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**Data Compression Techniques
In Modern Communication Networks
For Enhanced Bandwidth Utilization**

by

Chigurupati Swarupa Rani

**A dissertation submitted to the Graduate Faculty in Computer
Science in partial fulfillment of the requirements for the degree of
Doctor of Philosophy, The City University of New York**

2001

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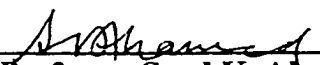
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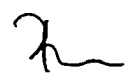
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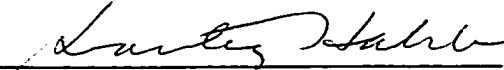
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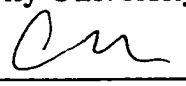
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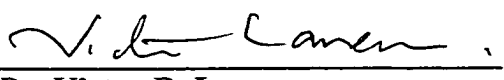

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Abstract**Data Compression Techniques
In Modern Communication Networks
For Enhanced Bandwidth Utilization****by****Chigurupati Swarupa Rani****Mentor: Professor Syed V. Ahamed**

This study represents an exploration of the advantages of compression of data for storage and transmission. The significance of compressing data cannot be overestimated in light of the mushrooming applications and globalization of communication. Compression of data is of particular significance or an imperative for the developing countries (DCs) because of their relatively underdeveloped telecommunications infrastructure. Data compression in the case of DCs may overcome the varied infrastructure bottlenecks as well as represent a case of ‘leap-frogging.’

The study traces the development of communication networks in terms of their diversity of applications that we feel in our daily lives. Several compression algorithms have been developed to make

transmission and storage of data more efficient as the volume of data transmitted grew. The study discusses the distinguishing features of these algorithms in terms of their applications. Simulation is an important tool for modeling a network to determine its performance characteristics. The study outlines the components of the simulation topology, i.e., nodes, links and arcs and simulation methodology pertaining to link utilization, WAN cloud, and message sources as a backdrop to simulation of data transmission.

The core of the study is the analysis and discussion of simulation results. Simulations are done for five different applications, i.e., e-mail, audio, image, video, and HDTV using Huffman Coding, LZW, MPEG-1 and MPEG-2 over five bandwidth percentages. The results show that the link utilization rate, drop rate and access link utilization rate and packets-dropped/packets-delivered ratio for compressed data compares quite favorably with the corresponding results for uncompressed data. The methodology used in the dissertation can be used for evaluating new compression algorithms through the evolving modern communication networks and data storage systems of the future.

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Chapter 1

1. INTRODUCTION

The dissertation topic is centered on the need for data compression as a means of coping with the Information Explosion in a cost-effective way. The topic addresses the need for efficient techniques of storage and transmission. It introduces data compression to the communication networks to enhance data transport services in short intervals and minimize cost. It seeks to experiment with different compression algorithms that would enhance the data handling capabilities of the networks. My work will place special emphasis on data communication needs in developing countries. Many developing countries do have access to networks though plagued by limited circuits and bandwidth problems. Data compression and computer communication can be of great benefit inasmuch as they compensate for limited circuits and bandwidth.

Information revolution has facilitated the movement of massive amounts of data over communication networks. Communication networks are a collection of entities that function in a collaborative role to convey information from its source to its destination. Networks (communication, transportation) have become the essential infrastructure in every society. The ability of communication networks to transfer data at extremely high speeds allows users to gather information in large volumes nearly instantaneously and, with the aid of computers, to almost immediately exercise action at a distance. These two unique applications form the basis for many existing services and an unlimited number of future network-based services. The networks should ultimately be designed to meet the requirements of user applications.

Just as we entered the new millenium, we entered the Information Era, with its many challenges and rewards. Today, society is becoming more and more complex as industry becomes more global and more interconnected. Hence, we find communication becoming more and more essential to help us manage our time more successfully and become more productive. It is indeed a time of more- the needs to communicate have literally exploded. Computers have left the accounting arena in the form of personal computers and entered every aspect of our lives.

In the 1990s, application expanded and interconnected, first within each industry and then across industries. In recent years, there have been dramatic changes and important successes in the field of telecommunications networks. In the last decade, telecommunications have moved from a background role of a utility to applications meant to create new competitive advantages in business, increased productivity in public services and economic development in cities, states, or nations. Advances in telecommunications technology, from POTS (Plain Old Telephone System) to Advanced Intelligent Networks (AIN) are discussed in Chapters 2, 3 and 4.

The merge of telecommunications and computer technology is one of the most exciting developments of 20th century and changed the entire fabric of society. The new types of networks are beginning to grow at a rapid rate. Communication links are a vital part of society's infrastructure. In a computerized society, we look towards electronic communications. In many ways telecommunications will act as a substitute for the increasingly expensive physical transportation. There will be no need to send mail physically; it will be cheaper to send it electronically.

With computers and communications merging into a new broad discipline, competition will center on services that promise a total solution to the end user.

The requirements of the marketplace, advances in technology, and a rapidly changing regulatory picture are all converging to bring about this change- with a great impact on society. The combination of computers and telecommunications technology has already borne fruit. Banks are an obvious example; no major financial institution today could contemplate running its business without online systems that not only serve clientele more efficiently, but also provide an invaluable marketing tool. Airlines industry is another example. They computerized their reservation systems as well as their arrival and departure information.

Communication networks enable users to transfer information in the form of text, voice, audio, video, data, graphics, animations and computer files. Users request the communication service they need through simple procedures using a telephone hand set or set-top TV box or through applications running a host computer such as PC or workstation. The growth of data services, of which the Internet offers an impressive example, is more recent.

Many data processing applications require storage of large volumes of data and the number of such applications is constantly increasing as the use of computers extends to new disciplines. At the same time, the proliferation of computer communication networks is resulting in massive transfer of data over

communication links. Compressing data to be stored or transmitted reduces storage and/or communication costs. When the amount of data to be transmitted is reduced, the effect is that of increasing the capacity of communication channel. Compressing a file to half of its original size is equivalent to doubling the capacity of the storage medium. It may then become feasible to store data at a higher, thus faster, level of the storage, hierarchy and reduce the load on the input/output channels of the computer system. Data compression may be viewed as a branch of information theory in which the objective is to minimize the amount of data to be transmitted [40] whatever the nature of the message – be it words, music, audio or video – it could be reduced to one of two digital bits (0/1). Data compression techniques and applications to modern communication networks are discussed in Chapters 6 and 7.

Computer communications in developing countries (DCs) take on added significance in light of low telephone penetration. Computers can be used to rapidly distribute text materials (including graphics etc), which need not necessarily be viewed on a computer monitor, but can also be printed as hardcopy to be displayed as posters, put on conventional bulletin boards, filed in subject folders etc. This is as important as the direct use of email and news by the relatively few individuals with access to a computer and phone line. In addition, the computer access to libraries and databases that is so useful in developed

countries, could be even more useful in developing countries where direct library services are so poor.

In developing countries, the telephone tariffs would be especially high on long distance calls and even more so on international calls. These tariffs are likely to rise further instead of falling, as in developed countries, since there is a shortage of circuits. Lines are of low quality as well as congested and tariff structures are arbitrary and irrational. If tariffs are based on a minimum of 3-minute or even 1-minute calls, it may be important to persuade telephone companies to make a revenue-neutral change to charge by the second so that the full potential of email for reducing traffic can be realized. Developing countries need to invest more in network capacity to reduce congestion rather than concentrate on an increase in the number of subscriber loops.

Low quality and congested lines are more suitable for computer communications than for voice communications, when used properly. If voice communication is possible over a bad phone line, then data communication is significantly easier, and simply involves a lower data rate than normal, due to error correction, instead of the great difficulty in getting any intelligible message at all through on a bad voice call. High cost calls on the limited capacity long distance lines increases the relative advantage of text communications over voice.

When international circuits were really expensive in developed countries, the bandwidth available was used primarily for telegraph traffic, with voice calls an occasional luxury. Where there is a phone service, but it is expensive to subscribe to and use, computer communications can significantly reduce the long distance phone traffic and provide a hardcopy bureau service similar to, but much more convenient and cheaper than telegraph, telex and fax bureaus.

In developing countries e-mail should have an even greater advantage over fax due to the higher cost of dedicating a phone line to a fax machine (whereas it is comparatively easy for e-mail to use a nondedicated line shared with voice calls). The extremely low telephone penetration and lack of circuits between the capital cities and some major regional centers (let alone small towns and villages), severely limits the section of the population that can directly benefit from any form of telecommunications in developing countries. The importance of compressing data for developing countries is discussed in Chapter 5.

Simulation is a valuable tool of networks because it provides a means for modeling a network to determine its performance characteristics. Network simulation provides a means of testing proposed changes prior to placing them into effect, performing what-if analysis concerning the reliability of key components in a network and the effects of losing a component, planning for

future growth, and planning the initial design of a proposed network. The cost associated with the building and operation of a network make simulation a viable option in making choices in the building, modification, and performance analysis of a network.

Systems designers and network planners are increasingly looking for support tools to help them decide on the best design. They use the results from these tools to make their cases to clients and management. Companies like Compuware (formerly CACI) provide analysis tools for this purpose. COMNET III (a Compuware product), used in this study, is a graphical software package. It is designed to accurately estimate the performance characteristics of computer and communication networks. A network description is created graphically using a window interface. The application was formulated primarily for the modeling of both Wide Area Networks (WANs) and Local Area Networks (LANs). Chapter 8 outlines the details of the choice of simulation tool, topology (first and second), and methodology.

Chapter 9 presents different types of applications and algorithms used for data compression. For each type of application the results of the simulation are presented for both compressed and uncompressed data types. The results are analyzed for each data type and are then compared one with another. Five

different types of applications are used for compression: (a) **E-mail**, (b) **Audio**, (c) **Image**, (d) **Video**, and (e) **HDTV** (High-Definition TV). The compression algorithms applied are (1) **Huffman Coding**, (2) **Lempel, Ziv and Welch (LZW)**, (3) **MPEG-1**, and (4) **MPEG-2**. Arithmetic coding is a possible choice, but the computation is with floating point which creates under/overflow problems. In addition it produces only one codeword, a real number in $[0,1)$ for the entire message to be transmitted. Decoding cannot be done until all bits pertaining to the codeword have been received. Also arithmetic coding is error sensitive compression scheme. A single bit error can corrupt the entire message.

Five different percentages of bandwidth are used for simulation of compressed and uncompressed data types. The percentages of the bandwidth are 50%, 60%, 70%, 80%, and 90%. One simulation run of each percentage for each data type is performed. Thus, ten simulation runs are performed in each application type for both compressed and uncompressed data types. The total simulation runs for all five applications are 50. Chapter 10 presents the summary of simulation results and conclusions.

Microsoft Glitch!

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Chapter 2

2. TRANSMISSION MEDIA

Transmission medium is the physical path between transmitter and receiver in a data transmission system. The transmission of data between computers and terminals can take place along several different media paths. The successful transmission of data depends principally on two factors: the quality of the signal being transmitted and the characteristics of the transmission medium. Various media can be used for the actual transmission, but in many instances, the data communication user has no option in choosing a medium. Each medium has its own niche in terms of bandwidth, delay, cost, and ease of installation and maintenance. Transmission media may be classified as guided or unguided [58]. Guided transmission medium is point-to-point- source to destination. The medium itself is more important in determining the limitations of transmission.

The waves are guided along a physical path, such as twisted pair, coaxial cable, and optical fiber. Unguided media provide a means for transmitting electromagnetic waves, but do not guide them, such as radio, terrestrial microwave, and satellite microwave [58]. The spectrum or frequency band of the signal produced by the transmitting antenna is more important than the medium in determining transmission characteristics. We describe the media most commonly used today, as well as their transmission characteristics in this chapter.

2.1 Twisted Pair

The simplest guided transmission medium consists of two parallel insulated conducting (e.g., copper) wires. The signal is transmitted through one wire while a ground reference is transmitted through the other. This two-wire system is susceptible to cross-talk or interference between neighboring pairs in the same cable. Because the wires are unshielded, there is also tendency to pick up noise from other electromagnetic sources such as broadcast radio.

A twisted pair, depicted in **Figure 2.1**, consists of two insulated copper wires. These wires are twisted together in a regular spiral pattern. The purpose of twisting the wires is to minimize electromagnetic interference between the pairs. The wires in a pair have thickness of from 0.016 to 0.036 inches [6, 27, 58, 60].

Applications

The most common application of the twisted pair is the telephone system. Twisted pairs can be used for either analog or digital transmission. It is the most commonly used medium in the telephone network and is the workhorse for communications within buildings. In the telephone system, individual residential telephone sets are connected to the local telephone exchange, or "end office," by twisted-pair wire. These are referred to as **subscriber loops**. Within an office building, telephone service is provided by means of a private branch exchange (PBX). The bandwidth depends on the thickness of the wire and the distance traveled. Twisted pair can run several kilometers without amplification. For longer distances, repeaters/regenerators are needed. Due to their adequate performance and low cost, twisted pairs are widely used and are likely to remain so for years to come.

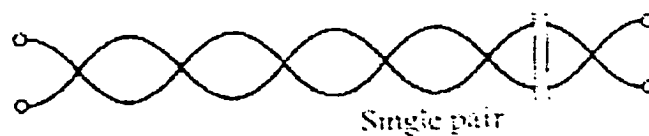


Figure 2. 1: Twisted-Pair Cable

Transmission Characteristics

For analog signals, amplifiers are required about every 5 to 6 kilometers. For digital signals, repeaters are required every 2 to 3 kilometers. Twisted pair is limited in distance, bandwidth, and data rate, compared to other transmission media. The medium is quite susceptible to interference and noise because of its easy coupling with electromagnetic fields.

A twisted pair can pass a relatively wide range of frequencies. For point-to-point analog signaling, a bandwidth of up to about 250 kHz is possible. The attenuation for twisted pair, measured in decibels/mile, can range from about 1 to 4 dB/mile at 1 kHz to 10 to 20 dB/mile at 500 kHz, depending on the gauge (diameter) of the wire. Thus the bandwidth of twisted pair decreases with distance. Practical limits on data rates that can be achieved in a unidirectional link over a 24-gauge (0.016 inch diameter wire, AWG, American wire gauge) twisted pair for various distances are given in **Table 2.1** [41].

Standard	Data Rate	Distance
T1	1.544 Mbps	18,000 feet, 5.5 km
DS2	6.312 Mbps	12,000 feet, 3.7 km
¼ STS-1	12.960 Mbps	4500 feet, 1.4 km
½ STS-1	25.920 Mbps	3000 feet, 0.9 km
STS-1	51.840 Mbps	1000 feet, 300 m

Table 2. 1: Data Rates of 24-gauge (AWG) Twisted Pair

The advantages of twisted pair include their availability and relatively low cost. Their disadvantages include susceptibility to signal distortion or error and relatively low transmission rates they provide for long-distance links.

2.2 Coaxial Cable

Coaxial cable consists of two conductors, but is constructed differently to permit it to operate over a wider range of frequencies. A solid center conductor is located, in coaxial cable, coaxially within a cylindrical outer conductor. The inner conductor is held in place by either regularly spaced insulating rings or a solid dielectric material. The outer conductor is covered with a jacket or shield. A single coaxial cable has a diameter of from 1 to 2.5 cm. **Figure 2.2** shows the signal and ground reference wires as a solid center conductor running coaxially inside a solid outer circular conductor. Because of its shielded, concentric construction, coaxial cable is much less susceptible to interference and cross-talk than is twisted pair. Coaxial cable can be used over longer distances and support more stations on a shared line than twisted pair. Coaxial cables provide for greater bandwidth and faster bit rates than twisted pairs. Typical coaxial cable systems can carry as many as 10,000 voice-grade channels. The current flows more on the outside of the wire at higher frequencies. However, since the wire is

encased in the shell, the current is actually flowing inside the outer shell, and outside the inner wire. This approach allows for a system above the 60 MHz bandwidth (and 10,800 voice-grade circuits). The technology is somewhat limited due to repeater design and signal loss at higher frequencies. The attenuation of coaxial cable increases as the square root of frequency [6].

