites, and ways to incorporate computer applications into the conference [Retinger, 95]. Since there can be a large geographical dispersion of military personnel across numerous time zones, there is also the problem of coordination of class times between the instructor and the student.

Using desktops to deliver videoconferencing has multiple advantages: As users become more familiar with the use of PCs, they will not need to learn how to provide instruction using a room based system, which usually requires a dedicated person to mange the equipment. The instructor does not have to deal with scheduling blocks of time to use the room-based systems. Conferencing over the desktop can be more relaxed and impromptu, contributing to better human interaction. Most desktop videoconferencing software has whiteboard capabilities, allowing the student and instructor to share data in real-time.

C. OBJECTIVE OF THESIS

The primary objective of this thesis is to describe how desktop videoconferencing technology and collaboration tools can be used either synchronously or asynchronously to deliver Distance Learning content over an IP based network to multiple students at remote locations. Instructors might be a Chief Petty Officer (CPO) at Fleet Training Center Pacific, an Admiral in Washington D.C., or a professor at the Naval Postgraduate School (NPS). The topics of desktop videoconferencing in regard to human/computer interaction aspect and social issues will not be discussed here, but can be found in [Rettinger, 95]. Test and evaluation of a prototype system at NPS provides an example demonstration how distance learning can be achieved via the PC to any remote user's desktop. Specifically, the research and experiments for this thesis were designed to collect data to address the following research questions:

- How can we leverage the Defense Information Systems Network (DISN) to implement desktop videoconferencing distance learning to the sea?

- What are some of the current protocols and standards available in order to multicast desktop videoconferencing applications via an IP based network?

- How can we leverage the Navy's current JMCOMMS/ADNS program to implement desktop videoconferencing distance learning to a shipboard LAN at sea?

- What are the technical and management concerns in order multicast videoconferencing applications to the user at sea?

- What impact will multicasting video over DISN have on the system bandwidth/availability?

- What are the hardware and software requirements for the instructor and student, in order to maintain reliable communications throughout a course of instruction?

- What are some of the available videoconferencing applications that can be used for distance learning?

- How much will desktop videoconferencing (distance learning) offset travel expenses for resident education?

Preliminary results are evaluated for each of these questions.

D. SCOPE OF THE THESIS

The scope of this thesis includes: (1) Show how multicasting across IP-based networks can be used to deliver desktop videoconferencing distance learning to sea. (2) Review some of the currently available videoconferencing products and how their use can be leveraged for distance learning, (3) Using a prototype, test and evaluate the feasibility of the delivery and storage of videoconferencing data over an IP based architecture to-sea. The goal is to evaluate and determine the economical and technical benefits of using currently available desktop videoconferencing applications (versus cart and room-based systems) as an alternative tool that an instructor and student can use to exchange course material over an IP-based Internet and DISN.

The demonstration incorporates desktop workstations with cameras, video capture card, audio card, and a network connection to IP multicast capable routers. Besides the standard Internet protocols normally found on current desktop computers, it also contains videoconferencing applications capable of multicasting video and audio, either synchronously or asynchronously, to naval students at remote locations and at sea.

E. METHODOLOGY

The methodology used to produce this thesis included the following tasks:

- Conduct a literature search of books, magazines, articles, Internet resources and other library information services describing videoconferencing technology and current software/hardware that can be applied to distance learning in the military.
- Conduct a search of books, magazines, articles, Internet resources, and consult with companies to determine the current videoconferencing software and hardware that are best suited for Internet-to-the-sea videoconferencing.
- Develop a model to demonstrate how distance learning courses can be seamlessly transported from the instructor to the Internet and the Navy's communication networks infrastructure, in order to provide Internet-to-sea videoconferencing.

- Develop a prototype videoconferencing system that might be used as a part of a "toolbox" that can be used export a correspondence course or graduate school class to a ship.
- Consult with the Space and Naval Warfare Systems Command (SPARWAR) and the Research, Testing and Evaluation Division of the Naval Command Control and Ocean Surveillance Center (NRAD) on current developments of the Joint Maritime Communications System/Automated Digital Network System (JMCOMMS/ADNS) and its current use with videoconferencing technology.

F. THESIS ORGANIZATION

This thesis is composed of eight chapters. This chapter provides the motivation, objectives, research questions, scope and methodology employed to conduct the research. Chapter II provides the history of videoconferencing, and related work. Chapter III discusses the current video and audio compression protocols and standards that are required for current videoconferencing systems. Chapter IV describes the various multicasting protocols and standards necessary to provide scalability, cross-platform support and quality of service (QoS) necessary to provide distance learning from the desktop over the commercial and naval IP based networks. Chapter V describes various options that can be applied over the DISN architecture that will support IP based desktop videoconferencing to sea. Chapter VI compares some of the desktop videoconferencing applications and protocols required to deliver distance education to sea. Chapter VII discusses the demonstration project and findings. Chapter VIII provides the conclusion, summary and recommendation for future research.

II. RELATED WORK

A. INTRODUCTION

This chapter provides a brief history of videoconferencing and the traditional methods used to provide distance learning to personnel in remote locations. It gives a brief overview of the various methods that can be employed to deliver distance learning across a network (WAN). Finally, it describes some of the current VTC/videoconferencing solutions used in the Navy and DoD.

B. BRIEF HISTORY OF VIDEOTELECONFERENCING

Videoconferencing was first introduced in 1926 when AT&T's President, Walter S. Gifford, used Video Teleconferencing to speak with the Secretary of Commerce, Herbert Hoover. [Nerino, 94] Not until the late forties and early fifties, with the advent of the television, did the next major breakthrough in video technology come about. After television, videoconferencing did not see another major breakthrough until AT&T introduced its picture telephone at the 1964 New York World's Fair. Even then, because videoconferencing contained frequencies that were beyond those used by telephone networks at that time, expensive satellites were used to provide the medium needed for high bandwidths required for videoconferencing. By 1983, full-bandwidth satellite transmissions still cost over $1 million per year [Nerino, 94]. Today such satellite links are becoming more affordable.

As the 1970's progressed, new advances in computing power and improved methods for converting analog signals to digital formats resulted in telephone service providers transitioning to digital transmission methods to compliment the existing analog processing systems. Although videoconferencing has become more widely used for services like business meetings, collaborative research, distance learning, etc., these service are generally performed over dedicated leased lines and usually requires expensive room-based or roll-about videoconferencing systems.

Today, due to faster desktop computers and the rapid expansion of the World Wide Web and the Internet, transmitting real-time video using desktop computers to remote locations has become practical. Although there is currently an explosion in the number of applications that can transmit and receive streaming audio and video to and from a PC over the Internet, there continues to be significant interoperability, protocol and architectural issues that must be addressed if videoconferencing is to become commonplace from the desktop.

C. DISTANCE LEARNING

1. Traditional Educational Methods

Educational development has been always been required in the career progression of naval personnel. This training is essential to achieving and maintaining national security, as well as national strategic objectives [Emswiler, 1995]. Traditionally, the primary methods of providing the necessary education to Naval personnel has been through the following methods:

- Short-term temporary duty seminars
- Resident education at technical schools (A, B, and C schools)
- Resident education at undergraduate or graduate educational institutions, e.g. Naval Postgraduate School (NPS) or Naval War College.
- Postal-based correspondence courses by postal mail.

Courses that require travel on a TAD basis are useful for initial or technical refresher type training nevertheless, this approach is costly and requires travel by the instructor, student or both.

Resident education at NPS requires students to stay away from the operational forces for two years, on average. Although many courses require the student to be present to obtain the desired educational benefit, others can be easily and readily exported to sea or a remote shore location.

Traditionally, postal-based correspondence courses have been necessary due to the remote locations that naval personnel are often stationed. If the course is the equivalent to a resident course, however, management of the correspondence course will be substantial. In order for the correspondence course to be successful, not only must there be a sustained commitment from the student, but the feedback loop to the student must be amenable to continuing, timely instruction. Often such a feedback loop is not the case, as sometimes it may be weeks, due to numerous reasons, before the student receives feedback or new modules. As a result many students do not finish.

2. The Value of Distance Learning in the Navy

Distance learning in the Navy can be beneficial in two important areas; cost and global reach. In a decreasing defense budget, the allocation of MILPERS dollars, which pay for travel and education, is ever decreasing. Besides costs, the naval environment requires personnel to be deployed at remote or isolated settings that are far from traditional educational resources. A more time efficient delivery of course material and feedback to the student can markedly improve the dedication of the student to complete the course of instruction. Figure 2-1 outlines the general situations when distance learning can be advantageous to traditional methods.

- **Target audience is widely scattered and it is not cost effective or possible to have them travel to a central training location.**
- **Content or consistency in delivery is so critical that it must be carefully controlled for accuracy or correct interpretation.**
- **Content is too dangerous for novices to participate in and distance education will allow for familiarization and confidence building prior to the actual situation.**
- **Scheduling difficulties arise because the student cannot take extended time from other critical missions to attend a normally conducted training program.**
- **The expense of conducting live training is cost prohibitive.**
- **There are a limited number of qualified trainers.**

Figure 2-1 Productive applications of a distance education approach [Biggs, 94]

3. Distance Learning via the Internet

The World Wide Web (WWW) provides a means of providing both time-efficient course material and research tools. Distance education can be as a simple as a correspondence course offered through electronic mail, something as complex as interactive video teleconferencing over the Internet, or combinations of both [Tiddy, 96].

As more ships, commands, and individual units become connected to local area networks (LAN's) and wide-area networks (WAN's), distance learning programs can be more easily implemented, ultimately providing more economical resources for training [Emswiler, 95]. Also, as video/audio application and transport protocols and standards become more established, commercially produced products become more readily available to furnish the tools necessary to provide distance learning over commercial and Department of Defense (DoD) networks. To date, however, the growth of Internet and DoD network applications and users are outpacing growth of bandwidth. With limited dollars for education and travel, DoD can not wait until this trend reverses itself. Therefore it is critical to use well-developed standards and protocols, i.e. multicasting, compression, etc., along with existing network infrastructures, in order to get the most efficient delivery to remote users.

4. Videoconferecing in the Department of Defense

DoD has used videoconferencing technology in a wide variety of applications. Some of the major areas where this technology is being used is in:

- Training
- Telemedicine
- Group Conferences/Meetings
- Crisis Response

Videoconferencing technology has started to bring significant savings to DoD, mainly in travel expenses. The need for military personnel to travel to attend meetings, conferences, training, and exercises has been greatly reduced for commands that have access to videoconferencing equipment. The following examples contain more specific descriptions of areas where videoconferencing technology is being or has been applied in DoD:

a.) Training: NPS Distance Learning via the Multicast Backbone (MBone) system: NPS has conducted "Distance Learning" or remote classroom instruction, through the use of videoconferencing technology over the MBone. In a 1995 thesis by Tracy Emswiler, it was demonstrated that videoconferencing technology could be an economically feasible approach to distance learning. It documented Dr. Richard Hamming's course, "Learning to Learn", being transmitted worldwide over the MBone for an entire quarter. [Emswiler, 95]

NPS is also currently delivering distance learning in Root Hall, using a PictureTel 4000 Video conferencing Systems over Integrated Services Digital Network, Basic Rate Interface (ISDN BRI) lines. Courses, and even some degree programs, are offered in Computer Science, Electrical Engineering, Aerospace Engineering and Information Technology Management.

The Chief of Naval Education and Training (CNET) Electronic Schoolhouse Network (CESN) is a two-way video and audio multipoint, secure distance learning network. It allows simultaneous instruction to multiple shore and shipboard sites, where individuals can interact both verbally and visually in a real-time mode. Its purpose is to provide effective training to a large number of personnel at or near their duty stations, eliminating the need for travel to distant schoolhouses, thereby reducing travel and per diem costs.

The Navy's Video Tele-training (VTT) CESN is linked via land lines and operates at a fractional T-1 data rate of 384 Kbps. Communication is provided through the government's long-haul communications network using FTS2000. Satellite capability is available for shipboard VTT. The network is made up of 16 sites nationwide and includes a site on board the USS George Washington [CNET, 97].

b.) Telemedicine: This is a field where videoconferencing is making significant inroads. Basically, the same idea from distance learning is applied to telemedicine: A central care facility with medical expertise (i. e. physicians, surgical staff, etc.) can provide care "remotely" to a distant site via videoconferencing. A huge potential for this technology exists in afloat applications, since most U. S. Navy and Coast Guard ships have medical personnel who can provide a only a basic level of care. One practical use was demonstrated when Telemedicine was in used on the USS George Washington (CVN 73) to provide mental health examinations, during a 1997 deployment. Psychiatrists successfully evaluated onboard patients, capturing their mood, body language and response to questions [Koenig, 97]. Additionally, during JWID 97, the Naval Medical Information Management Center (NMIMC), Bethesda, Maryland sponsored a demonstration of telemedicine technologies aboard the submarine USS Atlanta (SSN 712) in Norfolk, Virginia. Once ships are routinely outfitted with this technology, a tremendous benefit in Telemedicine will surely be realized.

c.) Group Conferencing: In September 1995, a major Joint Task Force (JTF) Exercise was conducted in Panama: Exercise "Fuertes Defensas" (Strong Defense). Led by the Commander, 18th Airborne Corps, this exercise was conducted to test United States readiness to support and defend the Panama Canal. Each day, the JTF Commander (an Army LGEN) was able to keep advised of exercise progress by conducting a morning

Videoconference with his Army, Navy, Air Force, and Marine Component Commanders. These commanders were sometimes physically separated by hundreds of miles. Because of videoconferencing technology, the commander was able to both remain well informed of exercise progress, and also was able to promulgate his own directives and intentions for the day.

d.) Crisis Response: There is a huge potential for further use of videoconferencing technology for Crisis Response Management. For example, Navy and Marine Corps Afloat and Expeditionary Commanders might receive real-time combat instructions from their superiors via videoconferencing. Also, these Task Force Commanders might promulgate their own guidance to their attached ships and elements in the same fashion, all the way down the chain of command. There is also a large potential for this technology in non-combat crisis management situations, such as humanitarian disaster relief operations.

III. MAJOR VIDEOCONFERENCING STANDARDS

A. INTRODUCTION

This chapter will discuss the major videoconferencing standards, as they are significant issues when implementing distance learning to sea from the desktop.

B. BACKGROUND INFROMATION

The International Telecommunications Union (ITU), a body of the United Nations that focuses on developing standards, tasks the Telecommunications Standardization Sector (ITU-T) with developing telephony standards. It develops some of the major protocols that are used by IP-based videoconferencing systems today, such as H.320, H.323, and H.324. Table 3-1 provides an overview of those standards.

Standard	Description	Remarks
H.320	H.320 is an "umbrella" standard that covers audio, video, videoconferencing, graphics, and multicasting	Mandatory standard by the Federal Government in 1993.
H.323	Visual (audiovisual) communications over LANs	Addresses audiovisual communications across LANs and gateways that connect LANs to the Internet.
H.324	Defines a multimedia communication terminal operating over the Switched Telephone Network. It includes H.261, T.120, and V.34.	Incorporates the most common global communications facility today -- (POTS)

Table 3-1 : ITU-T Videoconferencing Standards

The videoconferencing systems and standards described above can be viewed to have evolved over three generations. The 1st generation systems were generally point-to-point, proprietary systems that usually required dedicated T-1 (1.5Mbps) networks or better. Videoconferencing coding and compression was usually done by hardware compressors/decompressors (codecs). There were not many standards initially because interoperability of the various systems was not perceived as an issue. 2nd Generation systems were driven by Integrated Services Digital Network (ISDN). The compression was also usually done by proprietary, hardware codecs. As the technology matured, and compatibility became more of an issue, videoconferencing application developers began to adopt universal standards, ultimately migrating towards ITU-T's H.320 protocol. Also, ISDN's inability to scale to a large number of users limited its acceptance. Today, as network-centric computing has migrated to the core of many organizations, compatibility has become a focal point in the development of videoconferencing systems, thus bringing about 3rd generation system protocols. These new standards are generally designed to match the ISO seven-layer reference model. Now, advances in modeling and simulation (such as MPEG-4 compression), and improved scalability due to multicasting, 4th generation standards are coming about.

C. 1st GENERATION STANDARDS

1st generation videoconferencing systems are usually large room-based systems that are connected via dedicated circuit switched or T1 connections. These systems are point-to-point, and use proprietary system standards to deliver and receive content. Additionally they are not very scalable, and many of the international standards-based systems used today are not are not backwards compatible with them. Therefore they would be not be feasible for providing IP based distance learning to sea.

D. 2nd GENERATION STANDARDS

1. H.320

H.320 - "Narrow-Band Visual Telephone Systems and Terminal Equipment" is the umbrella standard that covers audio, video, videoconferencing, graphics and multicasting. ITU-T recommends it as the minimum standard that will ensure that videoconferencing systems will communicate with each other. H.320 covers a family of standards that governs videoconferencing systems that use coder/decoders (codecs) between 64 Kbps to 1920Kbps (64Kbps x 30). It became the mandatory standard for the Federal Government in 1993 [Nerino, 94].

The difference between the various videoconferencing systems will depend upon the optional requirements that each can support, which will ultimately effect the quality of the audio and video. How well the features are implemented is left up the each manufacturer. Table 3-2 shows H.320 recommendations and their titles.