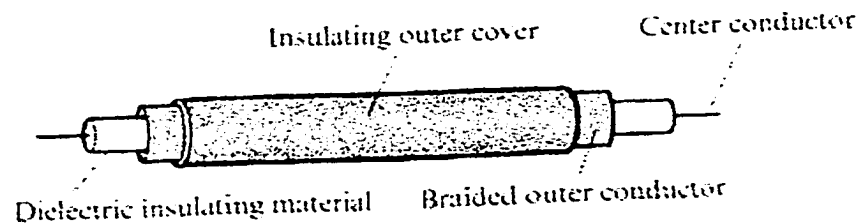


Figure 2. 2: Coaxial Cable

Applications

Coaxial cable is the most versatile transmission medium. The widest use of coaxial cable is for distribution of television signals to individual homes- cable TV. From its modest beginnings as Community Antenna Television (CATV), cable TV will eventually reach almost as many homes and offices as the telephone. Existing coaxial cable systems use the frequency range from 54 MHz to 500 MHz. A National Television Standards Committee (NTSC, used in North America and Japan) analog television signal occupies a 6 MHz band, and a phase alternating lines (PAL, developed in Germany, used in Europe) analog television

signal occupies 8 MHz, so 50 to 70 channels can be accommodated. France and most Middle-Eastern and Eastern-European countries use a third, less popular, color commercial broadcast television standard called SECAM (Sequential Couleur avec Memoire), developed in France in the late 1950s. SECAM uses techniques that are similar to NTSC and PAL to create a color composite video signal.

Coaxial cable has been an important part of the long-distance telephone network. Today, it faces increasing competition from optical fiber, terrestrial microwave, and satellite. Coaxial cable is also commonly used for short-range connections between devices.

Coaxial cables are also used in LANs (Local Area Networks). LANs are available as either broadband or baseband systems. A broadband network is characterized by the use of analog technology. It uses a high frequency modem to introduce carrier signals onto the transmission channel. Because of the analog nature of the network, broadband systems are often frequency division multiplexed (FDM) to provide multiple channels (e.g., data, video, audio) on one path. Broadband LAN is unidirectional- the signal travels in one direction. It is not economically feasible to construct amplifiers that operate in both directions, so the broadband LAN uses two separate channels to provide an inbound and outbound channel.

The baseband network uses digital technology. A line driver introduces voltage shifts onto the channel to represent binary 1s and 0s. The channel then acts as a transport mechanism for the digital voltage pulses. Multiple access to the medium can be provided by a time division multiplexer (TDM) or by other protocols. Baseband systems are bidirectional and use one cable. The technology is limited in distance (0.6 mile/1 km is typical), because the digital pulse attenuates severely over longer distances. Baseband LANs can use either coaxial cable or twisted pair. The coaxial scheme usually is implemented with 50 ohm cable (in contrast to broadband CATV 75 ohm cable) because it works well on multipoint lines and under conditions where low-frequency noise is present [6].

Transmission Characteristics

Coaxial cable is used to transmit both analog and digital signals. Coaxial cable has frequency characteristics that are superior to those of twisted pair, and can hence be used effectively at higher frequencies and data rates. Because of its shielded, concentric construction, coaxial cable is much less susceptible to interference and cross-talk than twisted pair. The principal constraints on performance are attenuation, thermal noise, and intermodulation noise. The latter is present only when several channels (FDM) or frequency bands are in use on the cable. For long-distance transmission of analog signals, amplifiers are needed

every few kilometers, with closer spacing required if higher frequencies are used. The usable spectrum for analog signaling extends to about 500 MHz. For digital signaling, repeaters are needed every kilometer or so, with closer spacing needed for higher data rates.

2.3 Optical Fiber

Optical fiber cable differs from both twisted pair and coaxial cable transmission media. It carries the transmitted information in the form of a fluctuating beam of light in a glass fiber, rather than as an electrical signal on a wire. Light waves have a much wider bandwidth than electrical waves, enabling optical fiber cable to achieve transmission rates of hundreds of megabits per second. Light waves are also immune to electromagnetic interference and cross-talk.

An optical fiber is a thin, flexible medium capable of guiding an optical ray. Optical fiber cable consists of three concentric sections: **core**, **cladding**, and **jacket** shown in **Figure 2.3**. The core is the inner most section and consists of one or more very thin fibers made of glass or plastic. Each fiber is surrounded by its own cladding, a glass or plastic coating that has optical properties different

from those of the core. The interface between the core and cladding acts as a reflector to confine light that would otherwise escape the core. The outermost layer is the jacket made of plastic and other material layered to protect against moisture, abrasion, crushing, and other environmental dangers. The information itself is transmitted through the core in the form a fluctuating beam of light.

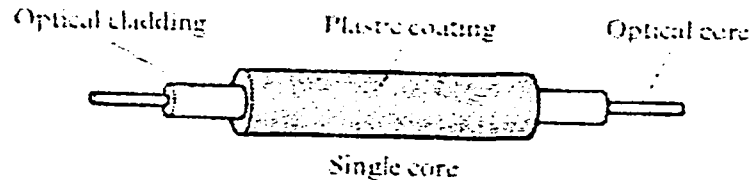


Figure 2. 3: Optical Fiber

Applications

Long-haul fiber transmission is becoming very common in the telephone network. Long-haul routes average about 1500 km in length and offer high capacity (20,000 to 60,000 voice channels). Metropolitan trunking circuits have an average length of 12 km and may have as many as 100,000 voice channels in a trunk group. Most facilities are installed in underground conduits and are repeaterless, joining telephone exchanges in a metropolitan or city area. Rural exchange trunks have circuit lengths ranging from 40 to 160 km and link towns

and villages. In the United States, they often connect the exchanges of different telephone companies. Subscriber loop circuits are fibers that run directly from the central exchange to a subscriber. These facilities are beginning to displace twisted pair and coaxial cable links as the telephone networks evolve into full-service networks capable of handling not only voice and data, but also image and video. An important application of optical fiber is for LANs. Standards have been developed and products introduced for optical fiber networks that have a total capacity of 100 Mbps to 1 Gbps and can support hundreds or even thousands of stations in a large office building or a complex of buildings [58].

The advantages of optical fiber over twisted pair and coaxial cable become more compelling as the demand for all types of information (voice, data, image, video) increases.

Transmission Characteristics

Optical fiber transmits a signal-encoded beam of light by means of total internal reflection. There are three different types of transmissions:

- **Step-index Multimode:** Light from a source enters the core. Rays at shallow angles are reflected and propagated along the fiber; other rays are absorbed by the surrounding material. This type of fiber is best suited for transmission over very short distances.

- **Single-mode:** When fiber core radius is reduced, fewer angles will reflect. By reducing the radius of the core to the order of a wavelength, only a single angle or mode can pass: the axial ray. This single-mode propagation provides superior performance. Because there is a single transmission path with single-mode transmission, the distortion found in multimode cannot occur. Single-mode is typically used for long-distance applications, including telephone and cable television.
- **Graded-index Multimode:** The higher refractive index at the center makes the light rays moving down the axis advance more slowly than those near the cladding. Rather than zig-zagging off the cladding, light in the core curves helically because of the graded index, reducing its travel distance. The shortened path and higher speed allows light at the periphery to arrive at a receiver at about the same time as the straight rays in the core axis. Graded-index fibers are used in LANs.

2.4 Radio

Radio waves are easy to generate, can travel long distances, and penetrate buildings easily. So radio waves are widely used for communication, both indoors and outdoors. Radio waves are also omnidirectional- travel in all

directions from the source- so that the transmitter and receiver do not have to be carefully aligned physically.

The properties of radio waves are frequency dependent. At low frequencies, radio waves pass through obstacles well, but the power falls off sharply with distance from the source, roughly as $1/r^3$ in air [60]. At high frequencies, radio waves tend to travel in straight lines and bounce off obstacles. At all frequencies, radio waves are subject to interference from motors and other electrical equipment.

The radio wave and transmitting antenna are designed around the following operations (see **Figure 2.4**) [6]:

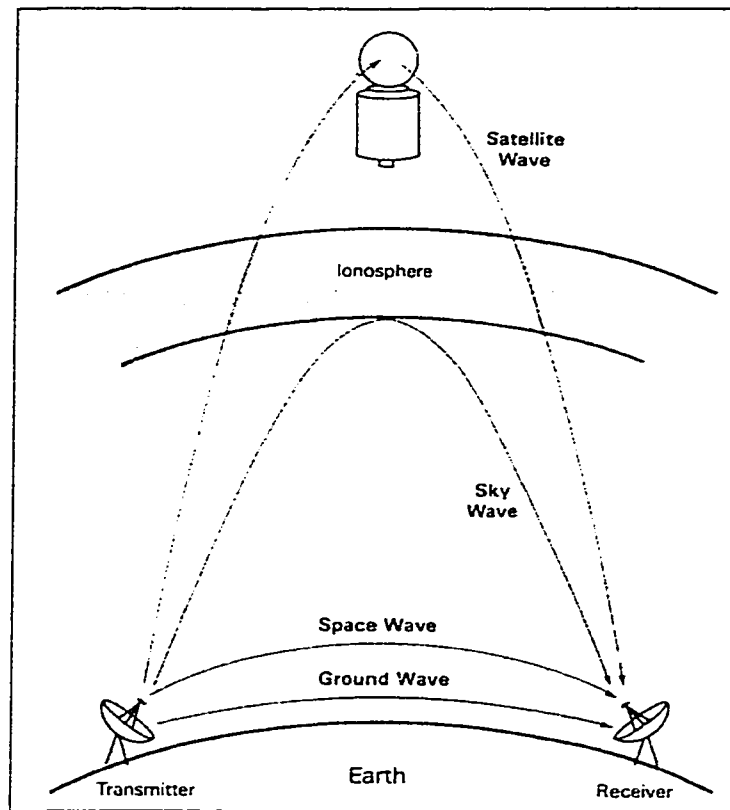


Figure 2. 4: Radio Waves

- **Ground or Surface Wave:** Ground wave is characterized by its propagation along the lower edge of the earth's surface. This signal is used generally in the low frequency (LF) bands (30kHz-300kHz). It is also used for broadcasting in the medium frequency (MF) band (300kHz-3MHz). Buildings or mountains do not affect the signal very much.
- **Space Wave:** Space wave travels in a straight line between the transmitting and receiving antennas. It is considered to be a line-of-sight transmission (direct wave). In addition, space wave has a component which travels by

reflecting from the earth. This type of signal is used principally in TV broadcasting operation in the VHF/UHF/SHF bands.

- **Sky Wave:** Sky wave is transmitted upward into the ionosphere. Based on the ionospheric conditions, the signal or a portion of it is returned to earth and received at the ground station. This type of transmission is commonly used for high frequency (HF) radio communications devices (3 MHz-30 MHz) and long-distance telephone operations.
- **Satellite Wave:** Satellite wave method uses a radio frequency transmitted to a satellite station which is amplified and then transmitted back down to earth at a different frequency.
- **Scatter Wave:** Scatter wave travels downward in a scatter approach toward the receiver. Scatter wave is used on long-distance high frequency links. It also provides multi-channel telephone links.

2.5 Terrestrial Microwave

Terrestrial microwave links are widely used to provide communication links when it is impractical or too expensive to install physical transmission media, for example across a river or a swamp or desert. The most common type of microwave antenna is the parabolic “dish”. A typical size is about 3 meters in

diameter. The antenna is fixed rigidly and focuses a narrow beam to achieve line-of-sight transmission to the receiving antenna. Microwave antennas are usually located at substantial heights above ground level to extend the range between antennas and to be able to transmit over intervening obstacles.

Applications

The primary use for terrestrial microwave systems is in long haul telecommunications service, as an alternative to coaxial cable or optical fiber. The microwave facility requires far fewer amplifiers or repeaters than coaxial cable over the same distance. But, it requires line-of-sight transmission. Microwave is commonly used for both voice and television transmission. It is also used for short point-to-point links between buildings. This can be used for closed-circuit TV or as a data link between LANs.

Transmission Characteristics

Common frequencies used for transmission are in the range 2 to 40 GHz. The higher the frequency used, the higher the potential bandwidth and therefore the higher the potential data rate. **Table 2.2** shows bandwidth and data rate for some typical systems [58].

Band (GHz)	Bandwidth (MHz)	Date Rate (Mbps)
2	7	12
6	30	90
11	40	135
18	220	274

Table 2. 2: Typical Digital Microwave Performance

Attenuation is the main source of loss. The loss can be expressed as

$$L = 10 \log (4\pi d/\lambda)^2 \text{ dB}$$

where d is the distance and λ is the wavelength. The loss varies as the square of the distance. In case of twisted pair and coaxial cable, loss varies logarithmically with distance. Attenuation increases with rainfall- the effects of rainfall become especially noticeable above 10 GHz. Interference is another source of impairment. With the growing popularity of microwave, transmission areas overlap and interference is always a danger. Thus the assignment of frequency bands is strictly regulated.

2.4 Satellite Microwave

Data can also be transmitted using electromagnetic (radio) waves through free space as in satellite systems. A collimated microwave beam, onto which the

data is modulated, is transmitted to the satellite from the ground this beam is received and retransmitted (relayed) to the predetermined destination(s) using an on-board circuit, transponder.

For a communication satellite to function effectively, it is generally required that it remain stationary with respect to its position over the earth. Traditional geostationary earth orbit satellites (GEOS) are placed in an orbit approximately 35,786 km above the earth's surface at the equator. The rotation of the satellite is then synchronized to that of the earth, so the satellite appears stationary with respect to the earth [41, 58].

Two satellites can interfere with each other, if both use the same frequency band and are close enough together. Current standards require a 4° spacing in the 4/6 GHz band and a 3° spacing at 12/14GHz. Hence, the number of possible satellites is quite limited.

Applications

The satellites are used for television distribution, long-distance telephone transmission, and private business networks. Satellites are well suited for television distribution and are being used extensively in the United States and throughout the world for this purpose. Satellite transmission is also used for

point-to-point trunks between telephone exchange offices in public telephone networks. It is the optimum medium for high-usage international trunks and is competitive with terrestrial systems for many long-distance intranational links.

There are a number of business data applications for satellite. The satellite provider can divide the total capacity into a number of channels and lease these channels to individual business users. A recent development is the very small aperture terminal (VSAT) system, which provides a low-cost alternative. A number of subscriber stations are equipped with low-cost VSAT antennas. These stations share a satellite transmission capacity for transmission to a hub station. The hub station can exchange messages with each of the subscribers and can relay messages between subscribers.

Transmission Characteristics

The optimum frequency range for satellite transmission is in the range 1 to 10 GHz. Below 1 GHz, there is significant noise from natural sources, including galactic, solar, and atmospheric noise, and human-made interference from various electronic devices. Above 10 GHz, the signal is severely attenuated by atmospheric absorption and precipitation. Most satellites providing point-to-point service today use a frequency bandwidth in the range 5.925 to 6.425 GHz for transmission from earth to satellite (uplink) and a bandwidth in the range 4.7 to 4.2

GHz for transmission from satellite to earth (downlink). The combination is referred to as the 4/6 GHz band.

Because of the long distances involved, there is a propagation delay of about a quarter second from transmission from one earth station to reception by another earth station. This delay is noticeable in ordinary telephone conversations. It also introduces problems in the area of error control and flow control. Satellite microwave is inherently a broadcast facility. Many stations can transmit to the satellite, and a transmission from a satellite can be received by many stations.

Chapter 3

3. SWITCHING TECHNOLOGY

The basic service required of communication networks is the transfer of data (voice, image, video) from source to destination. The communication network is designed to transfer variable-length blocks of information up to some specified maximum size through switching nodes. Data entering the network from a source are routed to the destination by being switched from node to node. The nodes are connected to one another by transmission links. Different switching techniques are used for data transmission in communication networks: **Circuit Switching**; **Message Switching**; and **Packet Switching**. The first published work on packet-switching techniques was that of Leonard Kleinrock. Using queuing theory Kleinrock's work elegantly demonstrated the effectiveness of the packet-switching approach for bursty traffic sources [Kleinrock, Baran, and Davies & Scantlebury, 38]. Two approaches that evolved from packet switching, namely

ATM (Asynchronous Transfer Mode) and **Frame Relay**. These technologies differ in the way the nodes switch information from one link to another on the way from source to destination. We will give a brief introduction to each one of them below.