Video Codec	**H.261**: Video Codec for audiovisual services at p x 64
Audio Codec	**G.711**: Pulse Code Modulator (PCM) of Voice frequencies **G.722**: 7 Khz audio-coding with 64 Kbps **G.728**: Coding of speech at 16 Kbps using low delay code excited linear prediction
Frame Structure	**H.221**: Frame structure for a 64 to 1920Kbps in audiovisual teleservices
Control and Indication	**H.230**: Frame-synchronous control and indication signals for audiovisual systems
Communication Procedure	**H.242**: System for establishing communication between audiovisual terminals using digital channels up to 2Mbps

Table 3-2: H.320 Recommendations [Nerino, 94]

H.320 only requires vendors to support the minimum standards. When deciding between systems, there are currently three classes of videoconferencing systems:

Class 1 – minimum level of support

Class 2 – Class 1 + support of some optional features

Class 3 – Class 1 + all optional features [VTEL, 95]

The major factors that affect system quality are picture resolution, frame rate, preprocessing and postprocessing, motion compensation, audio, data rate and quality.

a. Picture Resolution

Picture Resolution is the frame format of the video picture. The National Television Systems Committee (NTSC) standard picture frame consists of 780 horizontal picture elements (pixels) and 480 active vertical lines. Due to bandwidth constraints of the standard videoconferencing channels used today, that picture size is not practical for current videoconferencing systems. H.320 uses quarter common intermediate format (QCIF) – 176 X 144 pixel resolution, and common intermediate format (CIF) – 352 X 288 pixel resolution. If there is a connection between different classes of picture resolution, systems negotiate a resolution to the lowest one.

b. *Frame Rate*

H.320 can support frame rates of 7.5, 10, 15, and 30 frame per seconds (fps). Class 1 systems can support a frame rate of 7.5 fps; Class 2, typically about 15 fps, using QCIF; and class 3 supports 30 fps, using CIF. Frame rate negotiation uses the lower class when two or more classes are used. [VTEL, 95]

c. *Preprocessing and Postprocessing*

Preprocessing reduces the amount of re-coding in the background. If there is poor camera lighting, video "noise" can make the system think that there is motion in the background when in fact there is none. Preprocessing prevents the video encoder from wasting time encoding "noise" caused by the poor lighting, ultimately ensuring that only real motion gets encoded [VTEL, 95].

Postprocessing compensates for the picture degradation due to fast motion. It can help reduce the "blocking" and noisy effects caused by video codecs (discussed in more detail under H.261). Postprocessing is also can be used to enhance the frame rate, thus reducing jerky motion [VTEL, 95].

d. *Motion Compensation*

Motion Compensation is another video quality enhancement. There are two aspects of motion compensation: motion estimation and actual motion compensation. Motion estimation is performed at the video encoder to determine the motion vector of the subject. Motion compensation is performed at both encoder and decoder. It consists of moving blocks of video data around based on the motion vector determined during motion estimation. Especially important at lower bit rates, motion compensation moves only the encoded section of video where motion has occurred rather than the entire video area of each frame. All H.320 systems have the ability to decode a motion compensation signal. Providing encoded motion compensation (where the real video quality improvements are made) is optional [VTEL, 95].

Although the aforementioned factors affect H.320 system quality, many other elements also affect quality. Table 3-3 provides a summary of H.320 compliance.

	Level 1 (Minimum)	Level 2 (Medium)	Level 3 (High)
Frame Format (Pixels)	QCIF (176 X 144)	CIF (352 x 288)	CIF (352 X 288)
Frame Speed (frames/sec)	5	Up to 15	Up to 30
Data Rate	56 / 64 Kbps	Up to 384 Kbps	Up to 1.544 Mbps
Motion Compensation	No	Limited (6X6 = 36)	Full Motion (30X30 = 900)
Pre and post processing on both encoder and decoder	Not Applicable	Not Applicable	Pre and post processing on both encoder and decoder

Table 3-3: Levels of H.320 compliance [Nerino, 94]

2. H.320 and ISDN

ISDN is a connection-oriented circuit-switched digital communication service that is provided by telephone companies and network providers. It provides end-to-end digital connectivity between local area networks (LANs). ISDN connects users to LANs and ca also connect LANs to widearea networks (WANs). The basic ISDN connection bandwidth is 128 kbps, split among two bearer (video, audio) channels at 64 kbps each. There is an additional 16kbps data channel that provides connectivity data.

Implementation of ISDN channels is fairly flexible. Telephone companies provide services that allow ISDN channels to split (i.e. 64kbps channel split into two 32kbps channels to provide low-fidelity digitized voice), or bonded together. Bonding is accomplished by creating one logical channel out of multiple virtual channels. For example, the Navy's Video Information Exchange System (VIXS) uses bonding to provide bandwidth of 112–384kbps in order to allow afloat and ashore nodes to conduct face-to-face meetings in real-time.

ISDN offers improved videoconferencing connectivity over dedicated, point-to-point systems, because it works over existing phone lines and does not require the installation of an extensive network backbone. Unfortunately some major reasons remain why ISDN is not a good long-range alternative for distance learning. One is the lack of access to remote users in a globally dispersed military environment. Also, in order to multicast, you must deal with how the end points are going to be handled, i.e. adding multipoint control units (MCUs). Finally, continuing to implement ISDN as the primary videoconferencing long-haul architecture in the Navy is at odds with the Defense Information Infrastructure Common Operating Environment (DII COE) migration towards the consolidation of voice, video and data networks.

Recent versions of videoconferencing systems that use ISDN as its transport medium have begun to migrate to the H.320 protocol, but many vendors still use proprietary protocols in their videoconferencing systems. One possible reason:
although the H.320 standard is technically sound, ISDN has had a poor showing in the marketplace, consequently bundling H.320 with ISDN has inhibited initial acceptance of H.320.

E. 3rd GENERATION STANDARDS

1. Internet Videoconferencing

As the Internet and client-server computing continued to grow, videoconferencing systems for LANs and WANs began to be developed. H.323 (an extension of H.320) covers videoconferencing over narrow-band WANs and also over LANs. Since H.323 is based upon the IETF's Real-Time Protocol (RTP) -- which will be discussed in more detail in Chapter IV -- it can be applied to streaming video over packet-switched networks such as the Internet. H.323 also applies to point-to-point and multipoint sessions. Some of the other components of H.323 include:

- Specifying messages for call control including signaling, registration and admissions, and packetization/ synchronization of media streams.
- Specifying messages for opening and closing channels for media streams, and other commands, requests and indications.
- H.261 (video codecs)
- H.263 -- Specifies a new video codec for video over POTS (< 64Kbps).

- G.711, G.722, G.728 and G.729 standards
- H.230 Frame Synchronous Control Standards
- H.245 Link Control Standards
- T.120 Data Sharing Standard

2. Plain Old Televeision System (POTS)

POTS is the acronym for Plain Old Telephone Service. It utilizes the existing infrastructure of telephone lines and was designed to address the need for an inexpensive, high-quality solution for video conferencing over the existing infrastructure. The H.324 standard addresses high quality video and audio compression over POTS modem connections. Specifically it addresses and specifies a common method for sharing video, data, and voice simultaneously using high-speed (V.34) modem connections over a single POTS telephone line.

Video conferencing over POTS has been the least attractive of the medium options due to the bandwidth constraints. However, because H.324 incorporates the most common global communications facility today, POTS currently has a broad impact on the current marketplace. Even though the actual bandwidth of POTS hasn't grown much, it is still becoming less of an obstacle, since today's modem technology and data compression make it technically feasible to transmit both very low frame rate video and voice over a single line. As processors have become more capable, codec functions are now performed primarily in software, often achieving full-color, 15 frames per second (under optimal conditions), full duplex video and audio, with real-time responsiveness. Some of the major components of H.324 are:

- H.263 -- Defines speech coding at rates less than 64 Kbps.
- H.261 – Video Compression from 64 to 2Mbps
- H.223 -- Defines a Multiplexing protocol for low bit rate multimedia terminals.
- H.245 -- Defines control of communications between multimedia terminals.
- G.723 -- Defines speech coding for multimedia telecommunications transmitting at 5.3/6.3 Kbps.

F. VIDEO COMPRESSION

In the past, due to the bandwidth constraints of terrestrial mediums, satellite was the traditional and reliable method for transporting videoconferencing between users. Due to technical improvements in routing and switching, however, optimal high-quality videoconferencing can also be realized with dedicated circuit-switched channels. Unfortunately, due the high cost and lack of widespread availability of these channels, most desktop computer users do not have access to a dedicated videoconferencing link that can transfer data at the necessarily data rates. The chief digital transportation medium that the average computer user has access to is the Internet, which is based upon a non-guaranteed bandwidth, packet-switching technology often connected to an end user via POTS. Even as more capable routers, switches and modems are used to deliver videoconferencing, providing coherent end-to-end video and audio streams across the Internet remains a major obstacle, due to lack of guaranteed bandwidth. Video and audio quality can be very poor due to Internet congestion, routing delay, packet loss/retransmission, packet constant rerouting, limited multicasting capabilities, and other factors.

One way to improve bandwidth is to the compress the data prior to its traversing a network. This can generally be accomplished using two types of data compression schemes: lossless and "lossy." Lossless compression schemes are generally used in algorithms like, zip, gzip and gif file types. When using these types of algorithms, no data is lost during the compression and subsequent decompression of the data with approximations. The lossy compression algorithms search for and replace redundant data. Fortunately, due to the inability of the human eye to discern small losses of data in a digital image (notably the fact that small color details aren't perceived as well as small details of light and dark) lossy compression techniques are very suitable for videoconferencing.

There are a number of compression techniques available for use in videoconferencing, and H.261 is one of the most widely used in commercial videoconferencing products. Motion JPEG, Indeo, MPEG1, and MPEG2 are also prevalent. H.261 is optimized for bandwidth efficiency and low delay, whereas MPEG is less bandwidth efficient. MPEG is editable and provides the high visual quality required by movie-type applications. Indeo compression, offered by Intel, is optimized for low decode processing requirements.

In order provide an appreciation of video compression algorithms, an overview H.261 will be given. Audio compression will not be discussed in detail since it uses the same basic principles used for video compression. A good reference for their details is [Rettinger, 95].

The H.261 is a widely used international video compression standard for videoconferencing that is designed for applications which use synchronous circuit switched networks as their transmission channels, e.g. ISDN. It was approved by the International

Telecommunication Union (ITU), (formerly CCITT) in 1990, and is currently used in conjunction with H.320, H.323 and H.324. H.261 is an interoperability standard that pertains to communication between encoders/decoders (codecs) used by videoconferencing systems. It is often called Px64, where P (1–30) represents multiples of frames sent at 64Kbps. H.261 is similar to other "lossy" compression standards like JPEG, MJPEG and MPEG. Although similar to MJPEG and MPEG, JPEG is a compression standard used for still pictures, whereas MPEG and H.261 deal with motion video. Motion JPEG (MJPEG) generally uses H.261 techniques, such as Discrete Cosine Transform (DCT) encoding, quantization, macroblocks, etc. Using "lossy" compression algorithms, H.261 has provided a major advantage in dealing with the bandwidth constraints of various transmission media, without losing any significant picture quality (as least as far as the human eye is concerned). Although both MPEG and H.261 handle motion pictures, MPEG is designed to handle compressed bitstreams for the moving picture components of audio/visual services at rates from 0.9 to 1.5 Mbps. H.261, designed to target videoconferencing applications where motion is naturally limited, is specified from 64 Kbps to approximately 2 Mbps.

Due to the computation-intensive algorithm used in codecs, in the early videoconferencing systems they were implemented in a separate piece of hardware. With today's more powerful processors, however, the computations can be done by the computer's onboard processor.

H.261 uses Discrete Cosine Transform (DCT), to take advantage of the intraframe spatial and interframe temporal redundancy found in picture data. Spatial redundancy keeps track of the similarities in information in the same picture frame. It relies on a small number of bits to describe areas (pixels) on a picture that are the same color, therefore eliminating the need to code each pixel for every transmission of data across the channel. Temporal redundancy, using motion compensation, takes advantage of similarities of information between adjacent frames in a group of moving pictures, therefore only pixels that have changed from one frame to the next are transmitted. In summary, DCT gets rid of redundant data bits in each block of picture frame data.

H.261 also takes advantage of limitations in the human eye. Even though NTSC's standard for transmitting moving pictures is 30 frames per second, the human eye can only discern movement up to about 24 frames per second. Actually, for the human eye even 15 –25 frames per second is considered smooth motion.

Using "lossy" compression algorithms in H.261 has provided a major advantage in dealing with the bandwidth constraints of various transmission mediums, without losing any significant picture quality.

1. H.261 Structure

Figure 3-1 depicts a flow diagram of a typical H.261 standards based system encoder.

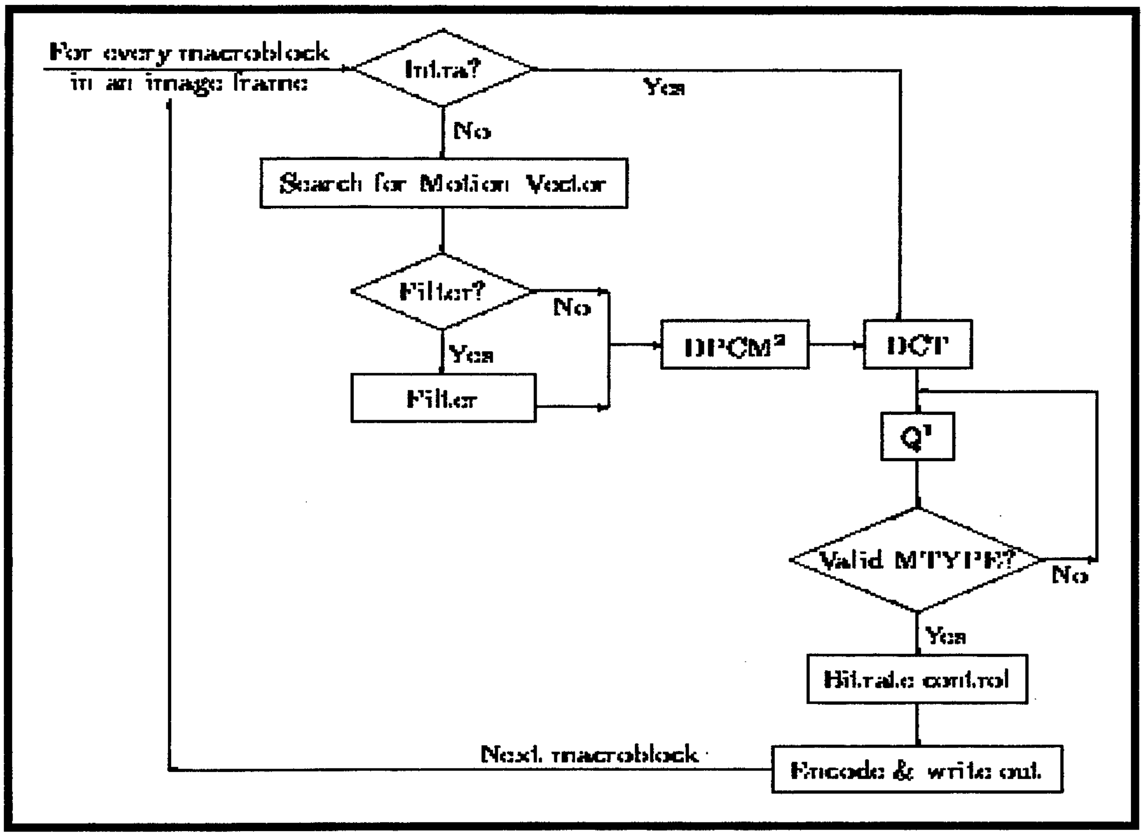

Figure 3-1 Encoder Flow Diagram [Jin, 96]

Except for the first frame, when a picture sequence is sent to the encoder, it figures out whether the reference frame is going to be from the present picture frame or the previous frame. If the reference frame is the present frame, intra-frame (I-coding) will be performed. When using I-coding, the data will go directly through a discrete cosine transform (DCT) where it will be transformed from the spatial to the frequency domain. The DCT coefficients are then sent to a quantizer where each coefficient is expressed as a level from a finite number of predetermined levels. After quantization, a decision is made to determine if the current macroblock (8 x 8 pixel array) is valid. Bit-rate control is performed, and eventually the bits are encoded and transmitted.

If the reference frame is going to be from the previous frame, inter-frame (P-coding) is performed. Here, a motion vector search is performed to determine the right direction to begin the search for the nearest, most similar macroblock between the current (target) and previous (reference) frames. After a match is found, either no filtering (subtract the pixel values of the matched macroblock in the previous frame from those in the current macroblock) or filtering (subtract the pixel values of the filtered matched macroblock in the previous frame from those in the current macroblock) is performed. A differential pulse code modulator (DPCM) codes the difference between the successive values instead of coding the actual values.

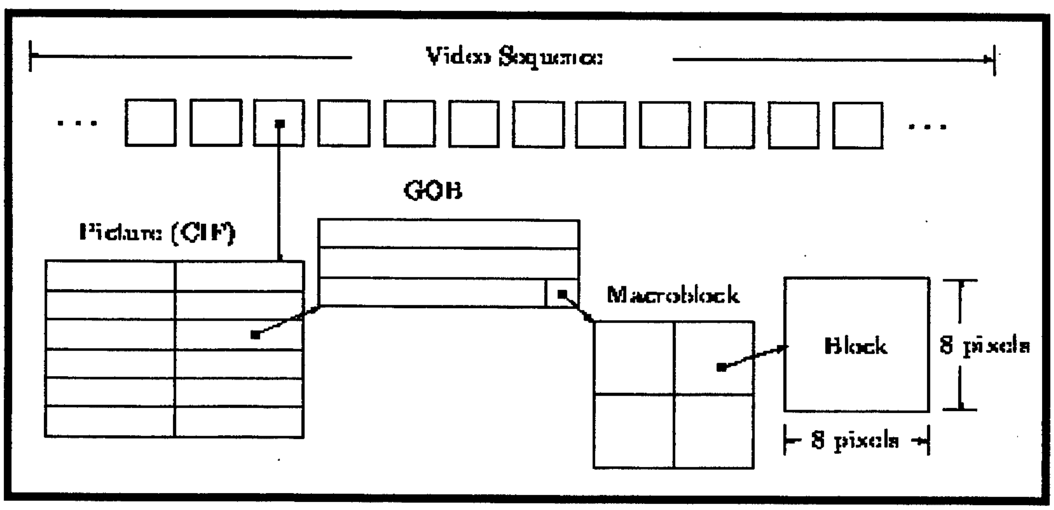

Figure 3-2 H.261 Data Structure [Jin, 96]

As shown in Figure 3-2, an H.261 video sequence begins with the picture frames, followed by Group of Blocks (GOB), Macroblocks, and Blocks. Each picture is divided into twelve or three GOBs for the CIF or QCIF frame format, respectively. Thirty-three macroblocks are organized in a fixed 11 x 3 format to form a GOB.

As shown in Figure 3-3, each macroblock consists of four 8 x 8 luminance (brightness) and two 8 x 8 chrominance (color) blocks.

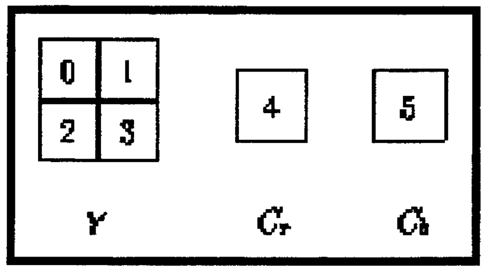

Figure 3-3 Macro Block Structure [Jin, 96]

2. Discrete Cosine Transform (DCT)

H.261 uses Discrete Cosine Transform (DCT), a form of frequency transformation which converts a signal from its spatial domain to its frequency domain in order to take advantage of the spatial and temporal redundancy in the picture data. Spatial redundancy keeps track of the similarities in information in the same picture frame. It relies on a small number of bits to describe areas (pixels) on a picture that are the same color, therefore eliminating the need to code each pixel for every transmission of data across the channel. Temporal redundancy takes advantage of similarities of information between adjacent frames in a group of moving pictures, therefore only pixels that have changed from one frame to the next are transmitted. Since the eye is more receptive to luminance than is to chrominance, bit representations of luminance both contain more bits and are sampled more frequently than the color components, which tend to be noisy.

In H.261, a two-dimensional DCT is performed on 8 x 8 pixel blocks (luminance and chrominance). Unlike the Discrete Fourier transform, all multiplications in the DCT use only real values, thus lowering the number of required computations. The 8 x 8 array is inputted into the DCT, and the output is an 8 x 8 array of DCT integer coefficients, with the number of nonzero values significantly decreased. This reduction in nonzero values is only the first part of the compression. For most images, much of the signal energy is in the lower frequencies, which appear in the upper left corner of the DCT array. The lower right values represent higher frequencies, and are often small enough to be neglected with little visible distortion. Figure 3-4 is mathematical model of the two-dimensional Discrete Cosine Transform (DCT).

$$\boxed{B(k_1,k_2) = \sum_{i=0}^{N_1-1} \sum_{j=0}^{N_2-1} 4 \cdot A(i,j) \cdot \cos\left[\frac{\pi \cdot k_1}{2 \cdot N_1} \cdot (2 \cdot i + 1)\right] \cdot \cos\left[\frac{\pi \cdot k_2}{2 \cdot N_2} \cdot (2 \cdot j + 1)\right]}$$

Figure 3-4 Two dimensional Discrete Cosine Transform [Jin, 96]

3. Quantization

The degree of quantization determines the image quality. A large quantization step size can produce unacceptably large image distortion. Similarly, too fine a step size can lead to lower compression ratios. The key challenge is to qauntize the DCT coefficients the most efficiently. H.261 does this by taking advantage of the limitations in the human eye's ability to discern high frequencies. The quantization matrix is an 8 x 8 matrix of step sizes (quantums), which provides an element for each DCT coefficient. As mentioned previously, step sizes in the upper left (lower frequencies) of the DCT array are small and are large in the lower right (high frequencies). The quantizer divides the DCT coefficients by their corresponding quantum and then rounds to the nearest integer. Large quantums drive the small coefficients down to zero, with the result that many high-frequency components easier to encode. The low-frequency components undergo only minor adjustments. Eventually, only the nonzero DCT coefficients that survive the quantization stage are encoded and transmitted. This quantizing is somewhat analogous to Mu-law and A-law non-uniform quantization, where the voice frequencies at the lower amplitudes (which we are more likely to encounter) will be conditioned to provide more information at a slight cost to information at higher amplitudes.

4. Motion Compensation and Estimation

When the motion of the source is generally limited, it is very likely that the luminance and chrominance blocks are not that much different between successive picture frames. In H.261, motion prediction is done on the luminance channel on blocks of 16 x 16 pixels. There are two aspects that cover these similarities; motion prediction and motion compensation. Motion prediction is performed at the encoder to determine what the motion vector should be, whereas motion compensation consists of moving blocks of data around, based upon that motion vector. As shown in Figure 3-5, by vectoring the reference block and comparing its bit structure with the bit structure in the target block, it looks for the closest match. Consequently, only the difference in the pixel values between the current macroblock and its matched macroblock are encoded.

The reason why motion compensation is effective is because it moves only the section of video where motion has occurred, rather the entire video area for every frame. Essentially each frame can be reasonably coded by detecting the changes (which are usually very small) from the previous one. This functions is a very important aspect in lowering the bit rate. Before a reference frame can be established, intra-frame coding must be done.

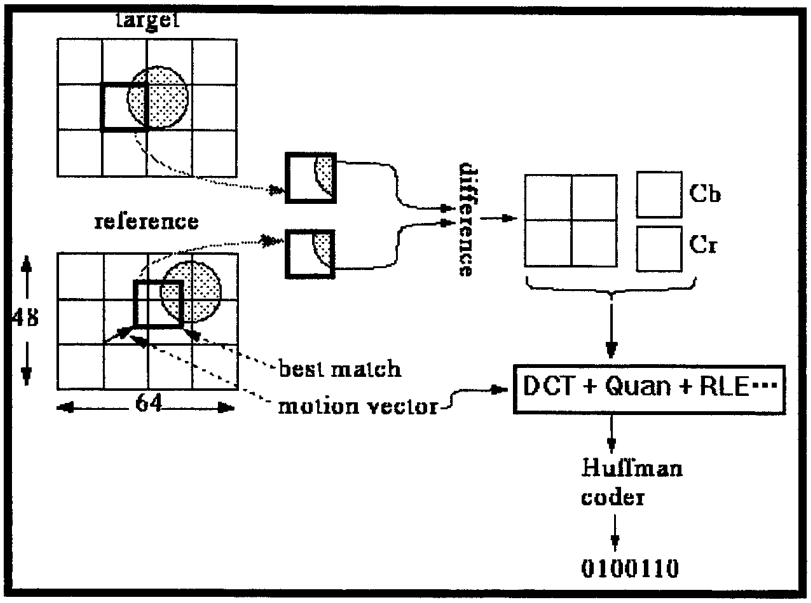

Figure 3-5 P-coding (interframe) [Jin, 96]

Figure 3-6 shows how each macroblock is intra-frame encoded. The intra-frame is used as an accessing point. Figure 3-7 shows the frame sequencing used in H.261.

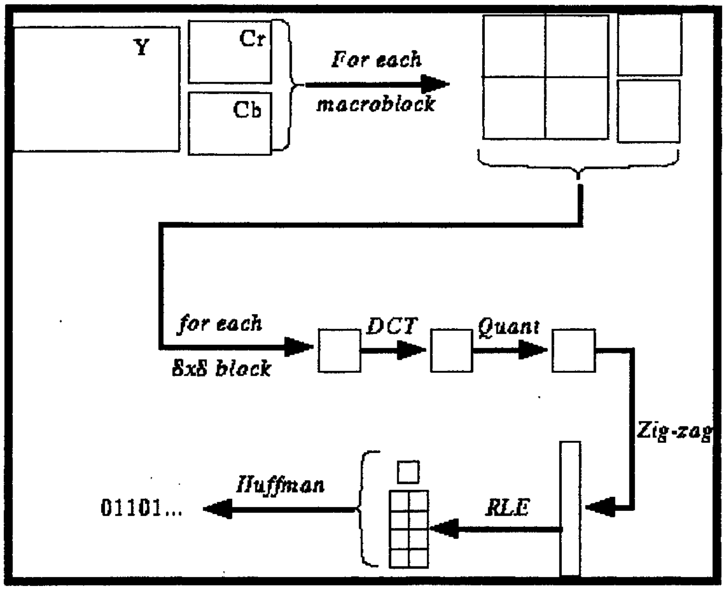

Figure 3-6 I-coding (intraframe) [Jin,96]

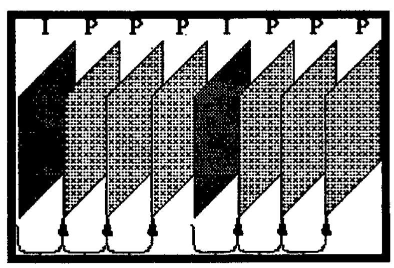

Figure 3-7 H.261 frame sequence encoding [Jin,96]

After quantization, it is not unusual for more than half of all of the DCT coefficients to be equal to zero. One coding scheme, run-length coding, is used to take advantage of this. In run length coding, except for the DC coefficients of the intra-coded blocks, all DCT coefficients are encoded using the run-length algorithm in a ziz-zag fashion, as shown in Figure 3-8. For each non-zero value, the number of zeros that preceded the number and the amplitude of the number itself form a pair. If the last nonzero value does not happen to be the last coefficient in the block, an End-of-Block code is attached to tell the decoder that there are no more nonzero coefficients left in the 8 x 8 block.

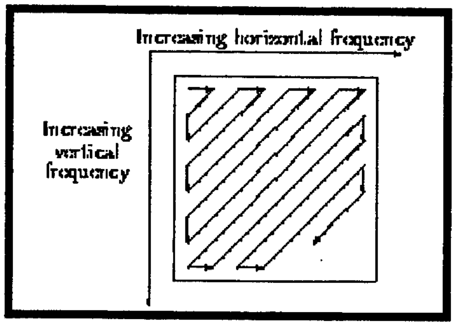

Figure 3-8 Run-Length Encoding [Jin,96]

The coded pair will then go through a variable length encoding where each pair has its own code word, assigned through a variable length code. The basic idea is to assign shorter code words to represent more frequently occurring values and longer code words to the less frequent values, in order to compress data even further. Huffman coding is the most common. Many Huffman tables used for different types of data are specified in the H.261 standard.

H.261 is only the baseline video compression standard for videoconferencing. There are many faster and more efficient codecs, which are H.261 compliant, that use their own proprietary algorithms. Nevertheless, even a minimum H.261 compliant codec can provide tremendous compression ratios (well beyond 100:1). The table in Figure 3-9 shows how well data rates can be increased with a 100:1 data compression using H.261 compression standard.

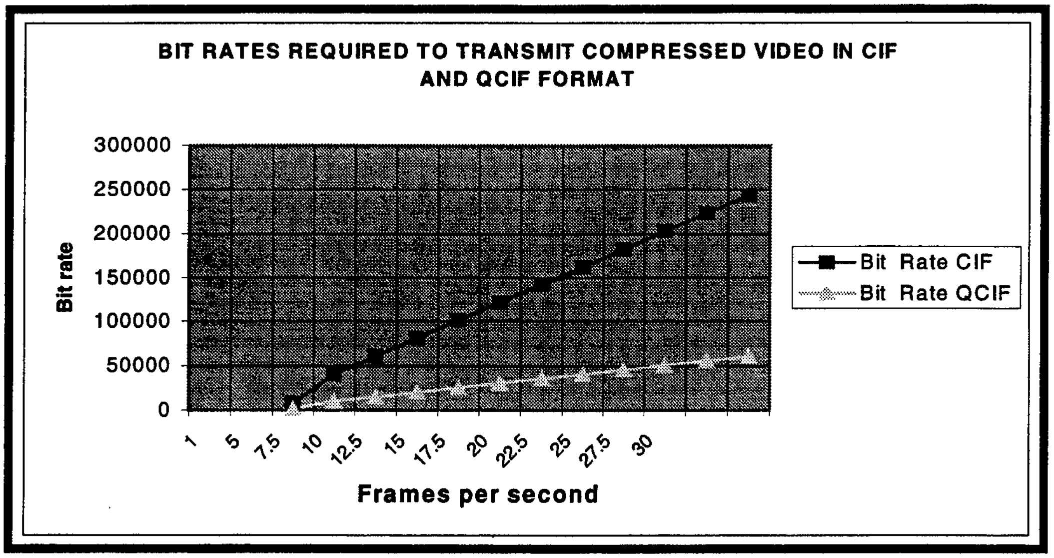

Figure 3-9 Frame Rate vs. Bit Rate for compressed data

5. MPEG

Many of the compression techniques used in the H.261 standard are similar to those used in the MPEG-1, but there are three major differences: data structure, coding type, and frame ordering [Zin, 96]. Because MPEG is targeted for more bandwidth-intensive applications than H.261, this thesis will not provide and in-depth description of MPEG standards.

G. AUDIO COMPRESSION

Audio compression standards are the most important function of videoconferencing systems, across all generations. Currently, Mu-law and A-law are the most common compression techniques used to condense audio data utilized in videoconferencing systems.

Both are non-uniform pulse code modulation (PCM) encoding techniques that use the quantized values of the samples in order present a discrete representation of the audio signal. Each sample represents a code word that is 8 bits in length. Mu-law and A-law transformations allow 8 bits per sample to represent the same range of values that would be achieved with 14 bits per sample using uniform PCM, which translates to a compression ratio of approximately 1.75:1. Due to the logarithmic nature of the transformation, the low amplitude samples are encoded with greater accuracy than the higher samples.

Major techniques that are designed for audio signals:

- G.711 - 48 – 64 Kbps Narrow-band
- G.722 - 48 – 64 Kbps Wide-band
- G.723 – Speech coding at 5.3/6.4 Kbps
- G.728 - 16 Kbps Narrow-band

ITU-T recommendation G.711, "Pulse code modulation of voice frequencies" provides telephone quality audio (narrow-band 3khz).

G.722 provides stereo quality (wide-band 7khz). At a typically higher data rate, usually > 256 Kbps, it provides the best audio quality available.[VTE, 95] G.722 uses adaptive differential pulse code modulation (ADPCM), which uses predictive algorithms to predict the values of adjacent samples. It uses the difference between the predicted and actual sample and encodes the difference. The adaptive part is because the encoders can also adapt to changing quantizing or prediction parameters. ADPCM generally achieves ratios of 2:1 as compared with Mu-law or A-law 1.75:1. G.722 has three modes of operation: 64, 56, and 48 Kbps. If a 64 Kbps communication channel id used, 48 or 56 Kbps modes will have an additional 8 or 16 Kbps of bandwidth for other data.

For audio over narrow-band POTS lines, there's G.723, which supports a compressed 3.4khz signal. If defines speech coding for audio transmitted at 5.3/6.4 Kbps.

G.728 provides narrow-band audio, which is important for lower bit rates < 256 Kbps. It is designed specifically for speech signals. G.728 uses another type of predictive coding called code excited linear prediction (CELP), which requires a bandwidth of 16 Kbps and is very computationally complex, requiring special hardware.

As described in H.320, if two different classes of audio compression are used, the less capable of the two will be used. For example, if a Class 3 system (G.728) establishes a call with a Class 1 system, the audio will be G.711. [VTEL, 95]

H. DATA STANDARDS

The T.120 standard focuses on collaborative computing, common whiteboard, and applications sharing during any H.32x videoconference. It defines the communication and application protocols and services that support real-time multipoint data communications. The specification also allows data-only T.120 sessions, when no video communications are required. In addition, T.120 supports multipoint meetings with participants using different transmission media. T.120 recommendations include:

- T.122 Multipoint Communication Service
- T.123 Network Specific Transport Protocols
- T.124 Generic Conference Control
- T.126 Still Image Exchange

I. SUMMARY

As network architectures have evolved, newer standards are continually implemented. But in order to provide cross-platform capability, flexibility, scalability, and accommodation of newer technologies as they emerge, the protocols and standards used in videoconferencing for distance learning must be compatible with the standards from the International Standards bodies. These standards should be the baseline used in videoconferencing systems for distance learning.

Using commonly available software codecs, not only will network bandwidth improve over already strained data pipes, but the allows for storing more data in a PC's storage device(s). This provides the ability for more course material to be streamed-on-demand, providing the asynchronous capability necessary for distance learning to sea.

Although software video codecs lack the compression speed of dedicated codecs, they have the advantage of low cost. Furthermore, more powerful processors like Intel's Pentium II with MMX technology improve video compression/decompression.

IV. IP MULTICASTING AND THE MBone

A. INTRODUCTION

This chapter focuses upon multicasting videoconferencing sessions over IP-based networks. It must be noted however that IP is very flexible. It can be used over a variety of network segments, including ATM, frame relay, switched multimegabit data service (SMDS), satellite, dial-up asynchronous, and ISDN. This chapter also discusses the major protocols supported by The IP Multicast Initiative (IPMI). Founded in 1996, the IPMI is a multi-vendor cooperative effort to promote the deployment of industry-standard IP Multicast technology, many of which are IETF Requests for Comment (RFC). Many members are leaders in the high technology industry including IBM, Intel, Microsoft, Cisco Systems, Silicon Graphics, and GTE, among others.