3.1 Circuit Switching

Communication via circuit switching implies that there is a dedicated communication path between two stations. A physical connection is established between two nodes, see **Figure 3.1** [1]. Channel capacity is dedicated for the duration of a connection, even if no data are being transferred. Switching and transmission resources within the network are reserved for the exclusive use of the circuit for the duration of the connection. There are five stages for the data transfer to take place through a switched channel: (a) an idle channel is identified and tagged; (b) a connection is established; (c) the data is transferred; (d) the channel is released; and (e) once again, the channel resumes idle status.

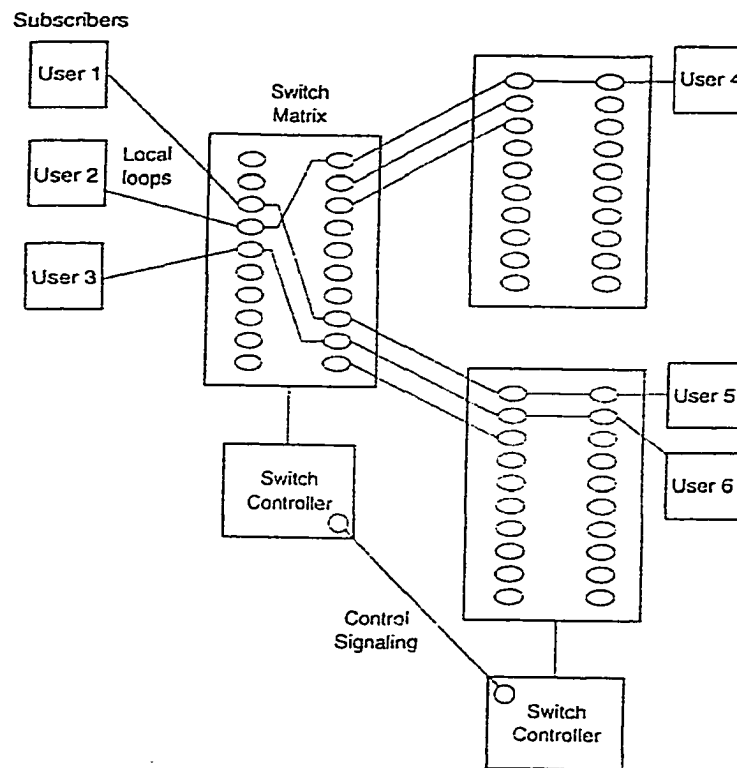


Figure 3. 1: Circuit Switching

This sequence of stages closely parallels the well-established network steps in a typical voice call that follows the five network functions: call setup, alert, connect, disconnect, and release. Circuit-switched networks are analogous to the voice telephone system, and such connections are termed virtual circuits. Virtual circuit establishes a single route for data between the nodes that does not vary for the life of the connection. Hence, they are connection-oriented.

Circuit switching was developed to handle voice traffic. But it can also be used for data traffic. Circuit switching can be rather inefficient. The best-known example of a circuit-switching network is the public telephone network. The telephone networks are connection-oriented because they require the setting up of a connection before the actual transfer of information can take place. This is an important property of circuit switching. IBM's SNA (Systems Network Architecture) is another example of a circuit-switched network [53].

Circuit-switching technology has been driven by those applications that handle voice traffic. One of the requirements for voice traffic is that there must be virtually no transmission delay and no variation delay. A constant signal transmission rate must be maintained, because transmission and reception occur at the same signal rate. These requirements are necessary to allow normal human conversation. Circuit switching is preferred for its simplicity, constant-delay capacity, and constant-link capacity. One of its key strengths is that it is transparent. Once a circuit is established, it appears as a direct connection to the two attached stations no special networking logic is needed at the station.

In the context of ISDN (Integrated Services Digital Network), the information necessary to accomplish these steps is incorporated in the ITU-T (International Telecommunications Union-Telecommunications Standardization

Sector; formerly known as CCITT (International Telegraph and Telephone Consultative Committee)) Q.931 protocol at the network layer used over the D (Delta) channel to set up the B (Bearer) channel.

3.2 Message Switching

When message switching is used, no physical path is established in advance between sender and receiver. A message of information is first received in its entirety with sender and receiver addresses and identification, stored in the first switching office (i.e., router) and then forwarded later, one hop at a time. The message path and its orientation, circuit (connection-oriented) or packet (connectionless-oriented), can be chosen by the agent receiving the message. An operator makes a routing decision based on destination address information. The operator then stores the message until the communication line becomes available to forward the message to the next appropriate station. This process is repeated until the message arrives at the destination station. A network using this technique is called store-and-forward network shown in **Figure 3.2** [1].

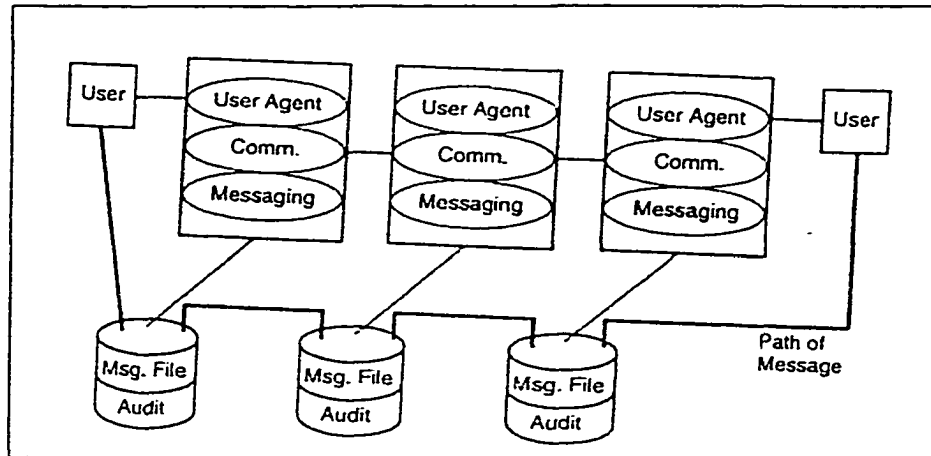


Figure 3. 2: Message Switching

The best-known example of a message-switching network is the first electromechanical telecommunication systems used for telegrams. With message switching, there is no limit on message (block of data) size. So the routers must have disks to buffer long blocks. A single block may tie up a router-router line for minutes, rendering message switching useless for interactive traffic. To get around these problems, packet switching was invented.

3.3 Packet Switching

When packet switching is used, the exact physical path is not established in advance between sender and receiver. In this connectionless-oriented data

transfer, all data to be transmitted is first assembled into one or more packets by the source data terminal equipment (DTE), see **Figure 3.3** [1]. There is an upper limit on packet size so that the packets can be buffered in router main memory instead of on disk. A typical length of a packet is 128 bytes. If a source has a longer message to send, the message is broken up into a series of packets. These packets include both the source and destination DTE network addresses in header. The packets are then passed bit serially by the source DTE to its local packet switching exchange (PSE). Each PSE contains a routing directory specifying the outgoing links to be used for each network address.

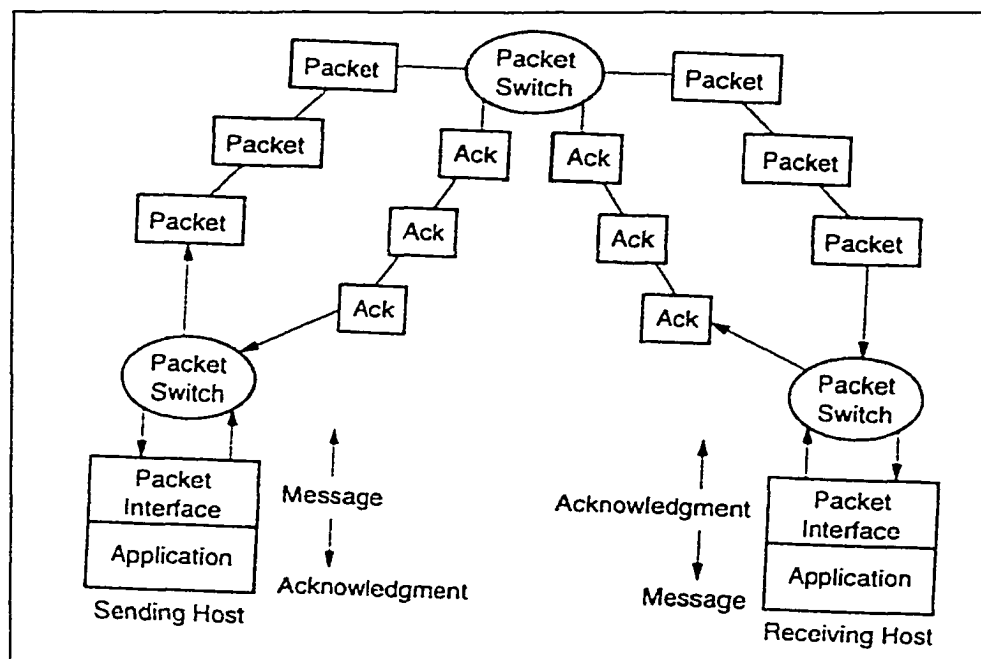


Figure 3. 3: Packet Switching

The PSE forwards the packet on the appropriate link at the maximum available bit rate. At the destination PSE, determined by the destination address within the packet, the packet is finally passed the destination DTE. The destination DTE sends an acknowledgement back to the source DTE when a packet is received error free.

Circuit switching and packet switching differ in many respects. The main difference is that circuit switching statically reserves the required bandwidth in advance- any unused bandwidth on an allocated circuit is just wasted. Packet switching acquires and releases the required bandwidth as needed and it may be utilized by other packets from unrelated sources going to unrelated destinations, because circuits are never dedicated. Another major difference is that circuit switching is completely transparent, the carrier does not determine the basic parameters. Packet switching is not transparent, the carrier determines the basic parameters. The differences are summarized in **Table 3.1** [60].

Item	Circuit-Switched	Packet-Switched
Dedicated 'copper' path	yes	no
Bandwidth available	fixed	dynamic
Potentially wasted bandwidth	yes	no
Store-and-forward transmission	no	yes
Each packet follows the same route	yes	no
Call setup	required	not needed
When can congestion occur	at setup time	on every packet
Charging	per minute	per packet

Table 3. 1: A Comparison of Circuit-Switched and Packet-Switched Networks

ITU-T (formerly CCITT) has specified the widely accepted X.25, X.75, X.28, and X.29 protocol at the network layer. Ethernet standards dominate the packet-switching usage for LANs, and X.25 for WANs. The best-known examples of a packet-switching networks are E-mail and Internet. The availability of the packet-switched networks can get extremely widespread with FDDI (Fiber Distributed Data Interface). SONET and ATM technologies provide additional applications. Both ATM and TCP/IP networks are based on packet switching. However, ATM is connection-oriented, whereas IP is connectionless. The packet networks are controlled for their flow and for any possible errors. Being highly multiplexed and time-shared, they are highly efficient and very inexpensive.

Two types of services are normally supported with a packet-switching network: **Datagram** and **Virtual Call (circuit)**.

3.3.1 Datagram Approach

The datagram service is analogous to sending a message by a letter, since each packet entering the network is treated as a self-contained entity with no relationship to other packets. Each packet is routed independently, with no reference to packets that have gone before. Each packet contains the address of its destination. So the packets do not all follow the same route. As a result, it is possible that the packets will be delivered to their destination in a different sequence from the one in which they were sent. It is also possible for a packet to be lost in the network. The destination node detects the loss of a packet and figure out how to recover it. In this technique, each packet, treated independently, is referred to as a datagram. This approach was used in the ARPANET (Advanced Research Project Agency Network) and later the Internet.

3.3.2 Virtual Circuit Approach

The idea behind virtual circuits is to avoid having to choose a new route for every packet sent. In virtual circuit approach, a preplanned route is established before any packets are sent. So the main characteristic of the virtual circuit technique is that a route between source and destination is setup prior to data transfer. When a connection is established, a route from source to destination is chosen as part of the connection setup and remembered. That route is used for all traffic flowing over the connection. The virtual circuit is terminated, when the connection is released. This connection is not dedicated as in circuit switching. A packet is still buffered at each node, and queued for output over a line, while other packets on other virtual circuits may share the use of the line. The node does not make a routing decision for each packet. The destination machine makes the decision only once for all packets using that virtual circuit.

The virtual circuit service of X.25 provides for two types of virtual circuit: **virtual call** and **permanent virtual circuit (PVC)**. A virtual call is a dynamically established virtual circuit using a call setup and call clearing procedure. A permanent virtual circuit is a fixed, network-assigned virtual circuit. Data transfer occurs as with virtual calls, but no call setup or clearing is required.

3.4 Frame Relay

A frame relay is essentially an electronic switch. Frame relay is defined by the ITU-T recommendation I.122 as an adaptation of the ISDN interface for the purpose of providing frame relay services. It is a connection-oriented data transport service for public switched networks (PSN). Frame relay transmits data frames with variable length just as X.25 does. Both X.25 and frame relay specify the lowest three OSI layers for virtual circuit networks.

Frame relay technology is used to provide semi-permanent connections for LAN interconnections. The protocol is based on ISDN signaling protocol LAPD (Link Access Procedure on D Channel) and is defined in the ITU-T (formerly CCITT) recommendations I.122/Q.922. Frame relay is a technology for data transmission and does not support voice or video. **Figure 3.4** illustrates its operation [53]. Let VC1 (A->C), VC2 (A->D), and VC3 (A->E) be three virtual circuits. All three circuits flow through Frame Relay B. Suppose data for VC1, VC2, and VC3 flow into multiplexer A. The MUX places them into the frame, storing an address and length with the data. The frame is transmitted from A to B. B must demultiplex it and then create frames for C, D, and E.

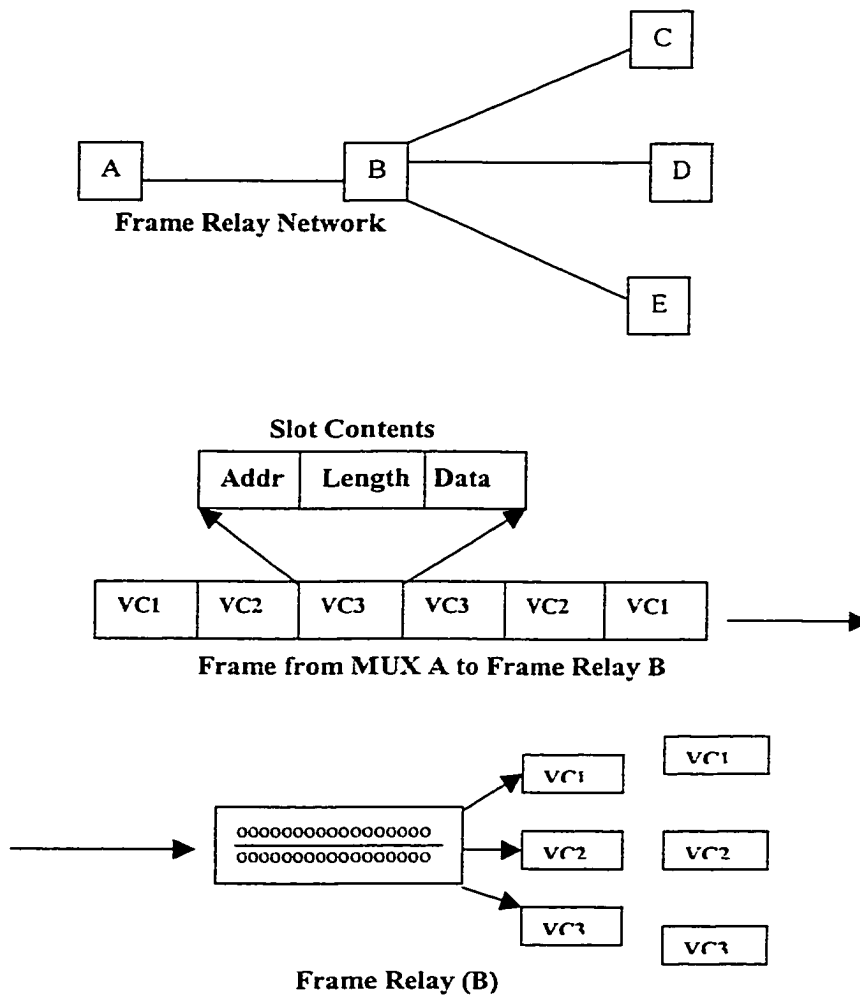


Figure 3. 4: Frame Relay

The frame format is shown in **Figure 3.5** [63]. The 2-byte header contains address information (DLCI, EA) for routing, congestion-control information (F/BECN, DE) for notification and enforcement, and the C/R bit whose usage is application-specific. The frame check sequence (FCS) is a 16-bit CRC for error

detection: erroneous frames are discarded and are not retransmitted by the network. The standard specifies the use of permanent virtual circuits (PVCs) for connections.