B. BACKGROUND

As shown in Chapter II, videoconferencing compression algorithms help reduce network bandwidth requirements, allowing videoconference applications to deliver real-time, quality video and audio data across networks. But compression solves only one area of the bandwidth issue. For example, what if a videoconferencing application needed to send data to multiple hosts simultaneously? One way to accomplish that task would be to retransmit identical IP packets to each recipient. If there are many recipients, this could potentially strain the network. To avoid this problem, the Internet Engineering Task Force (IETF), an arm of the Internet Architecture Board (IAB) that approves Internet standards, endorsed IP multicast as a standards-based solution to this problem. There are two items that make multicasting practical on the Internet. They are the lack unlimited bandwidth on the Internet backbone connections, and the widespread availability of workstations across a wide global network infrastructure [Macedonia, Brutzman 94].

C. IP MULTICASTING

RFC 1112, "Host Extensions for IP multicasting," authored by Steve Deering in 1989, was designed as an extension of IP Version 4. It is described as "the transmission of an IP datagram to a "host group", i.e. a set of zero or more hosts identified by a single IP destination address [Johnson, 97]. IP multicast allows applications to send data over the Internet to many simultaneous recipients in a more economical fashion than unicast or broadcast IP transmissions. Unicast IP is from a single source to a single destination (one-to-one), so in order to send information to multiple recipients using unicast, an application needs to send multiple copies of IP datagrams, which might saturate the transmission medium. Broadcast IP sends data to all of the participants in a network whether they want it or not.

When Internet Protocol (IP) was developed, Class D IP addressing was designed to facilitate multicasting. Unlike unicast IP addresses, which identify specific destinations, Class D addresses identify a particular transmission session. Class D addresses are reserved for groups rather than individual hosts. The addresses range from 224.0.0.0 to 239.255.255.255.

There are also certain special addresses (listed in RFC 1700 – "Assigned Numbers"):

- 224.0.0.1, the "all host group" -- addresses all multicast hosts on a directly connected net.

- 224.0.0.2 addresses all routers in a LAN.

- 224.0.0.0 through 224.0.0.225 is reserved for routing protocols and other low-level topology discovery or maintenance protocols.

- 224.0.1.3 through 224.0.13.255 is reserved for Network News.

With IP multicast, the source application is not necessarily aware of the destinations. Multicast applications send one copy of an IP packet over the network to a group address. A group of receivers may then participate by joining the particular multicast session group. The multicast IP datagram is delivered to all members of its destination host group (group Class D address) with the same 'best effort' reliability as regular unicast IP datagrams [Johnson, 97].

Some of the rudimentary requirements of IP multicast are:

- Since hosts may leave or join a group at anytime, membership in a host group of an IP multicast session must be dynamic.

- There should be no restrictions on the location and number of groups that can participate.

- At the application level, a host may have multiple data streams on different port numbers, on different sockets, in one or more applications [Johnson, 97].

The minimal hardware/software requirements needed to deliver IP multicasts end-to-end are:

- Support for IP multicast transmission and reception in the host's TCP/IP protocol stack and operating system.[1]

- Software supporting Internet Group Management Protocol (IGMP), in order to communicate requests to join a multicast groups(s), and receive multicast traffic.

- Network interface cards that efficiently filter for LAN data link layer addresses that are mapped from network layer IP multicast addresses.

- IP multicast application software such as videoconferencing or file transfer. The end-node applications should be flexible in terms of their support for existing compression technologies and accommodation of newer technologies as they emerge.

- Intermediate routers between the sender(s) and receivers(s) must be IP multicast-capable.[2]

- Firewalls (i.e. packet-filtering software) may need to be reconfigured to permit IP multicast traffic. [Johnson, 97]

Figure 4-1 is an overview of the requirements.

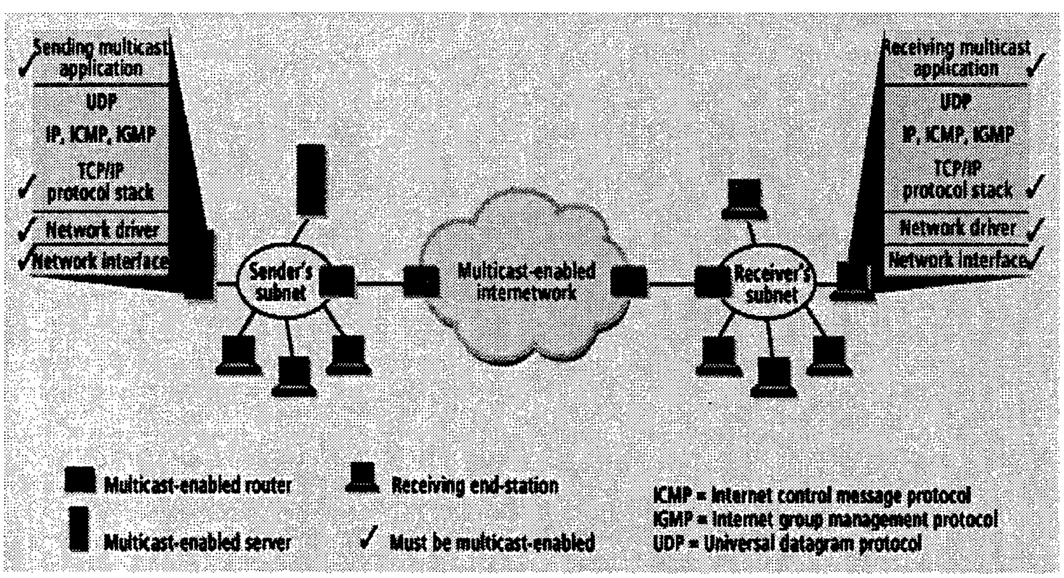

Figure 4-1 Requirements for IP Multicasting

[1] Windows NT, Windows 95, and the latest versions of UNIX support IP multicast.
[2] Multicasting capability can be enabled in most routers by simply updating the software and adding memory.

When a host application requests membership in the host group associated with a particular multicast session, the request is communicated to the subnet's multicast router and, if necessary, on to intermediate routers between the sender and receiver. When the requested session is found, the router delivers the requested incoming multicast IP datagrams to the requesting host, passing it to the TCP/IP stack, which makes the data available as input to the user's application. Other stations filter out multicast packets at the hardware level.

Multicast routers do not need to know the list of member hosts for each group. It only requires knowing a group for which there is one member on the subnet. A multicast router attached to an Ethernet need associate only a single Ethernet multicast address with each host group having a local member.

1. IP Multicast Protocols

Like any other means of transporting data over network infrastructures, IP multicast comes with an array of protocols that help provide the framework for multicasting IP datagrams. The most fundamental of IP multicast protocols, Internet Group Management Protocol (IGMP Ver. 2), described in RFC 2236, is used by multicast routers in order to learn the existence of host group memberships. It is the baseline protocol necessary to conduct an IP multicast session.

The protocols used to ensure that the needed bandwidth and QoS are available include Real-Time Transport Protocol (RTP), Real -Time Control Protocol (RTCP), Real-Time Streaming Protocol (RTSP), and Resource Reservation Protocol (RSVP). There area also associated routing protocols such as Protocol Independent Multicast (PIM), Multicast

Open Shortest Path First (MOSPF), and Distance Vector Multicast Routing Protocol (DVMRP).

There are also transport issues that need to be addressed with IP multicast. Applications that are IP multicast capable are not designed for use with reliable, connection-oriented transports (TCP), therefore layer 3 does not invoke destination addresses in the datagrams. They also do not require guaranteed in-sequence delivery of IP packets. Furthermore, since the delivery of IP will not have a fixed path, there is no assurance that the bandwidth needed for video and audio will be available. Videoconferencing applications are better off tolerating missing data than overcoming the lengthy delays caused by TCP retransmissions. Therefore, a simpler transport framework, such as User Datagram Protocol (UDP), a transport layer protocol that only provides error detection, does a more than an adequate job of transporting videoconferencing data.

a. Internet Group Management Protocol (IGMP)

Internet Group Management Protocol (IGMP) performs two main functions. It is used by hosts to join IP multicast sessions, and by multicast routers to learn the existence of host group members on their directly attached subnets, identify designated multicast routers in a LAN, and propagate group information over the Internet. It is loosely analogous to Internet Control Message Protocol (ICMP), which is used in PING applications. [Johnson, 97]

Each multicast router sends IGMP queries (Host Membership Query), and the hosts respond by reporting their host group memberships (Host Membership Report). This query and response session is accomplished by IGMP messages encapsulated in IP datagram packets. To determine if any hosts on a local subnet belongs to multicast group, one multicast router per subgroup periodically sends a hardware (data link layer) multicast IGMP Host Membership Query (network address 224.0.0.1) to all IP end nodes on its subnet. This message asks them to report back on the host group memberships of their processes. These query messages have a time to live (TTL) of 1 to limit their transmission to the network directly attached to the router. [Petitt, 96]

Each host then sends back one IGMP Host Membership Report to the group address, so that all group members see it. When hosts see a Host Membership Report for the group transmitted, they cancel their own transmission. Hence, only one member of the group will report membership to the router for a particular group address. Periodically, local multicast routers will send IGMP Host Membership Queries to the "all hosts" group, to verify current memberships. Although IGMP packets are routinely transmitted, compared to the multicast application's traffic, its bandwidth use is insignificant. Figure 4-2 shows an IGMP request on a LAN.

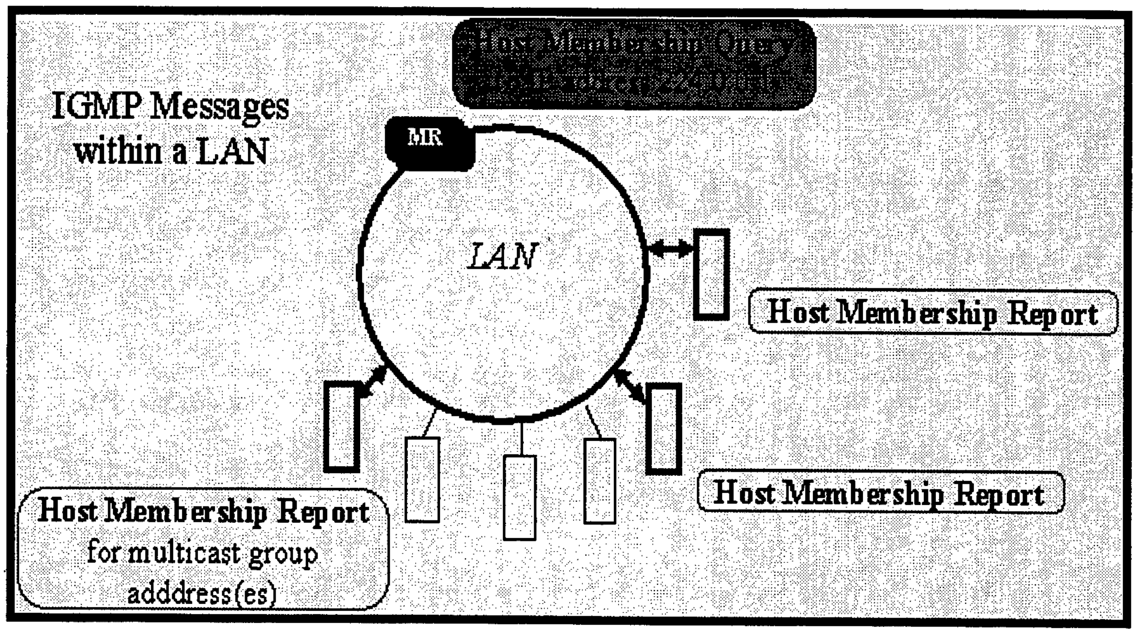

Figure 4-2 IGMP Messages on a LAN [Johnson, 97]

When the last station on a subnet leaves a multicast group, the router "prunes" the multicast data stream associated with it by ceasing to forward the data stream to subnet.

b. Real-Time Transport Protocol Version 2 (RTP)

Real-Time Transport Protocol (RTP), defined in RFC's 1889 and 1890, provides end-to-end delivery services to support applications transmitting real-time data.[Johnson,97] Among the services that RTP provide are payload type identification, packet sequence numbering, and time stamping. The delivery of RTP packets is monitored by Real-Time Control Protocol (RTCP), which is discussed later.

RTP does not provide all of the typical functionality of typical transport protocols. It is a header format running in combination with other transport protocols in order to take advantage of their functionalities. The RTP header provides timing information to synchronize and display audio and video data, and also to determine if packets are lost or arrive out of order. In order to allow multiple data and compression types, the header specifies the payload type by characterizing what type of audio and video encoding is carried in the RTP packet. This enables users to have the option to change the encoding methods during a conferencing session, in response to network congestion, or to accommodate low-bandwidth requirements of a new conference participant [Johnson, 97].

RTP does not ensure timely delivery or provide QoS guarantees. It does not guarantee delivery or prevent out-of order delivery, nor does it assume that the underlying network is reliable. For applications like videoconferencing that require these types of guarantees, RTP must be accompanied by other mechanisms [Johnson, 97].

c. *Real-Time Control Protocol (RTCP)*

RTCP, also standardized in RFC's 1889 and 1890, is a control protocol that works in conjunction with RTP. The information, periodically transmitted by each participant in an RTP session to all other participants, is used by the applications to control the performance of the conference and for diagnostic purposes.

RTCP performs four primary functions. a) First, RTCP provides feedback information about the quality of the transmission to the applications. The statistics include the number of packets sent, the number of packets lost, interval jitter, etc. b) RTCP also identifies the RTP source address through its transport-level identifier called the canonical name (CNAME). The CNAME is used to keep track of participants in a session in order to synchronize audio and video. c) RTCP controls its transmission intervals in order to prevent control traffic from overwhelming network resources. RTCP control traffic is limited to five percent of the overall session traffic. This control on RTCP allows RTP to scale up to a large number of session participants. d) An optional function can be used to convey a small amount of information to all session participants. In distance learning, this information can be used to identify the participants in a particular training session. For example, RTCP might carry a personal name to identify a participant on the user's display. [Johnson, 97]

Since RTCP sends feedback to all of the recipients of a multicast stream, individual users can determine if a problem is specific to the local end node or system-wide. RTP and RSVP information is simply data from the point of view of the routers that move the packets to their destinations. To prioritize data streams and provide a guaranteed quality of service, other protocols must be used [Steinke, 96].

d. Resource Reservation Protocol (RSVP)

In an Internet environment with a myriad of routers and switches, packet queuing can lead to variable packet delivery delays in different parts of the network. QoS considerations for a multicast application include tolerance to jitter, delay, and lost packets. In order for the network to provide QoS, applications must be able to reserve and control network services [Johnson, 97]. This is not an issue on networks with sufficient bandwidth, but considering the packet-based networks targeted for use in this thesis, QoS is a major issue.

The Resource Reservation Protocol (RSVP) is a draft protocol for resource reservation, still under development [Hurwitz, 97]. Elementary RSVP requests consist of dynamic request specifications for end-to-end desired QoS and definitions of the set of data packets to receive the QoS. It aims to efficiently set up a guaranteed QoS resource reservation, supporting unicast and multicast routing protocols, and is expected to scale well for large multicast delivery groups. RSVP is useful in environments where QoS reservations can be supported by reallocating (rather than adding) resources. In IP multicast, a host sends an IGMP message to join the group and then sends an RSVP message to reserve resources along the delivery path(s) of that group. The RSVP service request is initially sent to a local server. The local server will validate the request and then forward the request.

RSVP promises access to Internet integrated services. The hosts and the network work together to achieve guaranteed quality of end-to-end transmission. However, in order to achieve end-to-end QoS, all hosts, routers and other network infrastructure elements between the receiver and sender must support RSVP. They must reserve system resources such as bandwidth, CPU and memory buffers in order to satisfy QoS reservations. RSVP rides on top of IP, and is used by routers to deliver QoS control requests to all nodes along the path(s) and to establish and maintain statistics in order to provide the requested services. After the reservation has been made, the router supporting RSVP determines the route and QoS class for each incoming packet and the scheduler makes forwarding decisions for every outgoing packet [Johnson, 97].

Since RSVP is receiver initiated, resource requests are in only one direction. At each node along the reverse path to the receiver, RSVP attempts to make a resource reservation for the requested stream. This receiver-initiated propagation delivers control messages only up to the node of the spanning tree where they merge with another reservation for the same source stream, thus preserving bandwidth. This receiver initiation achieves two goals: scalability, because the receiver-initiated joining delivers control messages only along those parts of the tree that need the information; and heterogeneity, because of the receiver orientation, individual receivers can choose to participate and request different levels of reservation. [Precept, 97]

Based upon the admission and policy controls of the underlying hardware, at each node, one of two general actions take place: The host makes a reservation or forwards the request upstream. These controls are not a part of RSVP, but are utilized by the equipment. Admission controls determine whether the node has sufficient resources, and the policy control determines whether the user has authorization to make a reservation. If the reservation is rejected, RSVP returns an error message to the appropriate receiver(s). If accepted, the node is configured to provide the desired QoS. If the RSVP request is forwarded upstream, it continues to propagate along the reverse path towards the appropriate senders. [Johnson, 97]

One drawback of RSVP is the computational requirements required by routers to inspect and handle packets in a priority order. Approaches such as tag switching are being developed to help with this drawback. Another area of research is enhancing RSVP to use routing services that provide alternate and fixed paths. Finally, RSVP has no way to handle network overload that may occur if multiple users request the maximum bandwidth at the same time [Andrews, 97].