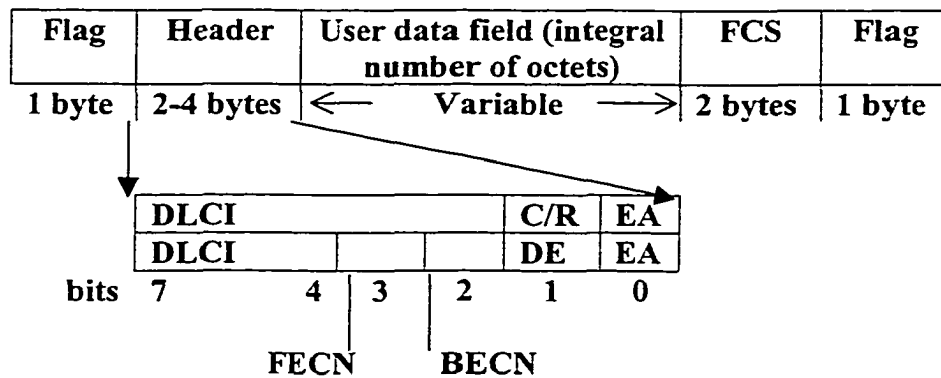


Figure 3. 5: Frame Format of Frame Relay

DLCI = Data link connection identifier (12-bit);

F/BECN = Forward/backward explicit congestion notification;

C/R = Command/response; DE = Discard eligibility;

EA = Extended address; EA=0 indicates that the next byte is also an address byte; EA = 1 indicates the last address byte.

Characteristics

- Frame relay is defined by ITU-T, T1, or E1 bandwidth.
- Frame relay is intended only for data communications, not voice or video, for the same reason that Stat MUXes are limited in this way.
- Only connection-oriented service is provided.
- Transmission errors are detected but not corrected (the frame is discarded). It is up to the user to discover that a frame is missing and take the necessary

action to recover. Frame relay facilities are assumed to have low bit error rates.

- Frame relay is faster than X.25 because only the data link layer is needed and no error correction is performed within the network.
- Frame relay is currently offered by most of the major carriers such as AT&T, MCI, and Sprint. It is not suited for voice and video because of its variable length frames.

Frame relay competes with X.25 permanent virtual circuits, except that it operates at higher speeds, up to 2 Mbps and provides fewer features. Frame relay leaves data checking and acknowledgement procedures to network users. The header contains a bit to indicate that problems exist; the bit is set by the users.

Frame relay is used, in the United States, in the private domain, where it serves to optimize dedicated high-speed links (1.544Mbps), and also in the public domain where numerous operators offer a permanent virtual link service from 56 kbps to 1.544 Mbps [7]. The primary application of frame relay is to interface high-speed LANs to WANs, where very large LAN packets must be quickly injected into WANs. Other applications include image transfer and remote mainframe access. Frame relay service, first offered commercially in 1993, has become one of today's most popular public telecommunications services.

The worldwide revenue of frame relay switches is estimated to \$927M in 1996 with a projected increase to \$1.5B by the year 2000 [68]. Frame relay takes advantage of high data rates and low error rates. It allows multiple calls to different destinations to be in progress concurrently. Frame relay is by far the dominant service, owing to its minimal overheads.

3.5 Cell Relay

Cell relay is operated using two technologies which have common characteristics: the **asynchronous transfer mode (ATM)**, chosen by CCITT (ITU-T) for the information transfer mode of broadband ISDN (BISDN); and the **DQDB** (Distributed Queue Dual Bus) technology. DQDB (Australian) is a standard defined by IEEE 802.6 committee for the construction of metropolitan networks [7]. Cell relay is a packet switching technology that operates at the physical and data link layers of the OSI model. It uses a small, fixed-length cell. These small cells can carry any type of information such as data, voice, and video at tremendously high speeds.

ATM breaks all traffic into 53-byte cells. The proper size of a cell was the subject of much debate in the standards committees. The telephone companies

wanted a small cell (to reduce delay for voice packets), while the data communications people wanted a big cell (to minimize the amount of segmentation and reassembly that had to be carried out). The committees, after much discussion, narrowed the cell size debate into two choices: 32-byte or 64-byte cells. As a compromise, a 48-byte payload was chosen. Each cell is composed of 53 (424 bits) bytes and contains two principal fields: header, 5 bytes (40 bits); and payload (user information), 48 bytes (384 bits) as shown in **Figure 3.6**.

Header (5 bytes)						Payload
GFC	VPI	VCI	PLT	CLP	HEC	48 bytes
4 bits	24 bits		2 bits	2 bits	8 bits	48 bytes

Figure 3. 6: Cell Format

GFC (Generic Flow Control): prescribes a standard method of terminal-network flow control that is fair, robust, and realizable as an option by the ATM device manufacturers.

VPI (Virtual Path Identification) and VCI (Virtual Channel Identification): provide numbering identification of the Virtual Paths and Virtual Circuits.

PLT (Payload Type Identification): identifies the type of traffic carried in the ATM cell.

CLP (Cell Loss Parity): assigns high/low priority to the cell (by the network or customer equipment) so that in case of congestion and other problems in the network, cell discarding may be exercised.

HEC (Header Error Check): error correction/detection field that applies to the four bytes of cell header only. Error correction/detection on the remaining user data may be provided by the ATM Adaptation Layer (ALL) if necessary.

Two interfaces are defined for ATM: (1) **User Network Interface (UNI):** multiplexers or customer equipment interface with an ATM network; and (2) **Network Node Interface (NNI):** ATM network switches interface with one another (see **Figures 3.7** and **3.8**).

Bits							
8	7	6	5	4	3	2	1
GFC				VPI			
VPI				VPC			
VCI							
VCI				PT			
HEC							

Figure 3. 7: ATM Cell Header Structure at UNI

Bits							
8	7	6	5	4	3	2	1
VPI							
VCI				VCI			
VCI							
VCI		PT		CLP			
HEC							

Figure 3. 8: ATM Cell Header Structure at NNI

Cell relay is used in the interface to SMDS (Switched Multi-Megabit Data Service, created as a MAN service by Bellcore), in BISDN. The first ATM switch (622Mbps) was installed in late 1991 [53].

Chapter 4

4. NETWORKS

A communication network is a set of equipment and facilities that provides a service: the transfer of information from point A to point B located at various geographical points. Examples of such networks include telephone networks, computer networks, television broadcast networks, and the Internet. Communication networks provide a service- flow of information- much like other ubiquitous utilities. Networks (communication, transportation) have become essential infrastructure in every society. The ability of communication networks to transfer data at extremely high speeds allows user to gather information in large volumes nearly instantaneously and, with the aid of computers, to almost immediately exercise action at a distance. These two unique capabilities form the basis for many existing services and an unlimited number of future network-based

services. The networks should ultimately be designed to meet the requirements of the user applications.

4.1 History of Communication Networks

Communication networks enable users to transfer information in the form of text, voice, audio, video, data, graphics, animations, and computer files. The information can be analog or digital. Users request the communication service they need through simple procedures using a telephone hand-set or set-top TV box or through applications running on a host computer such as a PC or workstation. In developed countries, the telephone network has grown so that for many years it has been accessible to virtually every business and home. In developing countries where penetration of the telephone network is significantly below 100%, there is a large unfilled demand, leading to a waiting time for telephone access of several months to years. The growth of data services, of which the Internet offers an impressive example, is more recent.

4.2 Telephone Networks

Two Americans, Elisha Gray- a trained and disciplined scientist- and Alexander Graham Bell- a teacher of the deaf and a tinkerer- were exploring independently of each other the possibility of sending multiple messages on a single telegraph circuit by using frequency separations. We know Alexander Graham Bell as the inventor of the telephone because he got to the Patent Office a few hours before Elisha Gray on 14 February 1876. By the 1880s, the telephone was well on its way in business-office uses, and soon to become a residential item.

Evolution of point-to-point telephone circuits into a switched network:

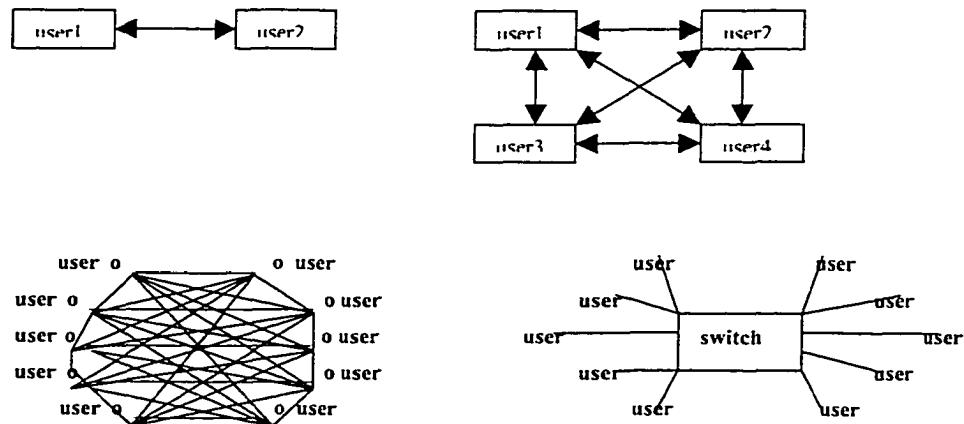


Figure 4. 1: Evolution of Networks

As additional users are added to the system, the need for connection grows exponentially, until one reaches practical limits on the number of linkages. The problem is solved by connecting users to a Central Office (CO), called a “switch”. The early switch was a manually operated toll switchboard. Most of the address decoding, routing and interconnect was done by skilled operators. This network is referred to as Plain-Old-Telephone Service (POTS). A telecommunication network is composed of three major elements: switching, transmission, and customer-premise equipment.

4.3 Evolution of Telecommunications Networks

The concept of stored program was initiated by John von Neumann during the 1940s. In 1965, AT&T commercially introduced the first software-driven (stored-program control, SPC) Central Office, the 1ESS (electronic switching system) switching system in Succasunna, NJ. The first software program for the 1ESS switching system had approximately 100,000 lines of code. Other major developments are as follows:

4.3.1 Local Switching Environments

This section discusses local switching environments in USA, Europe, and Japan. In Europe, British Telecom in the United Kingdom, Telefonica in Spain, and Societa Italiana per l'Esercizio (SIP) in Italy have adopted the IN (intelligent network)-type of network modernization for offering newer services.

4.3.1.1 United States

Several different systems like the 2ESS, 2BESS and 3ESS were introduced during late 1960s and early 1970s. The 2ESS became 3AESS with faster logic and automatic code conversion.

Switch in Toll Environments:

The electronic translator with SPC Processor No. 1 was introduced within the 4A Crossbar Toll system in Grand Rapids, MI, in 1969- the first toll system to include some of the functions and features of computerized systems.

The 4ESS, a logical and managerial extension of 1ESS, with a trunk capacity of 100,000 calls/hour was introduced in early 1970s which can complete

800,000 calls/hour (peak time about 1,200,000 calls). The synchronization between switch clock and network clock is retained. The 4ESS reduced the fixed costs and running expenses. It can interface with DS3 carrier systems.

The next one, 5ESS, was introduced in 1982 in Seneca, IL. The multimodule 5ESS came in 1983 in Sugar Grove, IL that offers several distinct features. It can serve: operator, local, toll and international. It is very reliable. The operating system of 5ESS is encoded in C language and permits CO capabilities to be added easily.

Network Protocol Architecture

The most basic reason for the existence of a communication protocol is to ensure that two communicating entities can send, receive, and interpret the information they wish to exchange. A protocol architecture is a framework into which protocol functionality is made to fit.

Networking would be facilitated and the expense much reduced if computer manufacturers and customer equipment vendors could agree on a common communication interface protocol. In 1978, the Geneva-based International Standard Organization (ISO) set forth the Open System Interconnection (OSI) reference model.

4.3.1.2 Europe

- **United Kingdom**

Two major business divisions in British Telecom are the national networks division, which caters to telephone and specialized services, and the local exchange networks. The analog derived services digital network (ADSN), implemented in two states, has offered the Freephone (0800) service since 1985 and the premium (0898) service since 1986. Eight participating analog switching nodes in ADSN have been replaced by digital switches, thus converting the ADSN to DDSN and offering full intelligent network services in the United Kingdom. The five building blocks of the network are: 5ESS® switching system, Network control point (NCP), Network subscriber transaction, administration and recording system (NETSTAR), Network services complex (NSCX), and Multifunction operating system (MFOS). Several distinct features of the British DDSN are: Alternate destination on busy, Call rejection, Call queuing, Digital private network signaling system (PNSS), Call processing enhancements, Call reporting, and NETSTAR resource logging and usage reporting, and Originating call signaling.

The DDSN's eight 5ESS® action control points (ACPs) are distributed at Cambridge, London, Birmingham, Leeds, Manchester, Guildford, Bristol, and Glasgow [1].

- **Spain**

The socioeconomic climate for Telefonica de Espana is unique. Only about one third of the market share are state owned. Poor service and rising telephone demand prompted a modernization plan to design an integrated digital network (IDN) with a Common Channel Signaling System (CCSS) and packet switching capability meeting all the multiple services ISDN requires. In this network, ISDN and the IN services are introduced concurrently. Digitization of Telefonica's PSTN gave rise to the very first BRISDN (2B+D) and PRISDN (30B+D) services in the early 1990s. A tandem switch may be added between the local exchanges. The signaling, facilities, and administration are considerably different than other national exchanges and are unique to Spain [1].

Initially, six major switching centers were put into operation throughout Spain in Barcelona, Bilbao, Leon, Madrid, Sevilla, and Valenica. Most of the major ESS™ (for example, 5ESS®) can be restructured to provide both the switch and service switching point (SSP) functions to detect the trigger conditions for IN

services. The major switching systems also serve as interrogators and receivers of network control information from the NCPs.

Signaling in Telefonía is conducted via a specialized signaling network. The routing and rate sometimes depend upon the particular source and destination for the transaction. For example, the ACPs are fully interconnected with 2.048 Mbps (E1) lines with CCISS7 (Spain) for the pulse code modulation (PCM) circuit-switched channels, whereas these offices are interfaced to NSCXs with 64 kbps CCS7 (North America) for circuit-related signaling.

- **Italy**

The state controls the Italian postal and telecommunication services. Two major organizations constitute the Ministry of Post and Tele-communications: Società Italiana per l'Esercizio (SIP) and Azienda di Stato per I Servizi Telefonici (ASST). SIP is responsible for these services: local telephone customer relations, design and construction of local, sector and district telephone networks, local telephone and auxiliary services, and long distance telephone services with ASST; and border international telephone service as agreed with Italian and Foreign administration. ASST is responsible for services: design and construction of the domestic telecommunication between the compartments and international network, long distance communication with SIP, international telephone

communication with European and neighboring countries and islands, and other administrative duties with the concessionaries.

Two public networks serve Italy's voice and data communication needs. Rete Publica (PSTN) and Rete Foni Dati (RFD). Operational since 1983, RFD deploys digital switching exchanges, such as AFD (digital switching centers), and peripheral concentrators, such as UFD (peripheral concentrators). Five AFDs reside in Milan, Turin, Rome, Bologna, and Naples, and connect to numerous UFDs dispersed throughout the country [1].

- **France**

During an explosive surge of growth for the French PSTN during the mid 1970s, an archaic version of the VPN (virtual private network) concept was in place for industry. Four Alcatel E10N3 exchanges in the Paris area could be used in the transit mode to provide interconnect capability for large businesses via leased lines. The service called Colisee to provide interconnect capability for large businesses via leased lines. The service is still viable today and has ISDN capability with an added Alcatel (MT25) exchange and provides interconnect with other PVNs from other carriers. The early versions of the French IN have the PSTN exchanges (Alcatel E12s) connect to the SSP (service switching point)

served by the SCP (service control point) with routing database, service database, and a charging center [1].