RSVP continues to be under review by the Internet Engineering Task Force (IETF), and is not widely deployed. Similar work has been done on Internet Protocol-Version 6 (IPv6) to support resource reservation and flow set up for multicasting. Figure 4-3 is an illustration of the method.

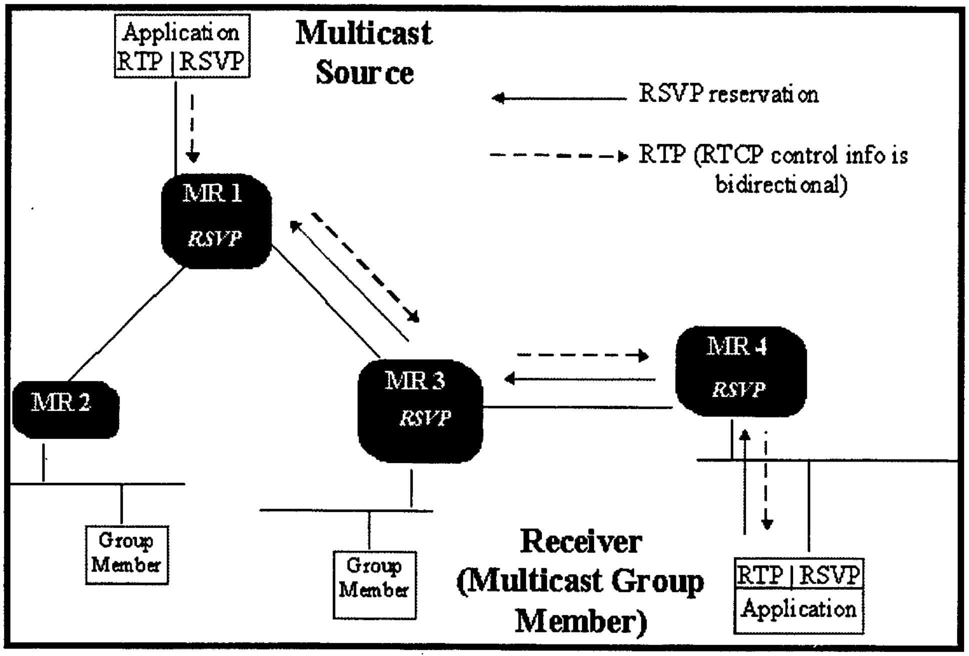

Figure 4-3 RSVP Protocol [Johnson, 97]

e. Real-Time Streaming Protocol (RTSP)

RTSP is considered more of a framework than a protocol. It works at the application level for unicast and multicast streaming and to enable operability between different vendors' clients and servers. RTSP essentially encodes and passes multimedia stream control commands. In many respects, it resembles a protocol that describes the functionality of a VCR remote control.

2. Reliable IP Multicast

Reliable connectivity ensures that all packets are received by all of the recipients. For unicast IP services, error correction and detection is provided at the TCP layer. But such traditional techniques for error detection and correction in a large-scale multicast environment might result in an "ACK explosion" or a "NAK" implosion, where the excessively large numbers of acknowledgement messages from large groups can swamp the originating hosts sending the desired streams.

There are currently no IETF standards for reliable IP multicast, but several Internet drafts have been submitted related to reliable multicasting, and an IRTF (Internet Research Task Force) working group has been formed to advance reliable multicast standards efforts [Johnson, 97].

Cisco's proposed Pretty Good Multicasting (PGM) Reliable Transport Protocol is intended to make multicasting appropriate for mission-critical uses. Although this work is still under development, this protocol can be useful in areas such as common tactical pictures.

As mentioned previously, videoconferencing applications are able to tolerate missing data and still provide discernable video and audio. They also do not require guaranteed in-sequence delivery of IP packets. Therefore, videoconferencing end-systems will not need bit-perfect, in-order, acknowledged data. For military purposes, the multicast reliability requirement is more essential with the common tactical architecture and cooperative engagement issues. (Petitt, 96) evaluates the design choices of several reliable transport layer multicast protocols that support those requirements.

3. Group Setup Protocols

Users of videoconferencing must not only know about upcoming or current IP multicast sessions, but also how to manage and coordinate them. Parameters for sessions will include information such as the name and topic of the session; its multicast address; date, time and duration; media types (e.g. audio), media encoding, and media ports; security parameters; etc. There are currently several Internet drafts for these types of protocols, but a clear standard has not emerged. Still there are current tools available. For example, the session directory tool, sdr, is widely used on the MBone. Similarly, Precept's IP/TV Program Guide has a directory embedded in a Web page.

4. Other IP Multicast Issues

a. *Router Support*

As with routing any IP datagram, multicasting requires routers to interact with each other and exchange information about their neighbors. One item that should be considered, in order to most effectively implement IP multicast, is to determine what is the best possible routing protocol based upon the network layout. On a routed network, which includes native multicast, IP multicast traffic for a particular source and destination group is typically transmitted via a spanning tree that connects all of the hosts in the group. There are basically two approaches to multicast routing; Dense-Mode or Sparse-Mode.

Dense-Mode multicast routing protocols follow an approach that assumes that the multicast group members are densely distributed throughout the network and bandwidth is abundant. These protocols rely on periodic flooding of the network with multicast traffic to distribute group membership information to all nodes in the network in order to set up and maintain the spanning tree. The protocols include Multicast Open Shortest Path First (MOSPF), described in RFC 1584, Protocol-Independent Multicast-Dense Mode (PIM-DM), and the earlier Distance-Vector Multicast Routing Protocol

(DVMRP), described in RFC 1075. DVMRP is currently used on the MBone, but is becoming obsolete.

Sparse-Mode protocols are based upon the assumption that the multicast group members are sparsely distributed throughout the network and bandwidth is not necessarily widely available. Flooding in this case is not economical because the waste of bandwidth and latency problems that occur when transmitting IP over large geographic regions. Sparse-Mode routing protocols like Core Based Trees (CBT), RFC 2189, and Protocol-Independent Multicast-Sparse Mode (PIM-SM), RFC 2117, are possible choices. They build a single distribution tree, which is formed around a focal router (called a core in CBT and rendezvous point in PIM-Sparse Mode). Multicast traffic for the entire group is sent and received over the same tree, regardless of the source. The use of a shared tree can provide significant bandwidth savings for applications that have many active senders.

Another concern is that many Internet Service Providers (ISP's) do not have a protocol to deal with inter-domain multicast routing (IDMR). IDMRs such as Protocol Independent Multicast (PIM), Multicast Open Shortest Path First (MOSPF), and Distance Vector Multicast Routing Protocol (DVMRP), were not designed for multiple autonomous systems that do not necessarily want to share all their routing information. [Hurwicz, 97]

Although Border Gateway Protocol (BGP) provides inter-domain routing capabilities for IP, there is no equivalent of BGP for IP Multicast. Currently the IETF's IGMP working group is developing a Border Gateway Multicast Protocol (BGMP) protocol specification. Until this shortcoming is addressed, the lack of an IDMR protocol limits to the scalability of IP Multicast, along with limited bandwidth, is one of the major reasons why MBone has only about 30,000 users. Furthermore, growth will continue to be limited if all of the routers will have to contain all of the routing information for the whole network [Hurwicz, 97].

Although new routers on the Internet are capable of supporting multicast, most are not IP multicast enabled, by default. Many ISPs are reluctant to deploy multicast because of concerns such as: cost and complexity of upgrading older routers, router resources consumed, reliability problems, an unclear business model (how does an ISP charge for traffic, who pays, and how does peering—communications between ISPs—work?), and lack of diagnostic/simulation/debugging tools. Even with these concerns, some ISP's have already deployed multicast. For example, UUNET offers IP multicast as a value-added service on its network. It has equipped each of its domestic Point-of-Presence (POP) with multicast routers, in order to provide multicast service connections throughout the continental United States. By next year, expect more ISPs to begin implementing multicasting, especially as backbone traffic continues to rise and cost threshold of user decreases.

There is also the issue of incorporating QoS routing with various multicast routing protocols. Native IP multicast protocols uses various approaches to construct delivery trees for efficient transmission. But without additional mechanisms, those routing approaches are not guaranteed to provide a specified QoS. For example, when QoS mechanisms are used to reserve and control network resources, the routers must not only satisfy the added QoS requirements, but in addition, it has to find the shortest path to a destination when constructing a delivery tree.

b. Other Network Issues

Many IP multicast implementations have not been thoroughly tested because many organizations have not enabled multicast capabilities in their networks [Hurwicz, 97]. Furthermore, there is no widely known data on how routers will react to a steady, high volume of multicast multimedia traffic. Because IP Multicast uses the connectionless User Datagram Protocol (UDP), the most popular type of firewall, application gateways, can not secure connectionless protocols, essentially rendering IP multicast incompatible with most firewall strategies. In some applications, in order to allow transmissions through a firewall, TCP is used in conjunction with UDP, by tunneling and the ported mulitcast routing program running on a host. Many firewall applications and routers will need to be reconfigured, replaced, or upgraded in order to deal with multicast address reliability and bandwidth issues.

D. MULTICAST BACKBONE (MBone)

When LANs, WANs, and the Internet were initially developed and designed, videoconferencing was not expected to be a viable possibility. Based of limited bandwidth, sending video or audio was not considered possible or practical. However, as the technology matured, the Multicast Backbone (MBone) and video/audio compression techniques were developed showing that videoconferencing was not only possible but also practical.

The MBone is an experimental, virtual network that lies on top of the Internet. It was initiated in early 1992 and named by Steve Casner of the University of Southern California Information Sciences Institute. It provides one-to-many and many-to-many network delivery services for multicast capable applications such as videoconferencing. MBone originated from a collaboration in order to multicast audio and video from meetings of the Internet Engineering Task Force, and has been the testbed for many of the multicast protocols mentioned earlier, such as IGMP, RTP etc. MBone is continually being developed by hundreds of researchers who are designing more effective and efficient protocols and applications for videoconferencing. This section gives a brief introduction to the MBone to provide an example of the viability of multicasting video and audio over IP-based network architectures.

1. MBone Requirements

The major technical prerequisite that makes multicasting possible over the MBone is the use of network routers called mrouters. Basically mrouters are upgraded commercial routers, dedicated UNIX workstation-class machines, or dedicated UNIX workstation-class machines running with modified kernels in parallel with standard commercial routers [Macedonia, Brutzman, 94]. More and more commercial routers are now supporting multicast. This will help eliminate the inefficiencies and management headaches of duplicate routers and tunnels [Macedonia, Brutzman, 94]. The mrouters use the IGMP protocol to learn the existence of host group membership on their directly attached subnets, to identify designated multicast routers in a LAN, and to propagate group membership information over the MBone. Tunneling further augments MBone by allowing multicast datagrams to be forwarded to other MBone subnets that support IP multicast. For example, at the sending mrouter, IP multicast datagrams are encapsulated by unicast IP datagrams and forwarded as unicast IP datagrams so that intervening unicast routers and subnets can handle them. The receiving mrouters will "strip" the multicast datagram of its encapsulated unicast IP datagram in order to determine if any of its attached hosts are requesting to join that multicast group.

As mentioned earlier, the overarching issue in videoconferencing is bandwidth. IP multicasting partly addresses this issue by enabling one packet of information to reach many destinations. For example, a 128-kilobit per second video stream (based the typical data rate of two channels of ISDN) uses the same bandwidth whether it is received by one location or 20. However, there is one disadvantage. If all mrouters permitted packets to touch every workstation in the MBone, video streams might potentially misspend valuable bandwidth by sending streams to LANs that are not participants. For that reason, controls are needed to limit the propagation of video stream packets across the MBone. Controls of multicast packet propagation are implemented two ways. MBone limits the time to live (ttl) of multicast packets or it uses complex pruning algorithms to adaptively restrict the transmission of multicast packets. [Macedonia, Brutzman, 94]. MBone protocol developers are successfully experimenting with automatically pruning and grafting subtrees, and thresholds can set maximum bandwidth limits. The truncation is accomplished by setting the ttl in a packet. The ttl is decremented, by one or more, each time it passes through an mrouter. For example, if ttl was set to 16, it would multicast on a smaller scale such as a school campus. If the ttl was 128, it could potentially traverse most of the subnets on the MBone. Adjusting the ttl can assist in limiting the transmission of video stream data to specific regions or areas. Consequently, effective controls over the MBone can save precious bandwidth that the uncontrolled transmitted packets might otherwise use.

In order to make the MBone community a viable and efficient topology, global coordination is used to minimize congestion on the Internet. To add a new node to the MBone, a new site announces itself to its Internet Service Provider (ISP) or the MBone mailing list. Then, the nearest network providers decide on the most advantageous path connection to minimize local or regional Internet traffic.

MBone uses various application tools in order for end-users to receive and deliver videoconferencing. The common applications are videoconference tool (vic), visual audio tool (vat), robust audio tool (rat), shared whiteboard (wb), and session directory (sdr). Vat is used for audio teleconferences. Shared whiteboard (wb), using T.120 protocols, can be used as a shared drawing surface, and it can be used to export and view postscript files. The sdr tool dynamically announces the availability of sessions by displaying active multicast groups. Sdr also launches multicast applications and automatically selects unused addresses for any new groups. Sdr makes announcements periodically over a well-known multicast address and port.

One of the first significant uses of the MBone came about when NASA Select set up an in-house cable channel broadcast during space shuttle missions, which then could be viewed live from any MBone user's desktop computer.

Although many practical applications have been developed on the MBone, it continues to be used as a testing ground for IP multicast research and how it can be leveraged for distance learning. One thesis, *Internetworking: Economical Storage and Retrieval of Digital Audio and Video for Distance Learning*, [Tiddy, 96], investigates the usefulness and feasibility of applying networked storage of digitized video and audio, all via the MBone for distance learning. Currently there are prototypes that are being used to deliver stored digitized data over the MBone. The *Interactive Multimedia Jukebox* Project, which can be found at *http://imj.gatech.edu*, is a research effort to investigate the scalable delivery of video-on-demand (VoD) service using multicast communication. The MBone VCR on Demand Project, at *http://www.informatik.uni-mannheim.de/informatik/pi/projects/MVoD/*, offers a solution for the interactive remote recording and playback of multicast videoconferences.

E. MBone ISSUES IN DISTANCE LEARNING

Because it was originally a developmental tool, the MBone has seen limited use in the commercial environment, but it has already proved the great benefits of IP multicasting. It has great potential to grow and cover the entire Internet. Nevertheless, many network service providers have not enabled multicasting in many of their routers for various reasons. Among them is the lack of maturity of the technology, not being sure if ATM or IP (or combinations of both) is the direction to take, and pricing issues. Many regional network service providers still don't have an MBone connection.

MBone is not easy to set up. Enabling a router for multicasting and installing MBone tools is still something not normally done by network administrators. Many are leery about how video services will impact network bandwidth. Also, MBone tools are mainly developed for running on UNIX machines, and there are still problems porting the tools to Windows machines. Finally, the tools aren't as user friendly as some of the commercial products.

The commercial sector is discovering the viability of multicasting and is starting to develop tools that are based upon the MBone standards. Companies such as Whitepine's CU-SEEME and Precept's IP/TV already have MBone-compatible applications. Videoconferencing applications will continue to mature, and likely the myriad of standards will eventually converge. This process can be accomplished more easily if the newer products are based upon thoroughly evaluated tools.

F. SUMMARY

This chapter discusses the major multicasting protocols, technologies and issues that are pertinent to using videoconferencing as a part of distance learning. It describes the baseline issues that need to be addressed in order to multicast distance learning lectures to numerous recipients across an IP-based network to sea. These proven protocols will make videoconferencing over IP networks in DoD a practical solution. One primary reason is (as opposed to dedicated networks) that multicast groups can be dynamically set up and torn down. This flexibility is needed because of the constantly changing location of end-users such as those receiving distance learning at sea.

Standards like IP multicasting, and the future implementation of IPv6, will address some of the QoS issues by supporting resource reservation and flow setup. Also, as older routers are replaced or upgraded to support multicast, videoconferencing over the Internet and NIPRnet between groups at numerous locations will become commonplace. IPv6 is designed to help improve delivery of data at regular intervals, which will help address Quality of Service (QoS) issues. Its packet headers will help define the types of service (high quality paths in underlying network) that can be used for real-time delivery of audio and video.

This chapter has also shown that based upon the thorough testing and implementation of multicasting, it is clear that the hurdles currently facing IP multicasting widespread emplacement is deployment the rather than the technology.

V. IMPLEMENTING IP MULTICAST ACROSS THE NAVAL NETWORK ARCHITECTURE TO SEA

A. INTRODUCTION

This chapter provides an analysis of numerous options that can be used to leverage DISN for IP multicast. They will include desktop connectivity, the Unclassified but Sensitive Internet Protocol Router Network (NIPRnet), satellite entry points (gateways), Defense Satellite Communications System (DSCS) and/or C band SHF terminals (Challenge Athena), and Automated Digital Networking System (ADNS).