4.3.1.3 Japan

Japan has a number of computer and digital networks. However, there are about five established data networks (N1 net, JUNET, BITNET JP, WIDE INTERNET, and Inter-Agency Research Information Network). Telephone service is well established throughout Japan, but the customer calling patterns are different, based upon the cultural tendency (that is, face-to-face communication) and the call tariff structure imposed.

In the Japanese (NTTs, Nippon Telephone and Telegraph) version, the NSSP (network services support point) has the customer database and communicates with the NSP (network services control point) via an X.25 packet network. The switching systems and the transmission links make up the transport layer for the data to be communicated to and from the service users. The Group Center (GC) Central Offices receive the trigger conditions from a group of zone centers and communicate the specific request to the NSP of the CCS7 traffic control network. A wide variety of new services are easily implemented by three essential plans: (1) Numbering plan, which identifies the service (such as 800 or

900); (2) Connection plan, which identifies the routing or performance; and (3) Charging plan, which provides sufficient information to the billing and accounting system.

Different network control point algorithms are activated, depending upon the prefix derived from the novel service associated with the call. Often in Japan, such new services are simply added and dropped by the IN architecture, because they tend to be privately owned and tightly controlled [1].

4.4 OSI Model

The basic objective of the OSI reference model is to standardize the rules of interaction among interconnected systems. It is not the intent of OSI to standardize the internal operation of a system. A system is considered to be one or more autonomous computer(s) and the associated software, and users that are capable of information processing and/or transfer. The purpose of the OSI reference model is to allow any computer anywhere in the world to communicate with any other, as long as both obey the OSI standards.

The OSI model divides the communication process into seven layers. The seven-layer OSI architecture is for data networks. The OSI model helps in understanding the design of packet-switched network architectures. One of the most important advantages of OSI model is the ability to truncate the communication process at any OSI layer at any node, and then resume the communication process at any other node at any other time, up or down the model.

4.4.1 OSI Layers

- **Physical Layer (PL):** The function of PL is to activate, maintain, and deactivate the physical connection.
- **Data Link Layer (DLL):** Synchronization of data streams and error control.
- **Network Layer (NL):** Provides, maintains, and terminates the connections in a switched environment of the end systems. Also handles routing and addressing information.
- **Transport Layer (TL):** Controls the end-to-end information exchange with the required degree of reliability.
- **Session Layer (SL):** Binds and unbinds various users and their applications into a logical units for dispatch and recovery to and from the network during a session.

- **Presentation Layer (PRL):** Interprets the information exchanged between the Application Layer and the network.
- **Application Layer (AL):** Assures the user that the service provided is consistent with the type of application.

4.5 Integrated Services Digital Network (ISDN)

The primary problem with present telecommunications networks is that separate networks and separate interfaces are required for every application. Both voice and data circuits can have either a digital or an analog interface, but the interface and terminating equipment are far from uniform. Analog networks are slow and inefficient to handle data, and low-speed data networks are ineffective for voice transmission. Hence, ISDN was proposed to provide connectivity over a single digital communications pipeline while opening the network to a range of services that are impractical with the amalgam of networks. ISDN is a digital service provider to its customers. The ISDN architecture was approved in 1984 by CCITT (International Telegraph and Telephone Consultative Committee, now called ITU-T, International Telecommunications Union-Telecommunications Standardization Sector).

The reason for the ISDN architecture is to allow the telecommunication systems to evolve without worrying about interface to other systems. ISDN is only concerned with the interface to a customer or a switching system, not with how the system operates. It uses the basic architecture of the OSI with enhancements to support out-of-band common-channel signaling.

4.5.1 ISDN Objectives

- To provide end-to-end digital connectivity
- To gain the economies of digital transmission, switching, and signaling.
- To provide users with direct control over their telecommunications services.
- To provide a universal network interface for voice and data.

The building blocks of the ISDN are:

- Bearer services
- Channels
- Interfaces
- Message sets

The ISDN will offer flexibility so users can initiate and terminate services from their own terminals. ISDN is an attempt to diversify the bearer services offered by the telephone network. The different services are provided by different

networks (rather than in one single network) accessed through a common ISDN switch. Diversity is limited because these services are built on top of the traditional 64 kbps channels, which constrains the bit stream rates that can be supported.

ISDN is limited to the ITU standardized signaling information on the D (delta) Channel. ISDN operates on four commonly used channels. The four basic channels in the United States are:

- **B (bearer) channel (64 kbps):** Information bearing channel that provides a transparent digital path between one access point of the network and another (user-to-user information transfer).
- **D (delta) channel (16/64 kbps):** Supporting channel that carries out-of-band signaling information-bearing B channels. The D channel standards are: 16 kbps for BRISDN; and 64 kbps for PRISDN.
- **H0 (384 kbps):** user-to-user information transfer
- **H1 (1.536 Mbps):** user-to-user information transfer

BRISDN (Basic Rate ISDN) depicted in Figure 4.2:

$$2B + D = 2 * 64\text{kbps} + 16\text{kbps} = 144 \text{ kbps}$$

B1 Channel: The number one B channel of BRISDN service that is used to carry 64 kbps PCM voice data for telephone service

B2 Channel: The number two B channel of BRISDN service that is used to carry 64 kbps packet-switched data.

PRISDN (Primary Rate ISDN), depicted in **Figure 4.2 :**

- **NA:** 23B + D. 24 channels * 8 bits + 1 bit (frame synchronization bit) * 8000 samples/second = 193 bits * 8000 samples/second = 1.544 Mbps.
- **Europe:** 30B. (32 channels * 8 bits)/125 us = 2.048 Mbps

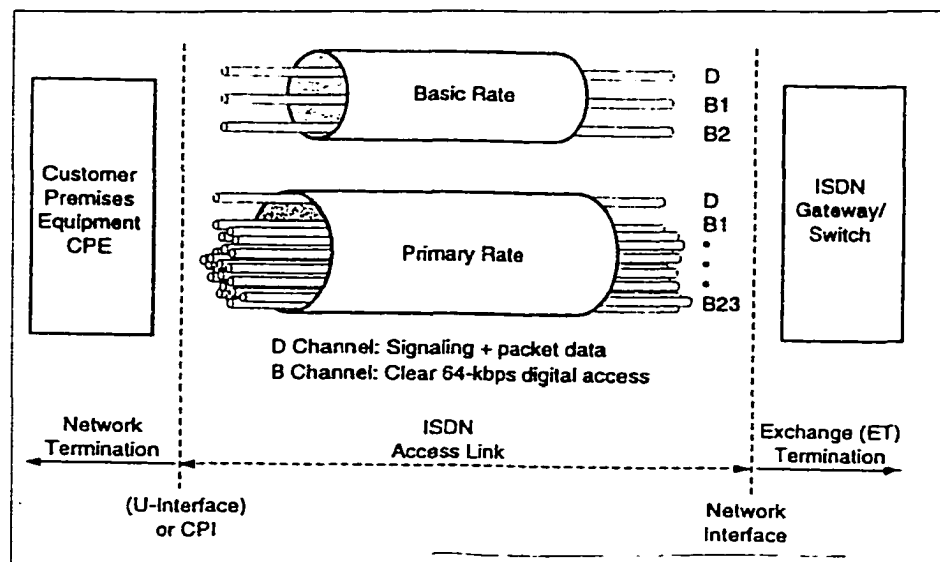


Figure 4. 2: ISDN Access from Customer to an ISDN Gateway/Switch
The Primary Rate in Europe is 2.048 Mbps or E1 with 30 Bearer or B Channels and one Delta or D channel.

4.5.2 Evolution Toward BISDN (Broadband ISDN)

The BRISDN at 144 kbps and the PRISDN at 1.544 Mbps (USA) and 2.048 Mbps (Europe) support a wide range of services. However, high-bit-rate services such as image and video services cannot be provided on 64 kbps ISDN. The advances in optical fiber, computing switches and other technologies such as digital signal processing (DSP) and video encoding techniques have stimulated a significant and rapid increase in the demand for new communication services and made broadband ISDN (BISDN) possible. It is the increasing demand for these high-bit-rate services that primarily triggered the evolution of BISDN. The availability of high grade fiber for transmission and the profusion of the rather inexpensive ATM switches are the major driving forces behind broadband digital networks. BISDN is based on fiber optic technology. BISDN allows for bandwidths between 1.50 Mbps and 600 Mbps.

BISDN Services

- Full-motion video and HDTV
- Image, videotelephony, videoconferencing
- Videotex, video surveillance, data, e-mail
- Voice, video and voice mail, data transactions
- LAN interconnections, and high-speed data communications

All of these services demand high-speed transmission and switching within the network.

The main technological innovations in circuit-switched networks begin with SONET (Synchronous Optical Network). The development of SONET was spurred by advances in the accuracy of clocks and in optical transmission.

4.6 SONET (Synchronous Optical Network)

The main technological innovations in circuit-switched networks begin with SONET. The development of SONET was spurred by advances in the accuracy of clocks and in optical transmission. SONET standards originally proposed by Bell Communications Research, specifies the standardized formats and interfaces for optical signals. SONET is ITU standard. It encodes bit streams into optical signals that are propagated over optical fiber. SONET's high speed and its frame structure permit it to support a very flexible set of bearer services [1]. In addition to providing higher speeds, SONET offers bandwidth-on-demand capabilities through SONET-based multiplexing technology.

The most important feature of the standard is that all clocks in a SONET network are synchronized to a common master clock so that the simple TDM

(Time Division Multiplexing) scheme can be used. SONET's frame structure and multiplexing methods also simplify equipment. The byte-interleaved TDM makes demultiplexing easy because individual streams can be determined simply by position in the frame.

SONET equipment is cheaper than current equipment. SONET provides higher speeds than the current transmission system. SONET provides end-to-end fixed-rate channels at speeds that are multiples of 51.84 Mbps. These channels can be used in a flexible manner to support a variety of bearer services. SONET can carry current telephone bit streams as well as it can be used to transport ATM cells. The transport of ATM cells over SONET is the basis of BISDN- the universal network. Its flexible frame structure accommodates both asynchronous traffic such as ATM and synchronous transfer mode (STM) traffic such as voice.

4.6.1 SONET Frame

In February 1988, the Synchronous Transport Signal, shown in **Figure 4.3** was accepted with 9 rows and 90 columns and at a bit rate of 51.84 Mbps. At this rate, the time duration of the frame is exactly 125 μ s (at OC-1 rate). This

corresponds to one 8 kHz voice sample- (This gives the STS-1 rate of 810 bytes/frame x 8000 frames/second or 51.840 Mbps).

Row	Header (3 bytes)			Payload (87 bytes)
1				
2				
3				
4				
5				
6				
7				
8				
9				

Figure 4. 3: SONET Frame
Header: 3 bytes, Payload: 87 bytes

SONET standards are being incorporated for asynchronous transfer mode (ATM) in broadband ISDN applications for voice and data. The standards for fiber optic digital hierarchy range from the OC-1 rate of 51.84 Mbps through OC-3 rate of 155.52 Mbps to the OC-192 rate at 9.95328 Gbps (see **Table 4.1**).

SONET Transport	Optical Carrier	Data Rate = 51.84Mbps * STS#
STS-1	OC-1	51.84 Mbps * 1
STS-3	OC-3	51.84 Mbps * 3
STS-9	OC-9	51.84 Mbps * 9
STS-12	OC-12	51.84 Mbps * 12
STS-18	OC-18	51.84 Mbps * 18
STS-24	OC-24	51.84 Mbps * 24
STS-36	OC-36	51.84 Mbps * 36
STS-48	OC-48	51.84 Mbps * 48
STS-96	OC-96	51.84 Mbps * 96
STS-192	OC-192	51.84 Mbps * 192 = 9.96 Gbps

Table 4. 1: SONET Digital Hierarchy

4.7 ATM (Asynchronous Transfer Mode)

The basis for ATM is high-speed digital bearer which is divided into fixed length cells. ATM is a bearer service. Packet and circuit switched networks are suitable for message and constant bit rate traffic, respectively. ATM networks combine the good features of both types of networks, making them suitable for variable bit rate (VBR) traffic. ATM networks potentially can provide bearer services with a specified quality of service to meet the needs of all traffic types.

ATM networks are designed to support both real-time applications such as video connections and telephone services and non-real-time applications such as

e-mail and file transfers. ATM also supports BISDNs. ATM is cheaper than telephone networks.

4.7.1 Main Features of ATM

- The service is connection-oriented, with data transferred over a virtual circuit (VC). The data stream from source to destination follows the same path.
- The data is transferred in 53-byte packets called cells.
- Cells from different VCs that occupy the same channel or link are statistically multiplexed.
- ATM switches may treat the cell streams in different VC connections unequally over the same channel in order to provide different qualities of service (QoS).

4.7.2 ATM Cell

ATM cell format is presented in **Figure 4.4**.

- **Header (5 bytes): GFC, VPI, VCI, PLT, CLP, HEC**
- **Payload: 48 bytes**

GFC	VPI	VCI	PLT	CLP	HEC	Payload

Figure 4. 4: ATM Cell Format

GFC: Generic Flow Control

VCI: Virtual Channel Identification

CLP: Cell Loss Parity

VPI: Virtual Path Identification

PLT: Payload Type Identification

HEC: Header Error Check

ATM layers are depicted in **Figure 4.5** [2]. ATM offers high-speed data transport capabilities. Since the ATM cells “ride” within the SONET frame, they inherit all properties of SONET. The speed of transport is derived from optical rates, optical speeds and low switching times (because of the simplicity/elegance in the SONET header information).

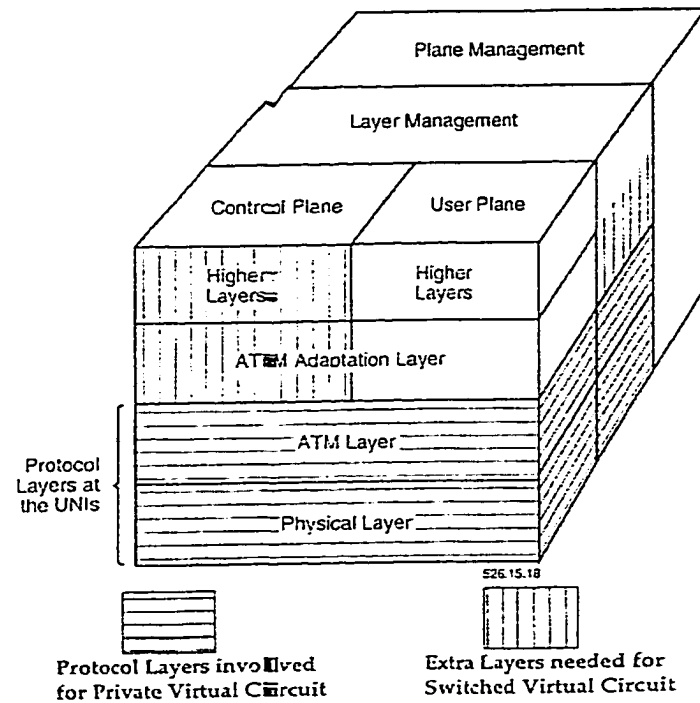


Figure 4. 5: Overview of the ATM Reference Model for B-ISDN Applications

ATM – Asynchronous Transfer Model; B-ISDN – Broadband ISDN; UNI – User Network Interface

ATM standards are applicable within LAN/MAN/WANs and to private LANs, campus networks, or any other data networks. These networks can be connected to public, regional or national ATM networks. They can also serve the needs of B-ISDN by functioning via connected-oriented ATM switches.

4.7.3 ATM Cells Within SONET Frame

SONET and ATM technologies provide additional applications. Typically, the packet networks are controlled for their flow and for any possible errors. Being highly multiplexed and time-shared, they are highly efficient and very inexpensive. Broadband services requiring more than one STS-1 (Synchronous Transport Signal Level 1) payload capacity are transported by concatenated STS-1s. For example, ATM services requiring 135 Mbps can be carried by three concatenated STS-1s, denoted by STS-3c, the transport overheads and payload envelopes of which are aligned. **Figure 4.6** depicts an STS-3c frame overhead and information payload format. In this STS-3c, the first of three H1 and H2 bytes contains a valid SPE pointer, whereas the second and third H1 and H2 bytes contain a concatenation indicator that prevents the STS-3c signals from being demultiplexed.

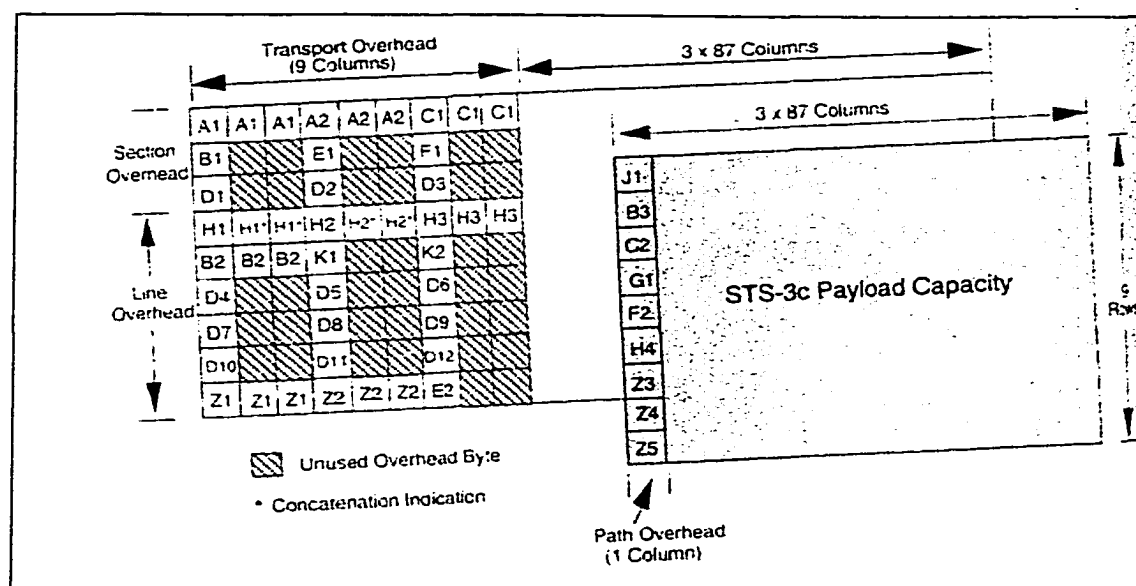


Figure 4. 6: SONET STS-3c Frame Structure

A1, A2: Framing; B1, B2: Bit Interleaved Parity (BIP)-8; C1: STS-1 ID; D1-D412: Data Com; E1, E2: Openwire; F1: User; H1, H2: Pointer; H3: Point Action; K1, K2: Automatic Protection Switch (APS); Z1, Z2: Growth.

J1: Trace; B3: BIP-8; C2: Signal Label; G1: Path Status; F2: User Channel; H4: Multiframe; Z3-Z5: Growth.

Concatenation specifies that these signals are considered as a single unit and transports them as such through the network. Compared with an STS-3 (nonconcatenated), the STS-3c can carry more information bits because only one set of path overhead (9 bytes) is required for the STS-3c. with the STS-3 SPE, one path overhead is required for each STS-1. In ANSI T1S1, STS-3c and STS-12c formats carrying ATM cells have been specified. These high-speed SONET interface cards are commercially available [68].

In ANSI/T1S1 standards, ATM cells are carried through a SONET transport platform; in ITU-T, ATM cells may be carried through an SDH (synchronous digital hierarchy)- or cell-based transport platform. SONET is a North American standard synchronous transmission network technology for optical signal transmission that terminates from OC-1 (optical carrier-1, 51.4Mbps) to OC-256 (13.2Gbps), with up to OC-192 available today. The functions of the SONET layer in the B-ISDN reference model is to carry ATM cells at high speed and provide very fast protection switching capability to ATM cells whenever needed.

The SONET link provides a bit stream to carry ATM cells that are logically carried on VCs (virtual channels) within VPs (virtual paths) are then physically accommodated by SONET STS paths. A VC is a generic term used to describe a unidirectional communication capability for the transport of ATM cells. A VP is a group of VCs. Each ATM cell header contains an ATM label that uniquely identifies the VP and the VC to which the cell belongs. The ATM cell label is the concatenation of the VPI (virtual path identifier) and VCI (virtual channel identifier).

The transport technologies used for SONET and ATM layers are completely different. SONET transport uses STM (synchronous transfer mode)

based on time slot interchange technology to cross-connect SONET paths; ATM transport uses cell switching technology to switch and/or cross-connect cells.

As specified by ANSI/T1S1, ATM cells are transported through SONET STS paths. SONET path formats that have been standardized to carry ATM cells are STS-1, STS-3c, STS-12c, and STS-48c, with STS-192c likely in the near future [68]. In addition to these service mappings, the ATM Forum has specified the ATM cell mapping formats for DS1, DS3, and 25.6 Mbps (over twisted pair) interfaces, as shown in **Table 4.2** [68].

Standards Organization	Interface Speeds
ITU-T	STM-1, STM-4, STM-16
ANSI	DS1, DS3, STS-1, STS-3c, STS-12c
The ATM Forum	DS1, DS3, STS-3c, STS-12c, 25.6 Mbps (over twisted pair)

Table 4. 2: Standardized Physical Transport Systems

Figure 4.7 depicts an example of ATM cell mapping. In this figure, ATM cells are placed into the payload of an STS-3c and this STS-3c then multiplexed with other STS-3cs and STS-3s to form an STS-12 stream. Similarly, this STS-12 path can be multiplexed with SONET paths to form an STS-48 multiplex stream

for OC-48 transmission, providing the underlying SONET system is an OC-48 system.

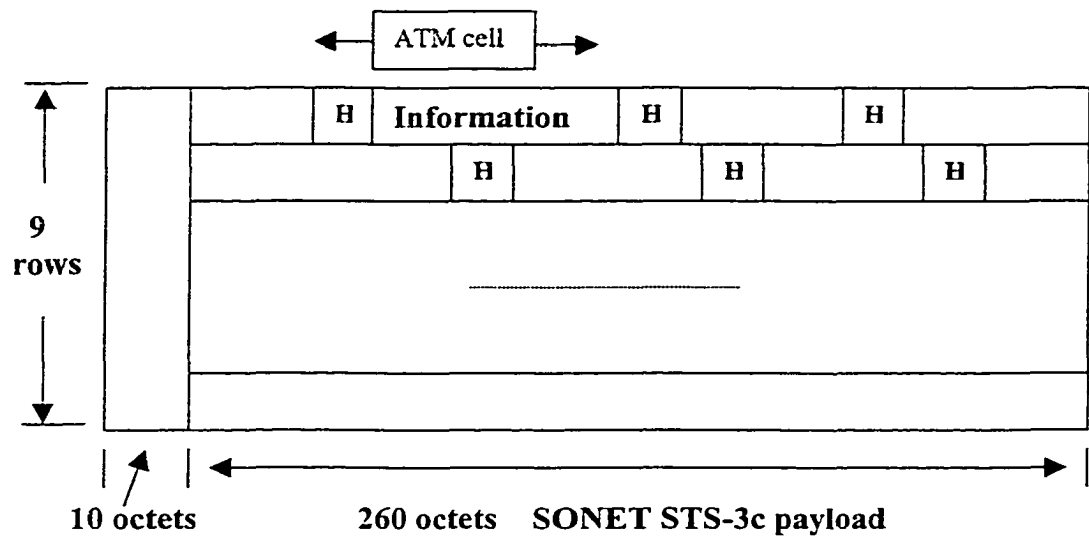


Figure 4. 7: ATM Cell Mapping on SONET STS-3c

4.8 OSI and ATM

The ATM functions occur at Data Link Layer (DLL) and Physical Layer (PL) of the OSI model (see **Figure 4.8** [2]). The DLL now has two distinct layers: ATM Adaptation Layer and the ATM layer. The PL is divided into two distinct sublayers: the transmission convergence sublayer (TCS) and the physical medium sublayer (PMS).

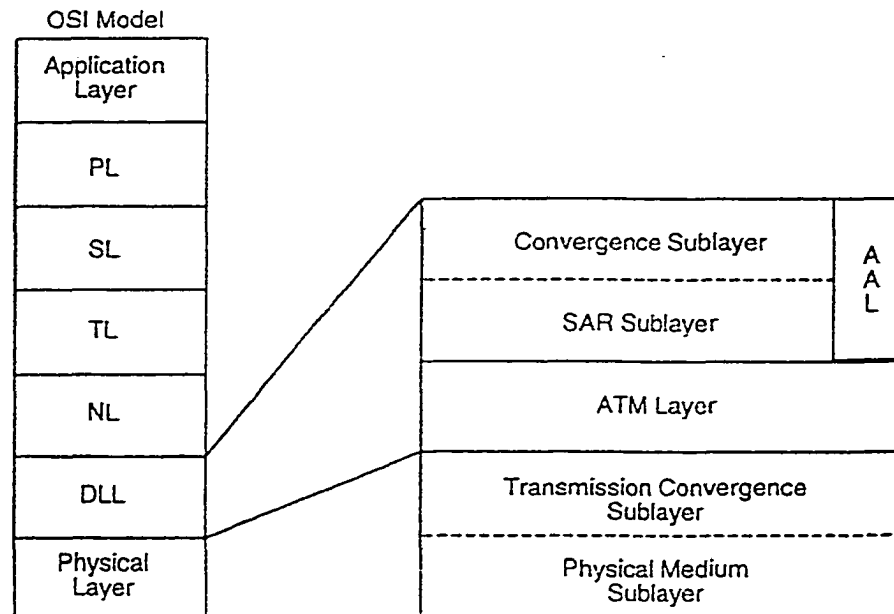


Figure 4. 8: ATM Layers within the Data Link and the Physical Layers of the OSI Reference Model

AAL – ATM Adaptation Layer; DLL – Data Link Layer; ISO – International Standards Organization; NL – Network Layers; OSI-RM – Open Systems Interconnection-Reference Model; PRL – Presentation Layers; SL – Session Layer; TL – Transport Layers

4.9 Intelligent Networks

Just as we entered the new millennium, we entered the Information Era, with its many challenges and rewards. Today, our society is becoming more and more complex as industry becomes more global and more interconnected. Hence,

we find communications becoming more and more essential to help us manage our time more successfully and become more productive. It is indeed a time of more- the needs to communicate have literally exploded. Computers have left the accounting arena in the form of personal computers and entered every aspect of our lives.

In the 1990s, applications expanded and interconnected, first within each industry and then across industries. In recent years, there have been dramatic changes and important success in the field of telecommunication networks. In the last decade, telecommunications has moved from a background role of a utility to applications meant to create new competitive advantages in business, increased productivity in public services, and economic development in cities, states, or nations.

The merge of telecommunications and computer technology is one of the most exciting developments of 20th century and changed the entire fabric of society. The new types of networks are beginning to grow at a rapid rate. Communication links are a vital part of society's infrastructure. In a computerized society, we look towards electronic communications. In many ways telecommunications will act as a substitute for the increasingly expensive

physical transportation. There will be no need to send mail physically; it will be cheaper to send it electronically.

With computers and communications merging into a new broad discipline, competition will center on services that promise a total solution to the end user. The requirements of the marketplace, advances in technology, and a rapidly changing regulatory picture are all converging to bring about this change- with a great impact on society. The combination of computers and telecommunications technology has already borne fruit. Banks are an obvious example; no major financial institution today could contemplate running its business without online systems that not only serve clientele more efficiently, but also provide an invaluable marketing tool. Airlines industry is another example. They computerized their reservation systems as well as their arrival and departure information.

4.9.1 Evolution of Intelligent Networks Concepts

The telephone networks around the world acquired some intelligence after the introduction of 1ESS (AT&T) in 1965. Three major milestones can be seen- in the United States- in this evolutionary path: (1) The concept of SPC (stored

program control) was introduced in 1965 for local switching; (2) The concept of CCISS6 (Common Channel Interoffice Signaling System 6) was introduced and implemented during 1976; and (3) The concept of service independence was introduced during 1984.

Intelligent Communication Systems represent the merger of telecommunications networks and computer networks to form intelligent networks (INs). INs actively interface with users, adapt and fulfill the switching and transmission requirements to convey any information to anyone authorized to receive it at any place and at any time. The quick access and dependability of computer systems are incorporated in these networks. The INs are adaptive, algorithmic resourceful, responsive and intelligent.

INs Functions

- Communicate relevant information from point A o----->o B
- Control network to facilitate the data flow
- Carry out necessary computations to seek out specific information

The requirements to be classified as an IN are: (1) The databases in SCP (Service Control Point); (2) CCS7 (Common Channel Signaling 7) network; and (3) Some service independence by having numerous databases in SCP.

INs are defined as the carriers of information with distinct algorithmic adaptation from one point to another in the network. Intelligence is embedded in (1) **hardware**: the customized integrated circuit chips; (2) **software**: coded as programs, utilities, or modules. Intelligence can also reside in the active memory of computer during the execution phase; and (3) **firmware**: microcoded into the control memories of the monitoring computers.

Basic computers (hardware, software, firmware) which control, monitor and process the information become an integral part of intelligent networks. Different INs have been designed by various long-distance carriers in the United States. They are:

AT&T

- **Universal Information Services Network (UISN), 1989, Figure 4.9 [1].**
Services: 911 (Public Emergency) Service, Virtual Private Network (VPN) and Automatic Calling Card.

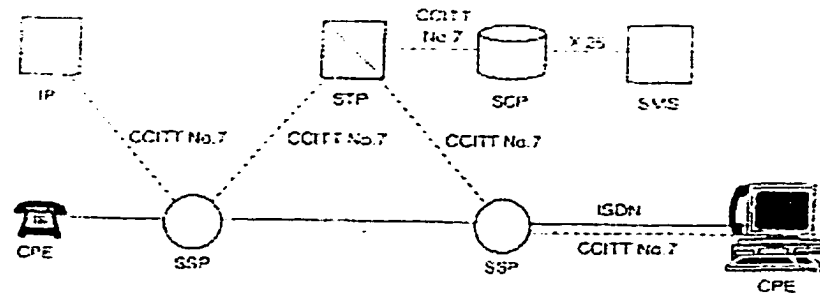


Figure 4.9: General Architecture of the UISN in 1989

IP = intelligent peripheral; SSP = service switching point; CPE = customer premises equipment; STP = signal transfer point; SCP = service control point; SMS = service management system; CCITT = International Telegraph and Telephone Consultative Committee, now called (ITU-T); ISDN = integrated services digital network.

- **Advanced Intelligent Network (AIN): Platform 0 and Platform 1**

Belcore

- **IN/1, 1987**
- **IN/1+, early 1990s**
- **IN/2, 1990s**

MCI

- **Vnet.** Uses two data access points (DAPs). The function of DAPs is similar to STP. DAPs are controlled by network control system (NCS- function is similar to SCP & BMS).

Sprint

- **VPN (Virtual Private Network).** Uses two STPs and several database control points (DCPs). The STPs are managed by a DCP management system.

The development of INs has taken place in several other countries for similar services. For example, Green Number Service (GNS) in Europe. Since 1987, Freephone Service, prevalent in British Telecom, has been available as GNS.

Other networks exist in the United States offering the following services:

- **ABS (Alternate Billing Service)**
- **ERS (Emergency Response Service)**
- **PVN (Private Virtual Network)**
- **AWC (Area Wide Centrex)**
- **PPV (Pay Per View)**

4.10 Intelligent Networks/1 (IN/1)

Bellcore introduced the first-level IN, namely IN/1 or the 800-network in 1987 in the United States (see **Figure 4.10 [1]**). The building blocks of IN/1 are:

(1) SSP (Service Switching Point); (2) SCP (Service Control Point); (3) SMS (Service Management System); and (4) STP (Signal Transfer Point).

As shown in the figure, IN/1 contains many SSPs and STPs, but only one SCP and SMS. The STPs are within the Common Channel Signaling 7 (CCS7) network. IN/1 satisfies the three requirements (a. service specific databases in SCP; b. CCS7 network; c. service independence by having a number of databases in SCP), as can be seen in the figure, to be an intelligent network. The services provided by IN/1 are: (1) 800 number service; (2) Alternate Billing Service (ABS); and (3) Private Virtual Network (PVN) Service. The subscriber-generated queries go up to the service-specific databases, and the call-completion information is returned to the appropriate switching office.