B. BACKGROUND

The goal of the Defense Information Infrastructure (DII) is to establish a seamless, secure, robust, agile, reliable and cost-effective telecommunications network that will serve as the end-to-end information transfer infrastructure for all DoD personnel and organizations worldwide [DISA, 96]. The Defense Information Systems Network (DISN) architecture, a component of the DII, is based upon a global network integrating existing Defense Communications Systems assets, Military Satellite Communications (MILSATCOM), Commercial SATCOM initiatives, leased telecommunications services, dedicated DoD Service and Defense Agency networks, and mobile/deployable networks; i.e. the consolidated worldwide enterprise level telecommunications infrastructure that provides the end-to-end information transfer component of the DII [DISA, 96].

Through the Defense Information Systems Agency (DISA), DoD is continuously identifying what architecture and standards DISN needs for a telecommunications infrastructure that can support voice and video. Currently, the Defense Video Service – Global (DVS-G), the transport network that DISN is used for videoconferencing, is mostly a collection of dedicated room-based systems whose terrestrial components are connected by ISDN services. One other segment of DISN that can be used to support videoconferencing is NIPRnet. NIPRnet is an IP-based network that consists of the wide-area and local-area network switching and transmission systems along with customer premises equipment (CPE) in order to provide connectivity to DoD users.

C. DESKTOP SYSTEMS CONNECTIVITY

1. POTS

Videoconferencing applications conducted on DISN over Ethernet, token-ring, or serial modem connections are straightforward. Under the DISN transmission services CONUS (DTS-G), AT&T provides information transport for the aggregate bandwidth of all customer Service Delivery Points homed off the Bandwidth Managers located in their respective access areas. Figure 5-1 is a diagram of the CONUS transmission service. To take advantage of the bulk transmission rates, AT&T bundles the access transmission into SONET for delivery to the Bandwidth Managers. At the customer access locations, transmission bandwidth interfaces at T1, T3 and SONET are provided. AT&T teams with Local Access Providers as required to accomplish the access area bandwidth requirements.

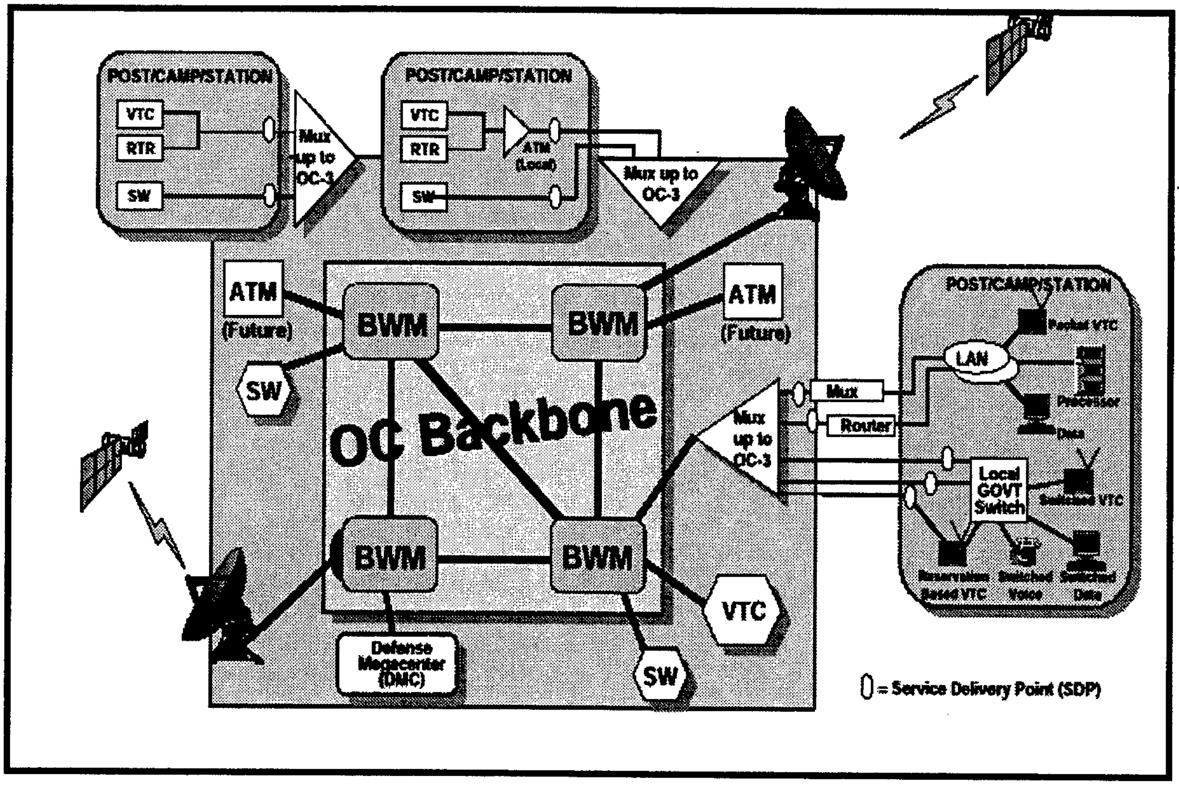

Figure 5-1 DISN Architecture [DISA, 96]

For POTS connectivity, commercial and DISN networks with dedicated dial up connectivity can be used. However, even with optimal desktop hardware and software, performance is always a question due to throughput problems associated with modem connections and dirty analog lines, which can cause bit errors and retransmissions.

2. Asynchronous Digital Subscriber Line (ADSL)

A twisted-pair phone line has a capacity far beyond the narrow 3-kHz channel used to carry an analog voice signal. Historically that capacity has not been used before, because it was reserved to compensate for signal loss in the line. A reemerging technology, Asynchronous Digital Subscriber Line (ADSL) overcomes this limitation and promises to provide download data rates up to 8Mbps to desktops, while transmission rates will be at least ten times of traditional modem data rates. ADSL is a modem technology that requires terminal devices at each end of the phone line (user to Local Exchange Carrier -- LEC). Because of the high frequencies ADSL uses, the distance between the modem and the central office plays a significant role in an ADSL modem throughput. The closer the modem is to the central office, the less signal degradation occurs. For example, the maximum distance from the central office for an 8Mbps download data rate would be approximately 1.7 miles, whereas 1.5 Mbps has a 3.4 mile limit.

Computer industry leaders such as Compaq, Intel, Microsoft and phone companies such as Ameritech, Bell Atlantic, SBC Communications, US West, Sprint and GTE have joined in an alliance to promote ADSL. ADSL technology has the potential to further enhance desktop videoconferencing by removing the bottleneck that currently plagues many users connected via standard POTS. Furthermore, what makes ADSL truly attractive is that the infrastructure required to support it, twisted-pair copper phone lines, is already in place. The current problems with ADSL are its lack of availability and high equipment costs.

3. Cable Modems

Cable Internet access is a relatively new transport technology that is still in its early stage of rollout. Except for the past year, phone companies had been slow implementing ADSL in their central offices, which was a favorable situation for the growth and accessibility of cable Internet access. At the end-point, a cable modem connects to the cable television coaxial wiring and also attaches to the end-user's desktop via a standard

Ethernet connection. Cable modems can theoretically deliver data at up to 350 times that of a 28.8 modem, (i.e. 10Mbps). Unlike point-to-point ADSL, cable modems are a shared medium, making its architecture a good fit for multicasting. Additionally, end users will not have to build from scratch to take advantage of multicasting. However, because cable modems are shared, they are bound to run into congestion problems on the wire as users fill up local cable loops.

Due to technical limitations, many cable Internet services do not allow users to send data via the cable link. Hybrid systems, in which incoming data comes via the cable connection, but the outgoing data travels over the POTS modem connection are the most common. Therefore, this current system works well if the end-user desires to receive videoconferencing data, but it is not a good set-up for delivering videoconferencing content from the desktop.

D. TERRESTRIAL TRANSMISSION

1. Routing

a. Tunneling

When deciding what routing protocol is most effective over a network, one must look at the network design and topologies. While the NIPRnet (as a whole) is not multicast enabled, the Cisco System routers used throughout the NIPRnet can be easily configured to support multicast. An alternative method is to form "tunnels" between selected multicast-enabled routers in the CONUS segment. Subnet islands can be created, similar to what is used in the MBone, to connect various end-users. These tunnels can be extended to gateways, that have multicast-enabled routers, with access to satellite terminals in order to provide a connection to remote (deployed) users. Some major ISPs (such as UUNET) are using tunneling to implement IP multicast across their networks. A tunnel is essentially a unicast virtual link that may cross several bridges and routers, which encapsulate multicast packets. Tunnel endpoints can be either routers supporting native multicast routing or workstations running the mrouted multicast daemon.

The advantages of tunneling is that it is quick and easy to implement and may be the best solution when both the number of customers using IP multicast and the quantity of IP multicast traffic is limited. Additionally, tunneling is a cost-effective way to gain the benefits of multicast without adding excessive risks or making mass hardware changes. However, there are two major disadvantages. The first disadvantage is setting up and managing multicast servers or gateways. The second is that tunneling inserts the process of encapsulating IP Multicast datagrams into unicast IP datagrams, essentially slowing down the transmission and introducing scaling problems [Hurwicz, 97].

b. PIM-SM

As mentioned in Chapter IV, Sparse-Mode protocols are based upon the assumption that the multicast group members are sparsely distributed throughout the network and bandwidth is not necessarily widely available. It addresses the need for a scalable wide-area, inter-domain, multicast routing mechanism in a large network infrastructure, such as NIPRnet. PIM-SM is available in Cisco System's routers (which comprise most of the routers used on the NIPRnet). PIM-SM solves the routing table problem, found in DVMRP, by using the unicast tables for multicasting [Hurwicz, 97], but there are still some drawbacks. Because unicast routes adjust automatically to equipment or link failures, if there are specific routes that multicast traffic should or must take, there is no guarantee that it will take that route. If all routers are not multicast enabled (which is highly likely) data may be lost.

NASA addressed this problem on its NASA Research and Education Network (NREN) by moving the responsibility for the multicast network to the same groups that were managing the unicast network. Since the hardware usually has a decisive influence on the choice of multicast routing protocol, NASA uses PIM in the Cisco-based portions of the network, and MOSPF on the Proteon router portion, since they are oriented towards MOSPF [Hurwicz, 97].

Since distance learning via videoconferencing in the Navy will require data to be transmitted worldwide, PIM-SM should be seriously considered as a routing protocol in NIPRnet routers used for multicasting.

2. IP over ATM

The NIPRnet has a 10-node ATM backbone in the Continental United States that is connected via SONET OC-12 (622Mbps) pipes. The ATM switches provide switched (SVC) or permanent virtual circuits (PVC), and has promised to handle the QoS issues that IP multicast traditionally did not address. Therefore, instead of the IP datagrams being routed across the long-haul pipes, they will jump to the ATM backbone and exit at a NIPRnet router closest to the destination.

Although it has been proven that ATM has the ability to scale under high traffic loads, one major problem with transporting IP over ATM is that the IP datagrams have to be mapped to ATM protocols before it goes over the ATM backbone, and then converted back.

Not converting IP datagrams to ATM cells eliminate three potential problems. First, IP-to-ATM protocols such as MPOA are complicated, and ATM is still unfamiliar to many network managers. Second, standards for the protocols to map IP to ATM are still not officially set, although they are close to being finalized. Finally, if the challenge is how to push more IP traffic across the data-oriented Internet, you can ignore all of the other things ATM is supposed to do (such as voice) and use ATM's fast hardware for switching IP traffic [Dutcher, 97]. Therefore, finding economical ways of trafficking IP datagrams across ATM network backbones can be a plus for IP based videoconferencing applications.

IP over an ATM network combines layer 3 scalability and flexibility with layer 2 switching and high performance, essentially amounting to VC's across a TCP/IP network, that can stream data at high speeds. Through the development of layer 3 routing in switches, two popular methods have emerged, IP switching and Tag switching.

a. IP Switching

Developed by Ipsilon Networks, IP switching software creates IP ability in ATM switches. The idea is to establish a path across a network. If a network of IP switches set up a "switched" virtual circuit (VC) among themselves across a network, they can improve traditional IP routing. The ATM switch acts as a router for low-duration traffic and as an ATM switch for long-duration flows. It is designed to allow network administrators determine how long a flow should be in order to activate switching instead of IP routing.

NASA is currently conducting studies on the use of IP switching. Its simulation studies have shown eighty-four percent of data packets can be IP routed [Breeden, 97].

b. Tag Switching

Tag Switching software is developed by Cisco Systems. Working with ATM networks, the software tags, or maps, the current network and stores the data in routers. The data packets are tagged and switched as they leave their starting points (in this case Bandwidth Management Centers). The tags can use the Last-in-First-Out (LIFO) method at the switch based upon its priority designation. The tags allow the network to plot a course through the ATM backbone portion. The ATM switches scan the tag and then send it to the next switch. A tag can be an aggregate of tags, allowing an iterative process that increases the scalability of the network. Unlike routers, the switches will need to know the complete path to the edge router destination.

One drawback is that tag switching only works with Cisco equipment. Since the vast majority of routers used in NIPRnet are Cisco routers, there will be no need for major hardware procurement to utilize this method.

c. *ATM Considerations*

If either of these two aforementioned methods is used over NIPRnet's ATM backbone, native routing will essentially be pushed to the periphery of the network, allowing IP switching or Tag switching to handle the backbone segment. Each method advertises the ability to provide almost the same bandwidth as ATM without having to add an extra layer of conversion to already time critical data. Also, IP over ATM may not only provide significant savings in architecture changes, but might also alleviate the need for customers being forced to implement ATM to the desktop, requiring even more spending. One potential problem with these two methods of IP over ATM is that they are still under development and have not proven their ability to scale under heavy network loads. Furthermore, most videoconferencing applications are already devoted mostly to IP.

E. VIDECONFERENCING OVER DISN's SATELLITE SYSTEMS

A deployed unit's means of transporting videoconferencing over DISN (i.e. NIPRnet) will be by using military and commercial SATCOM (C-band and Ku-band), Ultra High Frequency (UHF) and Super High Frequency (SHF) SATCOM, MILSTAR Extremely High Frequency (EHF) Medium Data Rate (MDR), DSCS (military), and/or C band SHF terminals (Challenge Athena) into an entry point or DISN gateway. To provide a gateway to the terrestrial segments of DISN, this integrated satellite transmission system will be further interconnected with the services of the Standardized Tactical Entry Point (STEP).

1. Space Segment

The space segment is composed of Ultra High Frequency (UHF) SATCOM, DSCS II/III multi-channel SHF 75bps – 1.5Mbps(T-1), MILSTAR Extremely High Frequency (EHF) Medium Data Rate (MDR) for medium data rate -- 4.8Kbps - 1.544Mbps, commercial SATCOM (L,C,and Ku bands) – 2.4Kbps – 8.448Mbps, and the Global Broadcast System (GBS), which is currently being readied.

Since satellites are inherently broadcast by nature, an implementation of a typical satellite link requiring satellite terminals and military or commercial satellite resources fits well within the IP multicast basic model.

Deployed units' entry point accesses are currently supported primarily at Navy SATCOM facilities, which serve three of the four NCTAMS. Navy access to non-NCTAMS sites requires circuits to be terrestrially back-hauled to the nearest NCTAMS site. Navy access procedures to terminal segments are described in Naval Telecommunications Publication (NTP)-4, NTP-2, and Communications Information Bulletins (CIBs).

2. Terminal Segment

Connectivity with shore communities can be leveraged using the Standard Tactical Data Entry Points (STEP). STEP is a Joint Staff directed upgrade to the DSCS portion of the Digital Communications Satellite Subsystem (DCSS) program, which is designed to improve and standardize Navy Tactical Satellite Communications (SATCOM). Fourteen DSCS sites will eventually be upgraded worldwide to provide access to DISN. STEP sites provide both ship-to-shore and ship-to-ship communications consisting of operational and administrative traffic. These sites could be either single or dual, whereas a single STEP site supports one satellite coverage area while a dual STEP site supports at least two satellite areas. These gateways can allow at-sea units to quickly connect to the DISN sustaining base services that they need for videoconferencing data. Under the ITSDN Program, NIPRNET routers are installed at the STEP sites, with a 512Kbps-transmission path provided from the STEP site ITSDN router to the NIPRNET backbone. One drawback is that tactical access to ITSDN is provided only on a temporary basis and may require CINC approval. The ITSDN IP router address assignments for tactical units are obtained and provided by the user.

3. Network Cache at the Gateways

Because the ship/shore gateway is a component of the paths of many videoconferencing sessions travelling across the NIPRnet, storing sessions on a cache server offers a potentially significant savings in bandwidth and end user latency by allowing end-users to retrieve data at the gateway, rather than having to reach-back to the original source.

Network caching can be used to deliver to sea video/audio from large disk caches at various gateways, while saving needed for bandwidth across the NIPRnet's territorial backbone network. Therefore, if a student were not able to receive a videoconferencing session real-time, he or she might download a session from a network cache server, where the stored (recorded) session would be. Since personnel will be enrolled in a variety of courses, it must be assumed that all units will not be downloading the same information. Therefore this type of flexibility would require a very large disk cache to store information. In order manage the resources, a certain amount of digital storage space would need to be allocated for each course on the cache server, and also it must be decided how long to leave a videoconferencing session "forward stored" on the server. For example, if a typical video and voice data stream, transmitted to the network cache at 300Kbps (near the upper transmission end of VIXS), were fifty minutes long, the storage space required for the lecture would be approximately 111Mbytes. Table 5-1 shows the estimated storage space.

300Kbps stream * 1 Byte/8 bits = 37.5KBps
37.5KBps * 3600 seconds/hour = 135MB/hour
135MB/hour*.825hours = 111.375MB required per lecture

Table 5-1: Estimated Digital Storage Requirements

If each course stored one week of lectures on a 5GB disk drive, leaving storage space for system operation, over 40 lectures can be stored on just that one drive. With digital storage expected to cost about .02 cents per MB by 1998[3], cost for storage is minimal.

Network cache systems could be used with the Global Broadcast System (GBS) to broadcast videoconferencing data to users. The GBS space-segment is a Ka-Band communications payload carried aboard U.S. Navy UHF Follow-On (UFO) satellites. By providing reliable multicast transport data protocols with GBS, users can download videoconferencing sessions from a gateway, and store data locally for future use. User requests can be made by a slower back channel. In order to manage bandwidth over the space segment, each unit can be given a download-time window, or the network cache can be controlled to deliver content to sea only during non-peak hours.

[3] Survey taken by consulting firm Disk/Trend Inc. of Mountain View, CA.

F. SHIPBOARD

1. ADNS

Because many shipboard networks are not interoperable and require some type of gateway to interface with other systems, SPAWARSYSCOM has developed the Automated Digital Network System (ADNS) within the Joint Maritime Communications Strategy (JMCOMS). ADNS is attempting to convert the Navy stovepipe systems into network-compatible systems without incurring the cost to completely redesign and procure new systems for delivery data to afloat forces [Bergdahl, 96].

Currently the bandwidth of ADNS cannot support real-time videoconferencing, but as it improves bandwidth capacity, ADNS's routing and switching system will provide the interface to end-user video and voice data across available RF media. The routing and switching subsystem should include an IP router and a suite of common multicast routing protocols. The routers should also support QoS protocols, such as RSVP. In order to prevent multicast packets from wasting unnecessary bandwidth on the shipboard LAN, multicast filtering switches might be used. IP multicast-enabled switches automatically set up filters so multicast traffic is only directed to participating end-nodes.

G. CONCLUSION

As shown in the chapter, the network infrastructure and technology is available to deliver IP multicast to sea. If used, delivering videoconferencing over DISN's IP-based networks can alleviate the need for dedicated systems that require people to travel anyway.

Because the architecture and management systems are already in place, using IP-based networks can provide distance learning to a broad audience with minimal spending.

VI. VIDEOCONFERENCING APPLICATIONS

A. INTRODUCTION

This chapter discusses typical videoconferencing software and hardware that can be used to deliver distance learning via videoconferencing from a desktop computer over an IP-based network. This chapter does not endorse any particular software application(s), but is merely providing some examples of common tools currently available. This chapter also provides the recommended standards when employing desktop videoconferencing.

B. VIDEOCONFERENCING APPLICATIONS

Although most of the newer routers and switches are configured to support IP multicast, many of them are, by default, not enabled. Also, many current software applications are unicast and must also be modified to interface with the multicasting capabilities of TCP/IP stacks, which in turn, join and leave multicast groups by using IGMP [Hurwicz, 97]. Because companies realize that there is great potential in videoconferencing, these issues have not inhibited application developers from eagerly creating new products.[4]

Bandwidth and picture quality is still a major impediment, but other barriers like standardization, costs, and installation costs continue to decrease. Microsoft has embedded its collaboration tool, NetMeeting, in its free Internet Explorer 4.0 browser. Netscape

[4] According to Multimedia Research Group, Inc. of Sunnyvale, CA, and Fuji Keizai USA, approximately 4,000 Web sites offered video clips in 1996. That number tripled to 12,000 in 1997 and is expected to triple each year for at least the next three years.

Communicator 4.0, which is also free, packages an analogous tool, Netscape Conference. Microsoft also has released a UNIX version of Internet Explorer 4.0. The MBone also used uses free videoconferencing desktop applications (vic, vat, sdr, wb) that are proven and reliable. Unfortunately, many of the commercial desktop applications, which are PC based, are not fully compatible with the MBone tools, which are mostly UNIX based.

Delivering synchronous/asynchronous video and audio streams to sea not only requires a network architecture, but it also requires software tools that are capable of providing quality content to the student. Even so, quality content delivery does not replace the need for occasional student/instructor collaboration. Today's desktop videoconferencing tools can generally be broken down into two categories. First are standards-based collaboration applications, which provide complete information-sharing solutions that span the spectrum from one-to-one to fully interactive meetings. Secondly, there are streaming applications that broadly distribute one-way, live or stored presentations. Desktop collaborating applications enable users to communicate with a small number of others, such as for desktop videoconferencing. Streaming applications are much more scalable, making it possible to reach a virtually unlimited audience.

Streaming applications will generally have both client and server software, whereas collaboration applications can be client-to-client. To inititalize multipoint sessions, collaborative application users register their contact information with a location server. Four11 and Microsoft's Internet Location Server (ILS) are two examples. These servers are based upon Lightweight Directory Access Protocol (LDAP).

Because audio is the most critical and sensitive aspect of videoconferencing, applications should provide features that allow audio adjustments to compensate for non-guaranteed bandwidth. Applications must support different audio codecs in order to allocate certain amounts of the data stream for different bandwidths. Chat room software can be used as an option when voice and video are bandwidth constrained. The ability to tune audio during transmission, and embedded Forward Error Correction (FEC) or redundancy schemes, used in CU-SeeME and the MBone's rat tool, can help minimize poor audio reception.

Desktop videoconferencing collaboration applications also need a combination of document management capabilities, such as file sharing, white board, and snapshot tools, which allow users to capture whole windows or parts of windows for cutting and pasting to the whiteboard. Standard e-mail applications can be used for administrative purposes, such as setting up time for point-to-point conferencing when additional help is required.

Multicasting videoconferencing applications use basically a straightforward extension to BSD 4.3 Berkley Socket API, which is supported by operating systems such as UNIX, and Windows 95 and NT. As these API's become cross-platform capable, and more readily supported by Winsock 2, they will be ready for widespread use on PC's, running OS's such as Windows.

C. RECOMMENDED STANDARDS

H.323 and H.324, T.120, along with multicast protocols, such as IGMP, RTP, RTCP, and RSVP make up the primary standards for desktop videoconferencing systems. As an extension of H.320, H.323 addresses multipoint videoconferencing over ISDN, POTS, as well as LANs and the Internet. H.324 is the standard for real-time multimedia standards over POTS. When using application from different vendors, ensure that each completely implements the standards it claims. For example Microsoft NetMeeting and Netscape Conference are both "H.323 compliant," but they do not have any common audio codecs, rendering them unable to talk with each other. Even with these misinterpretations, the standards-based support and deploying an application base required for most desktop videoconferencing is no longer an inhibitor. As in the past, network bandwidth and interoperability across different platforms are still the major problems. Dial-up with modems over POTS still continues to be a choke point for delivering and receiving videoconferencing. As H.324 matures, manufacturers will begin to build more H.324 compliant chip sets into hardware. As of now, H.324 is acceptable for point-to-point collaboration, but not for supporting IP multicast.

Although the ITU-T has provided the baseline codec standards for videoconferencing there are several de facto standards that have emerged. Microsoft Video for Windows and Apple QuickTime are common video codecs. QuickTime is compatible with both Windows and Macintosh environments and has been accepted by ITU-T as the basis for MPEG-4. The use of hardware codecs can alleviate some of the CPU usage, but today's multimedia capable processors are more than capable.

One of the first companies to market a product fully based upon IETF standards that relate to real-time video and audio streams, and ITU-T standards for data compression and decompression was Precept Software. Its Flashware Server software and IP/TV viewer client were initially available for Wintel based systems. Because of the implementation nonproprietary standards, this product can receive MBone group sessions, giving it the capability to interoperate with UNIX platforms. Until more companies adopt universal standards, this is one of the few options for cross-platform capability between UNIX and PC users.

Table 6-1 describes the minimum standards needed for videoconferencing systems.

	LAN/WAN, Internet H.323	POTS H.324
Video	H.261 H.263	H.261, H.263
Audio	G.711,G.7.22, G.728, Full-Duplex	G.723 Full-Duplex
Whiteboard	T.120,JPEG,GIF TIFF, Postscript, Still Frame Capture, File Transfer	T.120, JPEG, GIF, TIFF, Postscript, Still Frame Capture, File Transfer
Additional Features	Chat Functions, Application Sharing	Chat Functions, Application Sharing
Multicasting	RTP, RTCP, (RSVP, RTSP when adopted) Multiple Simultaneous Sessions	
Controls	BW Controls(Frame-Rate, Image Size)	BW Controls
Asynchronous Support	Yes	Yes
Additional Support	Firewall Configuration, Trial Copies for testing	Trial Copies for testing
Router Support	MOSPF PIM	

Table 6-1: Videoconferencing Standards over IP Networks

D. HARDWARE

Today's desktop computers provide most of the hardware components needed for videoconferencing. A good camera and video capture card, which can cost as little as $200, is all of the upgrading that is normally required. This is a markedly low price in comparison to roll-about and room-based systems. The release of its newer, faster multimedia based processors is sealing the fate of expensive hardware codecs. This is the recommended desktop system hardware requirement to support desktop videoconferencing:

- Desktop w/ processor that supports multimedia
- Digital camera for face view[5]
- Microphone
- Speakers and/or headphones
- 16 bit sound card, (full-duplex)
- Video Card
- Video capture card[6]
- Web Server[7]
- Minimum 28.8Kbps Modem

[5] Cameras that connect to video capture boards are recommended. Parallel port cameras place requires excessive CPU cycle time (for lesser powerful CPU's- less than Pentium 133, use an Analog Camera).

[6] Video capture cards may include onboard codecs, but as processor power has increased, these more expensive boards are unnecessary. This recommendation is based upon a face-to-face conference. If a server is the capturing device it will use a video capture board.

[7] For Streaming Video Applications over Internet/Intranets

Cameras that connect to video capture boards are recommended. Parallel cameras are unacceptable because of inadequate data throughput, and because they require excessive CPU cycle time.

E. SUMMARY

The ITU-T and IETF standards will likely gain broad acceptance since they are based upon videoconferencing over the commonly existing network architectures. In order for videoconferencing to gain full acceptance, H.320, H.323 and H.324 must work together integrated applications.

Although desktop videoconferencing is becoming more capable, the frame rates and and small picture size of streaming videoconferencing applications are still lacking. If used in conjunction with collaborative software such as whiteboards, shared application and shared control, there is adequate functionality to conduct meaningful learning.

VII. VIDEOCONFERENCING DEMONSTRATION

A. INTRODUCTION

This chapter provides a proof of concept that demonstrates how current videoconferencing software can be used to deliver synchronous or asynchronous material for distance learning over an IP based network via multicast. The demonstration is follow-on work accomplished in *Internetworking: Economical Storage of and Retrieval of Digital Audio and Video for Distance Learning* [Tiddy, 95] and *Internetworking: Worldwide Multicasting of the Hamming Lectures for Distance Learning* [Emswiler, 95].

B. OVERVIEW

Several free software tools were considered, and the one selected was the MBone VCR on Demand (MVoD), developed by Wieland Holfelder at the University of Mannheim, Germany. The MVoD is a free, experimental software solution for the interactive remote recording and playback of multicast videoconferences. The MVoD Service offers a graphical user interface (GUI) environment where the user can interactively record audio/video conferences on a remote server, controlling the recording session with a local client application. Later, that same user or other users can play the session back on demand, via multicast or unicast.

Through the use of this tool, the goals of this experiment was to demonstrate:

- A successful download and installation the MVoD Service software.

- Multicasting a prerecorded taped lecture over the MBone via an SGI workstation while recording the multicast lecture using the MVoD Service from a second workstation that has the MVoD Service software installed.
- Use the MVoD Service to playback and multicast a satisfactorily replicated session over the MBone, which can be received by multiple users.

C. DEMONSTRATION

To begin the testing, the MVoD Service software was downloaded from the site *http://www.informatik.uni-mannheim.de/informatik/pi4/projects/MVoD*. Version 0.9a7 of the software was installed on a Silicon Graphic Indy, running IRIX 6.2 OS, 128 MB of RAM, running a MIPS R1000 processor. The MBone tools sdr, vic and vat were already installed. The MVoD architecture consists of three components:

- The MVoD Server: handles the user and session management
- The MVoD Client: offers the users a GUI to access the MVoD Service
- The RTP DataPump: is responsible for the recording and playback, the synchronization and the administration of the RTP data streams.

A number of internal protocols have been developed to provide communication between the various MvoD software components. They include the:

- VCR Announcement Protocol (VCRAP)- the server announces its services to all clients.
- VCR Service Announcement Protocol (VCRSAP)- the clients have access to the server.
- VCR Stream Control Protocol (VCRSCP)- the client use to access and control a session on the server.

- RTP DataPump Control Protocol (RDCP)- the server uses to control the RTP DataPumps (one per session).

An interface has also been implemented with the Session Announcement Protocol [Perkins, 97], which is used by the MBone tool sdr, in order for the MVoD server to learn about ongoing MBone sessions. Figure 7-1 is the MVoD architecture with its various protocols, which are used in conjunction with MBone tools. Detailed explanations of the various protocols can be found at the web site.

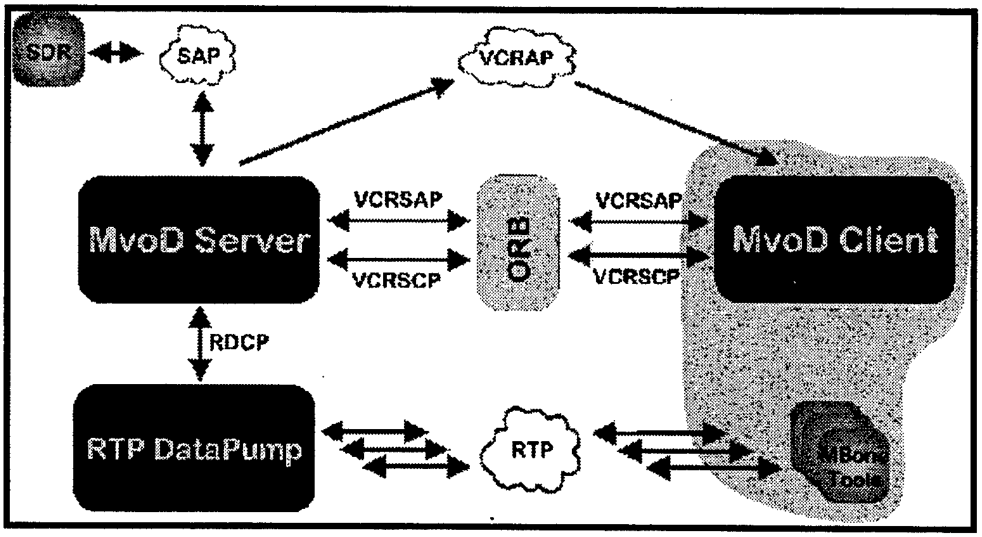

Figure 7-1 MVoD Architecture [Holfelder, 97]

The testing was accomplished using two SGI workstations (Indy and Octane models) on the NPS LAN. The test lectures for the multicast transmission, which had been developed from the thesis "*Internetworking Worldwide Multicast of the Hamming Lectures for Distance Learning*" (Emswiler, 95), were input to an SGI Indy workstation (blacknoise) from the line output of a VCR . The MVoD Service was running on an adjacent workstation (electric). The MVoD Service, and the MBone's sdr, vat and vic were the software used for the experiment. The MBone tools are also free and can be downloaded form many ftp sites that provide MBone tools. The MBone tools used have already been proven effective, therefore the focus of the Chapter will be on the effectiveness of the MVoD recording and playback processes.

D. RECORDING A BROADCAST

The first step was to set up the workstation expected to multicast the lecture over the MBone by providing video and audio line connections from the VCR. The sdr tool on blacknoise was used to create a new MBone session. The video and audio source of the MBone transmission was provided by a VCR that played back the Hamming Lecture Series. Once the VCR was connected and the session created, the lecture was multicast over the MBone using vic and vat (RTPv2). For the multicast, default bandwidth settings for vic H.261 (128Kbps) and vat PCM audio (64Kbps) were used. The time-to-live (ttl) was set to 15, in order to keep the transmission restricted to the campus LAN.

On the workstation "electric," the MVoD Service was running. The MVoD client GUI was used to control the MVoD server and RTP data pump. The May 26th 1995 lecture was recorded by MvoD using a 128kbps (maximum) vic video stream, which lasted for 37 minutes. After the session was recorded, the file size of the recording was noted. Based upon the five files that the MVoD server creates for each recorded session, the total file size for the transmission was approximately 46 MBs. Therefore the recording averaged 1.24MB per minute of data stored. If a standard 50-minute lecture were held, the storage requirements would be approximately 62 MBs. An expected file size is thus approximately 75MB per hour. (This size fits conveniently inside of a 100MB zip disk).

E. PLAYING BACK AN MBone RECORDED SESSION

The next step in evaluating te MVoD Service was to play back (multicast) the recorded session while simultaneously transmitting it over the MBone. Using the MVoD client GUI on electric, a list of sessions previously recorded (which was only one, in our case) by the server was displayed. Once the session was selected, the GUI also provided the option of playing back either audio, video or both mediums. Both audio and video were selected. When the play button was clicked, the session was multicast over the MBone and vic and vat were automatically launched locally in order for the person playing back the session to observe it. The transmission used the same bandwidth settings that were used during the original session and can not be changed.

The rebroadcast (play back) of the session was observed using vic and vat tools on "electric" and "blacknoise." From the observation, there was no discernable difference between the recorded session and the original. There was no packet loss due to the fact that there was no congestion on the LAN containing the multicasting and receiving workstations.