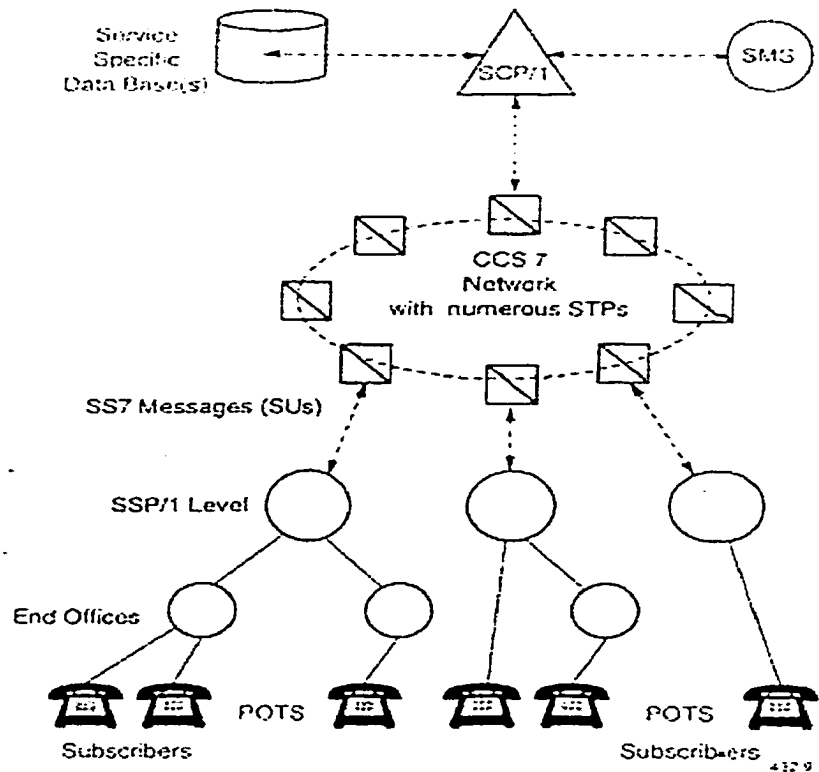


Figure 4. 10: Bellcore Architecture of IN/1 (1987)

SCP = service control point; SMS = service management system; STP = signal transfer point; SU = signaling unit; SSP = service switching point; POTS = plain old telephone service.

4.10.1 Service Switching Points (SSP)

SSPs are physical/logical entities. They are located at or near the switching systems. The switching systems contain call processing software which

monitors all steps required to complete any call through the network. SSPs functional aspects are: (a) the physical switch that performs the switching function; and (b) the logical software modules, located in the switch, execute all SSP functions.

Each SSP reacts to specific trigger conditions from the customers. It then transmits queries to SCP through STP via Signaling System 7 (SS7). SSP instructs the switching system to complete the call upon receiving the response to the query from SCP. The switching system executes the call-processing software to complete the process.

4.10.2 Service Control Points (SCP)

SCP, a physical/virtual node, contains the active database of customer records. There is a database for each service provided by IN/1. So there exist three databases: (a) 800 number database; (b) ABS database; and (c) PVN database. The database is actively queried by SSP within the network to obtain information for service completion. Thus, effective and efficient management and utilization of this database is crucial.

SCP is considered like an efficient parallel processor with mass storage-retrieval facility because it can handle a very high volume of calls – several million calls per hour. SCP is made up of hardware and software components. SCP functions include the validation of the caller to call, the address translation of the called party (look-up database, if necessary) and identification of the carrier authorized to carry the call.

4.10.3 Service Management System (SMS)

SMS performs the administrative work for SCP. SMS is a support facility used to enter customer data into the SCP's database within IN/1 environment. SMS is also used to enter the call-processing software into the SCP. It communicates with SCP through interface cards that process the BX.25 protocol used for communication between SMS and SCP.

4.11 Intelligent Networks/1+

The Regional Bell Operating Companies (RBOC), in conjunction with the research done at Bell Communications Research (Bellcore), proposed the concepts of IN/1+ in order to overcome the limitations of IN/1- provide digital services associated with ISDN. The phased introduction of IN/1+ was during the

early 1990s (see **Figure 4.11 [1]**). The transition from IN/1 to IN/1+ is twofold: (1) the transplantation of more intelligence in the telecommunication networks; and (2) the introduction of ISDN services such as circuit-switched voice, voice band and packet-switched data services.

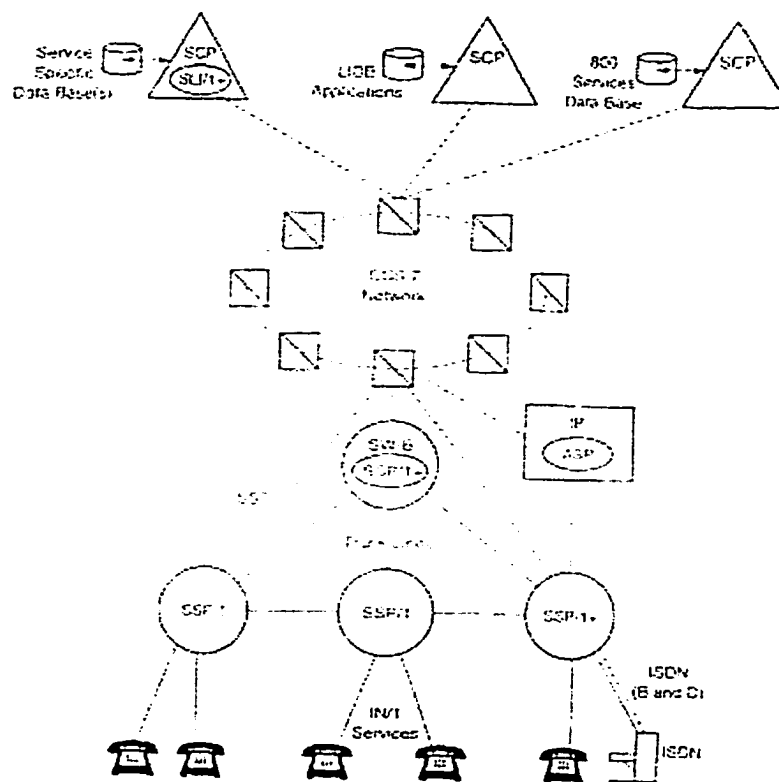


Figure 4.11: Network Architecture of In/1+

For IN/1, IN/1+ Routine Voice and Voice-Related Services and for ISDN. The IN/1+ services are limited to SSP/1+ lines. Numerous databases are individually accessed depending upon the type of IN/1+ service invoked.

SCP = service control point; SLI/1+ = service logic interpreter/1+; LIDB = line information database; CCS7 = Common Channel Signaling 7; IP = intelligent peripheral; ASP = adjunct service point; SW = switch; SSP/1+ = service switching point/1+; ISDN = integrated services digital network; SS7 = Signaling System 7.

The building blocks of IN/1+ are as follows:

- **SSP/1+**
- **INAP (Intelligent Network Access Point) Switch**
- **Non-SSP/1+, Non-INAP Switches (End Office)**
- **SCP/1+ with logical module SLI/1+ (Service Logic Interpreter)**
- **IP (Intelligent Peripheral) with a logical module ASP (Adjunct Service Point)**
- **STP/1+**
- **ISMS (Integrated Services Management System)**

IN/1+ consists of the IN/1 building blocks plus some new network elements such as INAP and IP. The IN/1+ services are based upon programmable collection of functional components (FCs).

SCP/1+ (Service Control Point)

The SCP/1+ is a physical unit that contains Service Control Logic (SCL). The SCP/1+ has four basic units: (a) the SCP node; (b) an efficient service logic

interpreter (SLI/1+); (c) service logic programs (SLP); and (d) the IN/1+ data.

The SLPs are driven by SLI/1+ to accomplish IN/1+ services.

New SLPs are encoded and debugged, outside of SCP/1+ environment, according to the service specifications. The function of SLI/1+ is to execute the SLPs simultaneously to perform IN/1+ services.

IP (Intelligent Peripheral)

IP is a network compatible element. IP contains an adjunct service-point (ASP). The ASP responds to the network requests to use the resources. IP is connected to an SSP/1+. The communication with STPs takes place via SS7 network. The IP is accessed by both the customer and operating-company personnel.

SSP/1+ Facility

The SSP/1+ switch is the physical unit and the SSP/1+ module is the software program contained in the SSP switch. The software module filters out any call that does not require IN/1+ functions. There are three possibilities for a facility: (1) a version of SSP/1+; (2) an end office (EO) with an INAP; or (3) an EO that simply forwards its IN/1 service calls to another EO with SSP/1+

capability. The differences between SSP/1 and SSP/1+ are significant. SSP/1+ is expensive to implement.

INAP (IN Access Point)

The INAP allows the triggers in the IN/1+ service requests and forwards calls to an SSP/1+ switch. If SSP/1+ facility is not available in an EO, the INAP logical module can be added in order to process the requests of the users.

ISMS (Integrated Service Management System)

ISMS facilitates the control by both the customer and the operating companies over many IN/1+ services. The five major functions of ISMS are as follows:

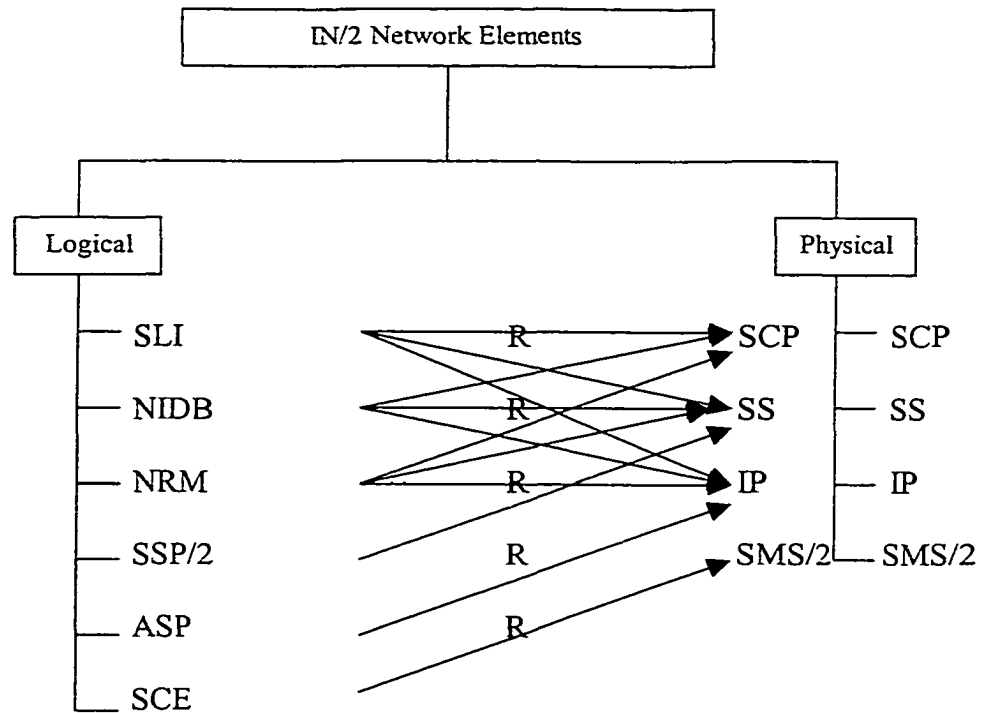
- ISMS receives/transmits data from/to the centralized administration and coordination (CA&O) pertaining to the independent networks of the participating service providers.
- ISMS exchanges updates of pertinent SCP, SSP and IP control information between the service providers so that the IN/1+ functions are transparent to the users.
- ISMS responds to the control information of its local service administration (SA) facility for updates and changes in the service environment.

- ISMS dispatches the SCP and updates it with information received from other ISMSs, CA&O facilities, and its local SA.
- ISMS dispatches the SSP and IP updates to the local service switching point and intelligent peripherals. This activity needs considerable management.

IN/1 is a subset of IN/1+. New services offered by IN/1+ are IN/1 services plus 900, 911 and additional services provided by IPs.

4.12 Intelligent Networks/2

The difference between IN/1+ and IN/2 is not very significant. IN/1+ is a transition phase from IN/1 to IN/2. Thus, IN/2 contains more hardware and software components to enable flexible services. The building blocks (physical/logical) of IN/2 and their associations are depicted in **Figure 4.12** [1].



Note: R indicates where the logical module may reside.

Figure 4. 12: A Complete Set of Logical and Physical Entities in IN/2

SLI: Service Logic Interpreter; NIDB: Network Information Database; NRM: Network Resource Manager; SSP/2: Service Switching Point/2; ASP: Adjunct Service Processor; SCE: Service Creation Environment; SCP: Service Control Point; SS: Switching System; SMS/2: Service Management System/2.

The architecture of IN/2 is very adaptable. Additional elements can be added for newer services. The vendor feature node (VFN) is absent, but this configuration provides some VFN functionality via the IP interconnected to the SSP/2 and the STP, thus reaching the SCP/2. The SCE (Service Creation

Environment) is another new feature of IN/2. It provides the staff of SMS/2 with service logic programs (SLPs) to SLIs. The SLPs are an integral part of the SLIs. The service programs can easily be modified or altered to customize any service. Customizing the service is as simple as altering the program structure of the individually written service programs that are presented to the SLIs for interpretation and execution of the network instructions.

The next version of IN/2 offers business community the IN/2 and ISDN services via the same facilities. The IN/2 services are introduced via VFN, IP or Universal Customer Interface (UCI, under the jurisdiction of SMS/2). VFN is associated with a switch (SS) for new vendor provided services. The memory administration system may or may not be necessary, depending upon the type of SS that is serving as the SSP/2.

The final version of IN/2 (**Figure 4.13** [1]) includes all logical and physical elements. The three essential functions to provide IN/2 services are:

- **Service Definition.** New services must be defined. And this is done by the service logic programmers. They use SMS of IN/2. They write SLPs which define the network actions specific to a particular service. The SLPs are a collection of functional components (FCs). They use the network resources to

accomplish specific submodular functions such as collect digits arriving from the customer, forward dialed number, etc.

- **Service Interpretation.** SLI performs this function. Interpret SLPs.
- **Service Logic Program Execution.** SLP must be executed to provide the new service. Queries can come from SSP/2, IP or VFN. SSP/2 dispatches a query to SLI. SLI interprets and executes the SLP for the requested service.

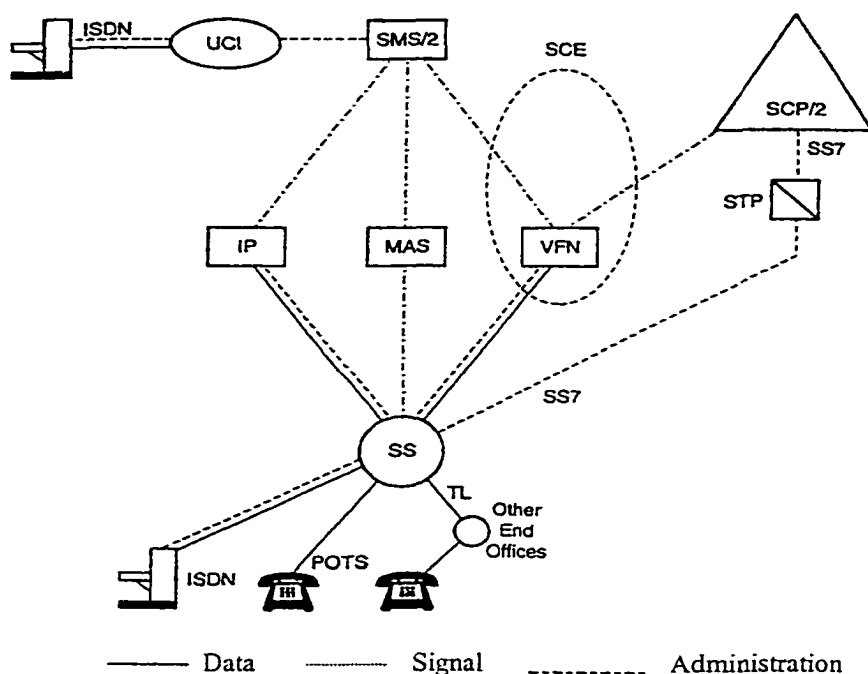


Figure 4.13: A Complete Architecture of IN/2

IP = intelligent peripheral; TL = turn line; SS = switching system; SCE = service creation environment; MAS = memory administration system; UCI = universal customer interface; STP = signal transfer point; POTS = plain old telephone service; SCP = service control point; SMS = service management system; VFN = vendor feature node.