F. EVALUATION OF RESULTS

Currently MvoD only runs on UNIX systems. For a user having little experience with UNIX command lines and environment variables, the MVoD tool is not easy to install. Therefore it is recommended that only System Administrators or experienced UNIX users install the software. During the initial installation, there were problems with killing processes. For instance, some processes could not access sockets even after prior process at the socket had been killed. After becoming more proficient with the tool, and properly shutting it down, this was no longer an issue. Further development of MvoD may make installation simpler, and will likely provide a Windows version as well.

One result important to note that having too many applications running on the workstation slowed down the CPU cycle time, effectively slowing down the compression rate of the transmission. In all of the playbacks (multicasts), the default transmission rates on the audio and video provided a clear reproduction of the original audio/video session. No experiments were conducted using the using the MBone wb tool. Whiteboard recording is not likely to occur soon due to the distributed asynchronous nature of events.

The results of the audio and video testing are satisfactory and demonstrate the successful recording and payback (multicast) of a distance learning lecture using the MVoD Service.

G. SUMMARY

The results of this experiment proved that the technology exists for software tools available to receive, archive, and retransmit distance learning lectures. Once set up properly, the software provides a simple GUI that is easy to use, and not only provides playback on demand but also recording on demand. Being able to record content for future use enables users to build a local library of distance learning content.

The MVoD tool, or a similar tool, can be used to remotely record an instructor's lecture. MvoD could be set up to perform as described in Chapter V. A student can use the MBone tools to connect to the session during the live broadcast, or use the MVoD client GUI to receive a prerecorded session at a more convenient time. If bandwidth over the network segment is restricted, which may often be the case, users can ftp the session from the cache server for local playback.

VIII. CONCLUSIONS AND RECOMMENDATIONS

A. SUMMARY OF FINDINGS

The underlying premise of this thesis is that desktop videoconferencing can be implemented over the currently available DISN IP-based networks instead of dedicated point-to-point, expensive, room based systems that can not provide the scalability necessary to deliver distance learning to a broad, globally dispersed audience. IP multicast is designed to scale well as the number of participants and collaborations expand so that adding one more user doesn't amount to adding a corresponding amount of bandwidth. It doesn't cost any more or require any more bandwidth for 100,000 viewers than it does for one. This fits well with desire to deliver distance learning to numerous participants.

Just within the past two years videoconferencing technology has made enormous strides, and the current capability to implement real time, off-the-shelf or free standards based products has advanced greatly beyond what was available in the past. There are sufficient, well-tested standards that can be used in IP based videoconferencing. Desktop videoconferencing via IP-based networks in the DII is a viable tool that can add numerous economical benefits, such as a decreased spending for travel and eliminating the need to rely on large, room-based videoconferencing systems.

B. RECOMMENDATION FOR FUTURE RESEARCH

This thesis provides a preliminary study on the technological and economic benefits of implementing IP multicast videoconferencing technology from desktops to remote locations. As part of the strategic planning process, additional research is needed to determine the bandwidth parameters, such as latency, delay, on videoconferencing technology within the DISN. Additional research is required in the areas of:

- comparing ATM multicast to IP switching and its viability in wide-scale videoconferencing
- conduct a comparison of current desktop videoconferencing software in its implementation in distance learning.
- determine the feasibility of tunneling over NIPRnet.
- setting up a course and delivering its contents using the MVoD Service is another area of research that can provide an actual demonstration of distance learning from the desktop.
- how network caching and web hosting can be used in videoconferencing
- the implementation of RSVP and RTSP over the NIPRnet.

APPENDIX A. GLOSSARY OF TERMS

API: Application Programming Interface; the generalized term for a defined software interface for software applications.

Asynchronous Transfer Mode (ATM): A connection-oriented technology defined by the ITU and the ATM Forum. At the lowest level, ATM sends all data in fixed cells with 48 octets of data plus five octets of header information, per cell.

Autonomous System: A network controlled by a single administrative authority; a routing domain.

Broadcast: The sending of information from one to all hosts in a LAN network.

Class A: A type of unicast IP address that segments the address space into many network addresses and few host addresses.

Class B: A type of unicast IP address that segments the address space into a medium number of network and host addresses.

Class C: A type of unicast IP address that segments the address space into many host addresses and few network addresses.

Class D: Multicast IP group addresses.

Connectionless: Term used to describe data transfer without the existence of a virtual circuit. UDP is connectionless and provides best effort- unreliable delivery.

CRC: Cyclic Redundancy Check; a mechanism to detect errors in frames.

Ethernet: An industry LAN standard sponsored by DEC, Xerox, and Intel in the early 80s. Became the basis for the official IEEE 802.3 LAN standard.

Frame: The link-layer data entity; data is packaged in frames, for the purpose of transmission over a network. Frames are bounded by flag characters or some other delimiter.

H.320: An ITU-T umbrella of standards for videoconferencing over narrow-band circuit-switched WAN services such as ISDN.

H.323: An extension of H.320, it covers videoconferencing not only over narrow-band WAN services, but also on packet-switched networks, such as LANs and the Internet.

H.324: The ITU-T's standard for real-time multimedia over standard POTS lines using 28.8Kbps V.34 modems or better.

Host: The generalized term for any device that can be a source or sink of information on a network. Generally, a host is a single-networked computer.

IETF: Internet Engineering Task Force; the body associated with the Internet that recommends and approves "standards" for use on the Internet.

IGMP: Internet Group Management Protocol, the protocol with which hosts communicate with the nearest router supporting multicast to notify them about membership in a multicast group.

IP: Internet Protocol; the network layer (layer 3) of TCP/IP. Network layer addresses are used by routers for routing purposes.

ITU-T: The Telecommunications Standardization Sector of the International Telecommunications Union, a body of the United Nations which controls the standards for telephone systems.

MAC: Media Access Control; the protocol used in a LAN or other shared transmission media for gaining access to the media.

MBone: Multicast Backbone is a virtual, experimental network that runs on top of the internet to provide multicasting of live video and audio around the world.

Multicast: The sending of information from one to many, but not all members of a network. See RFC 1112.

Multicast Group: A group set up to send and receive messages from multiple sources and receivers. These groups can be set up based on frame relay or IP in the TCP/IP protocol suite, as well as in other networks.

OSI Model: A seven-layer model of data communications protocols standardized by the International Standards Organization (ISO).

PVC: Permanent Virtual Circuit; a permanent logical connection set up with packet data networks such as frame relay or ATM.

RFC: Request for Comment; the document that the IETF uses to define standards for use and recommend practices in the Internet.

RTP v2: Real-Time Transport Protocol Version 2 is a real-time transport protocol that provides end-to-end delivery of services to support applications transmitting real-time data, for example, interactive video and audio, over unicast and multicast network services. See RFCs 1889 and 1890.

RTCP: Real-Time Control Protocol is a control protocol used in conjunction with RTP. RTCP provides information to applications, identify RTP resources, control RTCP transmission intervals, and conveys minimal session control information. See RFCs 1889 and 1890.

RSVP: Resource Reservation Protocol is an experimental resource reservation set up protocol designed for an integrated services network, that is currently under development. An application might invoke RSVP to request specific end-to-end QoS for a data stream.

SVC: Switched Virtual Circuit; a switched logical connection set up on a temporary basis with packet data networks such as frame relay or ATM.

TCP/IP: The protocol suite used in the Internet. The most important protocol suite used in networking.

TTL (time to live): A counter that is decremented each time a packet passes through a router.

Unicast: The sending of information from one to one in a network; point-to-point data packet communication.

APPENDIX B. INSTRUCTION FOR the BASIC OPERATION OF THE MBone VCR on DEMAND SERVICE (MvoD)

This user's guide has been developed from experience and the information provided by the HTML files that accompany the actual the MBone VCR Service program. It is also a follow-on guide of the MVoD instruction manual from *Internetworking: Economical Storage of and Retrieval of Digital Audio and Video for Distance Learning* (Tiddy, 95). It is designated to provide basic assistance to anyone that desires to use the MBone VCR Service to record or playback a multicast session, and is in no way all encompassing. There are no instructions in this appendix for operating MBone tools. Information about the MBone can be found at *The MBone Information Web* available at *http://www.MBone.com/.*

A. OVERVIEW OF THE MVoD SERVICE

During the recording, the MVoD Service will synchronize the data streams based upon the information provided by the RTPv2 protocol. As with any multicast capable application, the MVoD Service does no need to know the source address of a data stream or the exact content of the data stream, as long as the data stream conforms to the protocols supported by the MVoD Service.

A session recorded by the MVoD Service can be one of as many as 100 multicast sessions that a user desires to record. As many as 20 clients can access the server simultaneously.

To playback a recorded session, the MVoD Service RTP data pump sends the data out to the network, recovering the original timing and synchronization of all the media streams included in this session and using the same network protocols used by the applications from which the data was recorded. The MVoD interface is shown in Figure B-1.

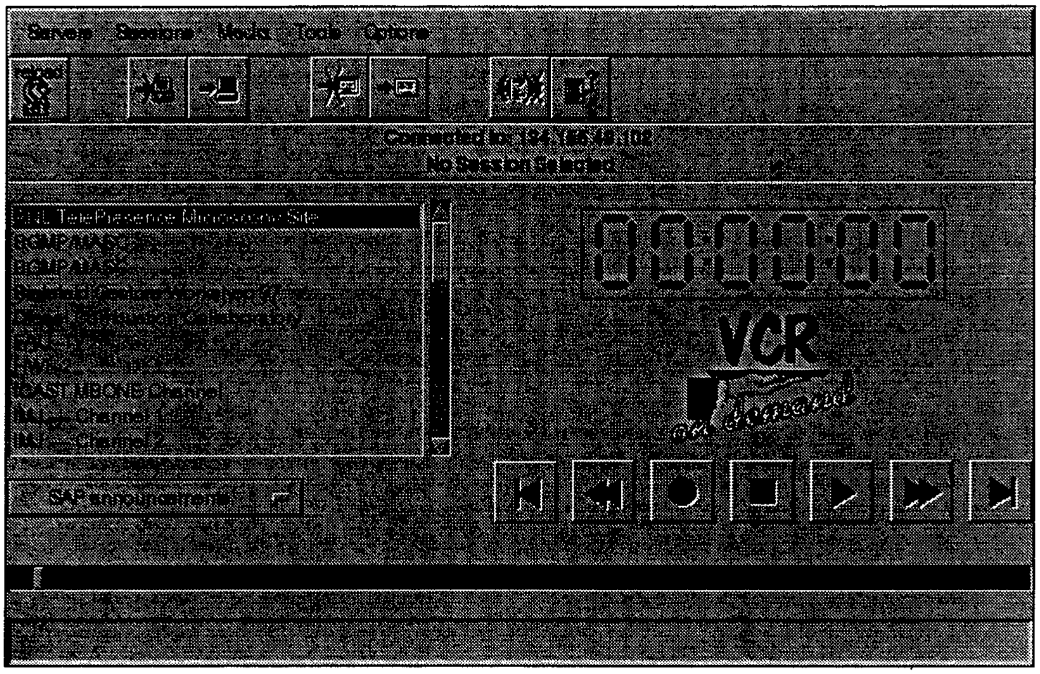

Figure B-1 MBone VCR on Demand Interface [Holfelder, 98]

B. DOWNLOADING THE TOOLS

The MVoD Service software can be downloaded from *http://www.informatik.uni-mannheim.de/informatik/pi4/projects/MVoD.* The site contains a description of the service as well as a source for the various versions of the tools (based the UNIX workstation). The version described in this manual is 0.9a7, downloaded to Silicon Graphics Indy, running IRIX 6.2 OS, 128 MB of RAM, running a MIPS R1000 processor. It also ran on a more powerful SGI Octane workstation. The workstation that the MVoD Service is downloaded to must have JDK 1.1.4 or higher in order to run the client and server components. This resource can be found at *http://www.javasoft.com /nav/download.*

Once the tool has been downloaded, it must be unzipped, using gunzip, and then un-tarred using the tar –xvf command line to install it on the local workstation. The readme file will be included. It will provide detailed instructions for installing and running the MVoD service.

C. USING THE MVoD SERVICE

The following sections describe the basic functions available to the users of the MVoD Service client, and assume that the system administrator has already properly installed the MVoD Server. Additional information can be found in the readme files.

1. Connect to a Server

The first thing that a user needs to do is connect to the desired MVoD server. The GUI will list the servers that the clients will be able to access. The servers announce themselves via the VCR Announcement Protocol (VCRAP). From the list of servers in the left window, highlight the desired server. On the toolbar, select the computer icon, and that will connect the client to the server. Then, the user will need to log on to the MVoD server.

2. Select an MBone session

Below the left window, select from the drop down menu, "SAP announcements." This will show the user a list of the current MBone sessions that the *sdr* is advertising to the MVoD Server. Highlight the desired session. Then go to the Session drop down menu and select "Connect to session," or go up to the toolbar and click the tape icon. This step will connect the user to the desired MBone session. At this point the RTP DataPump will create five files related to that particular session in a directory called **data** (the location of directories is explained in the readme file). In the **data** directory, you will find one session description file (*.rdcp) for every session and two files (*.rec and *.idx) per media in a session in this directory. An index file ends in .idx and a data file ends in .rec. The filenames for these files are automatically generated out of the session filename and the corresponding rdcp-id. For example, given that a session stored in the session file whd-007.rdcp consists of one media with rdcp-id 0 and and one media with rdcp-id 1. Then the automatically generated media files would be: whd-007-0.idx, whd-007-0.rec, whd-007-1.idx and whd-007-1.rec. The content of the .rec file is more or less the raw rtp-data dumped into the file as it was received from the network. The .idx file contains a fixed-length header per data packet that holds a mapped timestamp generated from information of the rtp-timestamps, an offset to the corresponding rtp-data packet in the .rec file and a few other details.

The user will also notice that, once the connection is made, the MBone VCR record function button, located on the lower right of the display will become enabled, and the left window will display the media (video and/or audio) associated with that session. At this point the user is ready to record the session that he or she is connected to.

3. Recording a Session

Once the user is connected to the session, and has verified that the data is being transmitted over the MBone, select the red record button.[8] The MVoD data files for the session are now being recorded and stored in the **data** directory. To stop the recording, use the left mouse button to click the square, black stop function button.

With most of the MBone tools, you can not record data that is sent from the same host where the RTP DataPump daemon is running (e.g. with vat, vic) because these tools do not perform so-called local loopback. However, for playback you can run the RTP DataPump daemon and the MBone tools on the same host since the RTP DataPump does not turn off local loopback.

[8] By default MVoD does not start to record if it does not receive a data signal from any of the media in the session. To start recording when no data is present, select the "Recording without signal" button from the Options drop down menu. Once the button is selected, the digital timer on the right of the display will activate.

In other words, you can not run the source multicast transmission and recording client on the same machine, but no single-machine restrictions exist during palyback.

4. Editing a Session or Media

In order to edit a session that has already been loaded and created, the user must display the available sessions. Click on the drop down menu on the lower left of the display and select "Recorded Sessions". The left window will display the sessions that have been stored (recorded). Select the session that is desired. Connect to the session by clicking the tape icon. Once connected to the session, the media types recorded from the MBone session will be displayed in the left window.

a. Mute a Media

A single click with the left mouse button on the media list will select a media so it can be muted/unmuted with the "mute/unmute" icon or the "mute/unmute" selection under the Media dropdown list. If the media is muted, angle brackets < > surrounding it.

5. Play a Session

To play a session back, simply click the "play" button. In order to listen to and/or watch the data, MBone tools vic and vat need to be launched. They can be launched by selecting the "Tools" dropdown list and then the "start MBone tools". To stop the session, click the stop function button.

6. Fast Forward and Rewind

To fast forward (ff) or rewind (rew) a session, click on the "ff" button or the "rew" button.

7. Random Access with the Session Slider

The slider on the lower part of the display enables random access within the session. Clicking with the middle button somewhere in the slider will forward or rewind the session to this point. Clicking the left mouse button on the slider to the left of the marker will rewind the session about one minute. Clicking on the left mouse button to the right of the marker will forward the session about one minute. At the lower right corner of the display, the total length of the session is displayed.

8. Loop Mode

If the "Loop Mode" entry in the "Options" drop down list is selected during playback, the session will start all over from the beginning when it reaches the end. This feature allows continuous transmissions.

9. Quick-Keys

The following quick-keys are available for the MVoD Client. They will probably continue to change as the product matures.

Key	**Meaning**
q	quit
backspace	go back one level
p	play
shift-p	play at
s	stop
shift-s	stop at
r	record
shift-r	record at
e	edit session
i	info about session
t	start tools
shift-t	start all tools automatically
l	loop-mode on
shift-l	loop-mode off
right	forward one second
shift-right	forward 10 second
ctrl-right	forward 1 minute
left	back one second
shift-left	back 10 second
ctrl-left	back 1 minute
up	goto end
Down	goto start

D. KNOWN BUGS and SHORTFALLS

This demonstration used Version 0.9a7, downloaded to a Silicon Graphic Indy, running IRIX 6.2 OS, 128 MB of RAM, running a MIPS R1000 processor. This version of the MVoD service is more user friendly, because it uses the GUI interface to alleviate many of the manual inputs required in the (Tiddy, 96) instruction guide. This section delineates the known bugs and some of the shortcomings of the MVoD Service.

- The MVoD versions for the SUN workstation could not be untarred. A checksum error was displayed.
- Whiteboard (wb) is not yet supported.

F. SUMMARY

Once installed, the MvoD tool is easy to operate. The GUI is user friendly, and provides context help in the status line, depending on the state of the client and the area the mouse pointer. Although not all encompassing, this instruction manual can aid a new user with simple operation of the MVoD client.

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