VFN (Vendor Feature Node)

VFN allows outside service vendors to work in conjunction with IN/2. VFN provides services that the network may not provide due to regulatory or government constraints. For example, stock market reviews, propaganda material and telephone advertising services can be offered by outside service vendors.

The chief attraction of IN/2 is the creation of new services much faster than the IN/1 type of network. The network users take advantage of the hardware facilities and software expertise (through SCE) to create a new service, for example, opinion-polling, gather public response for a major event, video shopping services.

Operations Systems (SMS/2 and SCE)

The SMS/2 handles the customer record administration, coordination and control of the network databases. The SCE provides the software environment to introduce customer-controlled capabilities and/or vendor services through the network facilities. Coordination between the database and the network facilities is necessary to avoid conditions leading to possible “deadlock” situations.

NIDB (Network Information Database)

NIDB contains the customer-access and network connection information. The databases may be cross-queried by other modules in IN/2.

NRM (Node Resource Manager)

NRM locates and identifies the address of network resources that are required to continue call processing. NRM also provides the SLI with information required to establish a connection to a module in IN/2.

Global Intelligent Networks

Intelligent Networks have been born in other countries to provide services similar to the INs in the United States. For example, the British Telecom IN, Digital Derived Services Network (DDSN), Telefonica IN Spain, etc.

The INCM (Intelligent Networks Conceptual Model) of ITU-T has four planes (see **Figure 4.14** [1]):

- **The Service Plane**
- **The Global Functional Plane**
- **The Distributed Service Plane**
- **The Physical Plane**

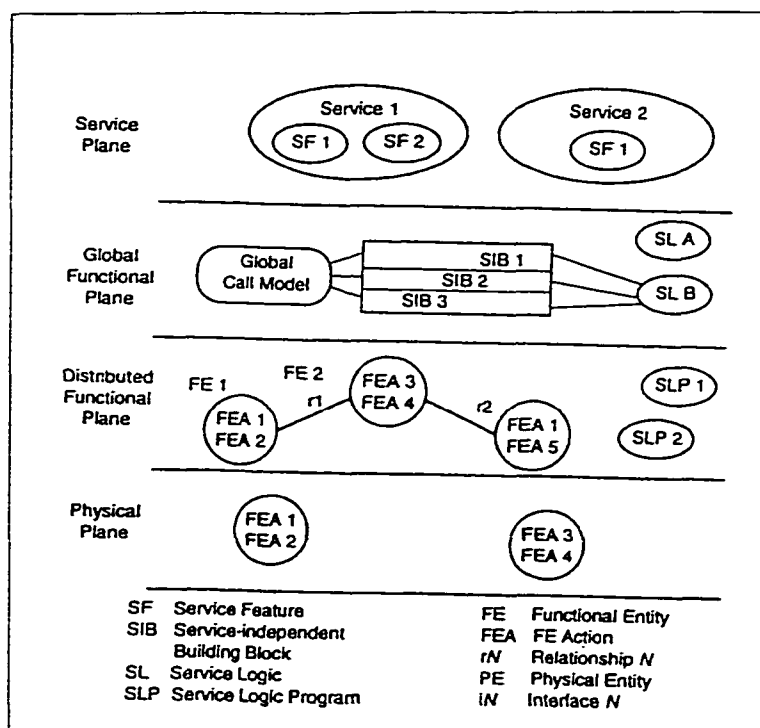


Figure 4. 14: Conceptual Model of Intelligent Networks (INCM)

Numerous IN capabilities are referred to as capabilities sets (CSs). Requirements for one or more of the six essential groups of functions within most INs and AINs (Advanced Intelligent Networks) are: (a) Service Creation; (b) Service Management; (c) Service Interaction; (d) Network Management; (e) Service Processing; and (f) Network Interworking.

Service Plane and Global Functional Plane at Functional and Physical levels are shown in **Figures 4.15 and 4.16 [1]**.

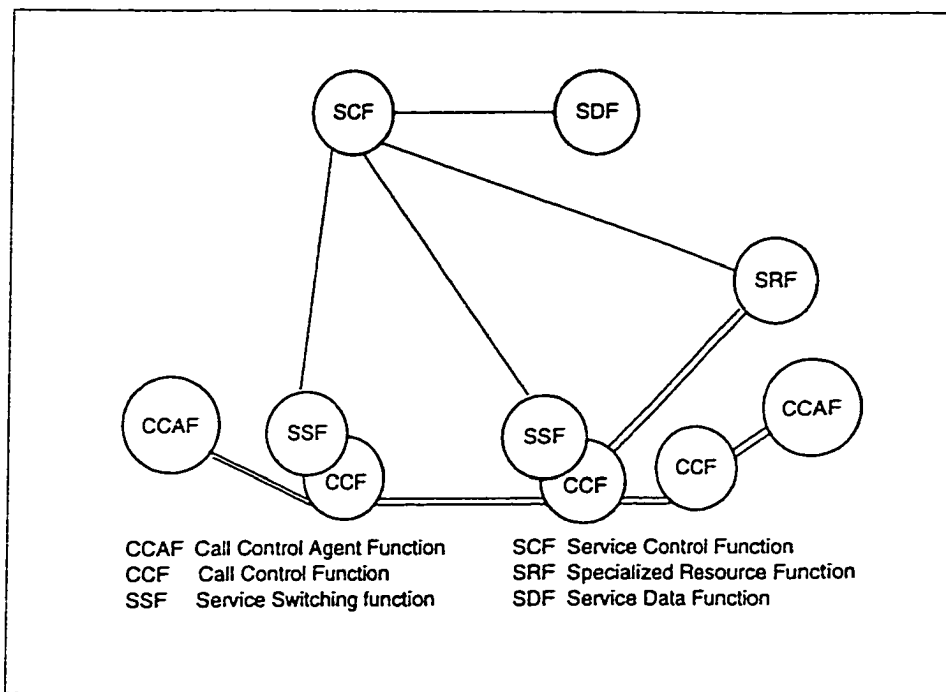


Figure 4. 15: Functional Plane Architecture of IN CS-1
CS = capabilities sets

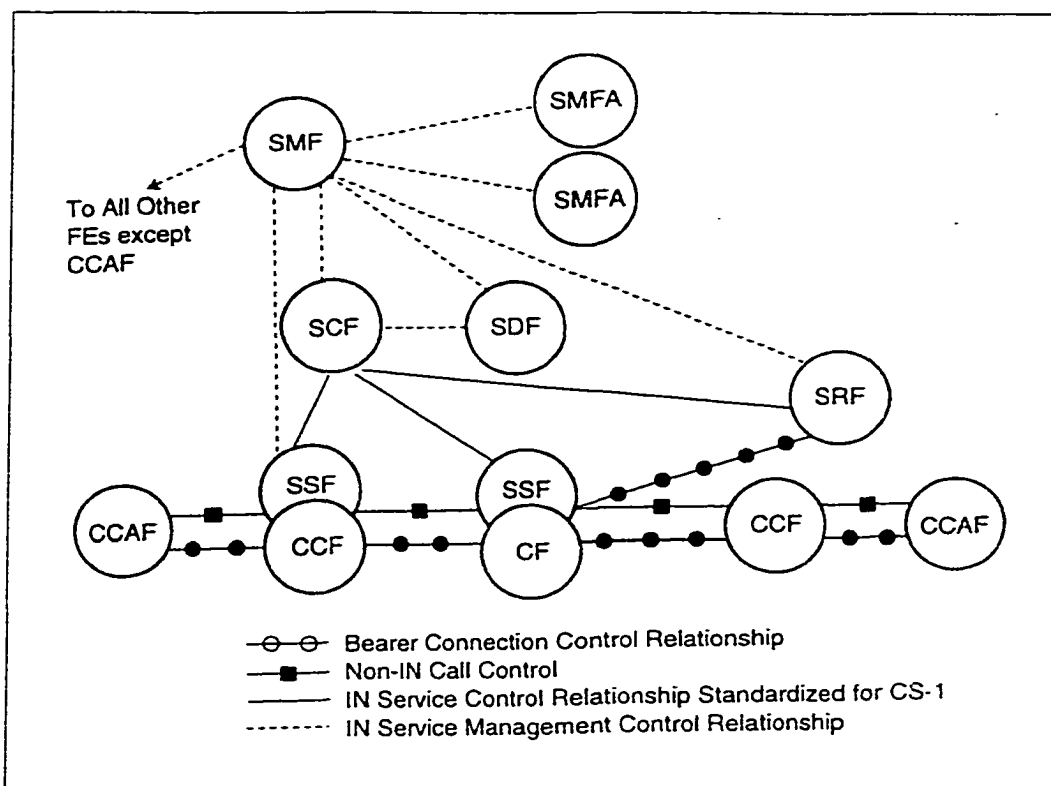


Figure 4.16: Functional Relationships for CS-1

(Q1211 specification). CCAF = call control access function; CCF = call control function; CF = control function; SCF = service control function; SDF = service data function; SMF = service management function; SMAF = service management access function; SRF = specialized resource function; SSF = service switching function.

4.13 Advanced Intelligent Networks (AIN)

The latest phase in modern communication networks is AIN and is tailored for business communication systems, introduced in the United States in

the 1992-98 time period. AIN has six basic components, already commissioned into service. They are:

1. Service Switching Points (SSP)

Hardware Aspects:

SSP must accomplish two categorically different tasks: (1) The conventional “computer-like” functions of the switch. The localized CPU controller is logically switched to interpret the opcode of the computer unit (CU) of the CPU. (2) The actual logical AIN channel switching (logical network channel switching functions). Three distinct pathways are essential: (a) Address of the control signal associated with the AIN operation and functions (for example, identify caller); (b) Control of the flow of data generated to actually operate the AIN (SS7 signals and SLI commands); and (c) Forces the hardware to perform logical channel switching (for example, completing a call on any given B or D customer channel).

Software Aspects:

The fifth and sixth generations of languages are expected to force the new SSP functionality to be more powerful and flexible. SSP functions will integrate database functions in the network language.

2. Service Control Points (SCP)

SCP contains the application programs that provide AIN services. The Service Logic Interpreter (SLI) is a resident in SCP. SCP can respond to messages from a switch and invoke the application programs.

3. Signal Transfer Points (STP)

STP controls the linkages to SSP, SCP and IP (Intelligent Peripherals) through SS7 network. The advanced signaling interfaces provide the signal flow. Most switching systems have access to this network via the SS7 protocol. The communication between SSPs and their corresponding STPs is highly standardized. Similarly, the communication among the STPs is standardized. This architecture permits Local Exchange Carriers (LECs) to control the new service creation process from the AIN platforms.

4. Intelligent Peripherals (IP)

IPs are installed to perform voice and data services in the public domain. Communication with the IP may occur over the SS7 network, the trunks/lines of COs (Central Office) or even ISDN line for a distant IP.

Voice services:

- Storage and retrieval of messages
- recognition of voice and its synthesis

- code conversion between the various coding techniques (example, adaptive delta pulse code modulation (ADPCM))
- pulse code modulation (PCM)
- toll quality, mu-law /A-law companding (Two different systems used to encode voice or audio signals which involve compression of the sampled signal according to a fixed formula.)

Data services:

- protocol conversion for narrow and wideband
- data-rate conversion
- video format (NTSC- National Television System Committee), high-definition television (HDTV), still picture etc.

5. Service Management Systems (SMS)

The function of SMS is intricately tied with the functions of the SCP. SMS is an off-line support facility used to enter customer data into the SCP's databases and the call-processing software into SCP. SMS communicates with the SCP through interface cards that process the BX.25 protocol used for communication between SMS and SCP.

6. Adjunct Processors (AP)

Adjuncts contain service logic and local customer data and have very localized capabilities in the SCP. Adjuncts support intensive services for centrex-type local transactions. Adjuncts are built into the AIN architecture with high-speed data or signaling links.

Two platforms exist for AIN services: Platform 0 and Platform 1.

4.13.1 Platform 0 Services

Services: 800-type network services in the international market; PVN (Private Virtual Networks) services; 700 number (follow me, connect to the called party); calling number-based (call forwarding services); and other specialized, customized local area services.

AIN Platform 0 and its earlier version (the Advanced 800 Network) are compatible with the LECs (local exchange carrier) in the United States and with some European network owners (British Telecom, Spain's Telefonica, and Italy's Societa Italiana per l'Esercizio – SIP).

4.13.2 Platform 1 Services

Services: Such as business and mobile wireless customers, network automatic-call distribution, CLASS (customized local area special services) can be enhanced via this platform to call forwarding on busy, customized call waiting, customized audio support and instant feature activation. Other new services such as medical, financial, etc. will be included. The emergency call forwarding service already exists.

ISDN services are provided by AIN. When all the capabilities of ISDN unfold, the eventual users are likely to be subscribers to multimedia services, video-on-demand, and other futuristic intelligent networks. ISDN is an important network because it provides integrated service facilities for POTS (Plain Old Telephone Service), facsimiles, PC connections (local, long distance, and global via bearer and packet-switched channels), and local area networks (LAN) services.

The emerging information networking services demand a large bandwidth from telecommunication networks for large volumes of data to be exchanged within a very short interval. Transport technologies to meet these broadband communications need SONET and ATM.

Chapter 5

5. COMMUNICATION NETWORKS IN DCs

Computer Communications in Developing Countries (DCs) take on added significance in light of low telephone penetration. Computers can be used to rapidly distribute text materials (including graphics etc), which need not necessarily be viewed on a computer monitor, but can also be printed as hardcopy to be displayed as posters, put on conventional bulletin boards, filed in subject folders etc. This is as important as the direct use of email and news by the relatively few individuals with access to a computer and phone line. In addition, the computer access to libraries and databases that is so useful in developed countries, could be even more useful in developing countries where direct library services are so poor.

In developing countries, the telephone tariffs would be especially high on long distance calls and even more so on international calls. These tariffs are likely to rise further instead of falling, as in developed countries, since there is a shortage of circuits. Lines are of low quality as well as congested and tariff structures are arbitrary and irrational. If tariffs are based on a minimum of 3-minute or even 1-minute calls, it may be important to persuade telephone companies to make a revenue-neutral change to charge by the second so that the full potential of email for reducing traffic can be realized. Developing countries need to invest more in network capacity to reduce congestion rather than concentrate on an increase in the number of subscriber loops.

Low quality and congested lines are more suitable for computer communications than for voice communications, when used properly. If voice communication is possible over a bad phone line, then data communication is significantly easier, and simply involves a lower data rate than normal, due to error correction, instead of the great difficulty in getting any intelligible message at all through on a bad voice call. High cost calls on the limited capacity long distance lines increases the relative advantage of text communications over voice. When international circuits were really expensive in developed countries, the bandwidth available was used primarily for telegraph traffic, with voice calls an occasional luxury. Where there is a phone service, but it is expensive to subscribe

to and use, computer communications can significantly reduce the long distance phone traffic and provide a hardcopy bureau service similar to, but much more convenient and cheaper than telegraph, telex and fax bureaus.

In developing countries e-mail should have an even greater advantage over fax due to the higher cost of dedicating a phone line to a fax machine (whereas it is comparatively easy for e-mail to use a nondedicated line shared with voice calls). The extremely low telephone penetration and lack of circuits between the capital cities and some major regional centers (let alone small towns and villages), severely limits the section of the population that can directly benefit from any form of telecommunications in developing countries.

Many developing countries do have access to networks, though at the moment the users are predominantly nongovernmental organizations (NGOs) or from the academic sphere.

5.1 Development of Telecommunication Technology in India

India is unique in the developing world. It has enormous population of one billion, 70 percent of whom live in 575000 villages. Only 8000 of India's

villages have a rural telephone exchange and about 40000 villages have long distance telephone public offices, few of which function [11]. There has been significant under-investment in the telecommunication sector in India. It has 6.1 million telephones, of which 90 percent are in urban areas. Although telephone services in large Indian metropolitan cities have improved in recent times, India reportedly has world's worst telephone service with only 0.8 telephones per 100 people compared to 75 in the USA, 65 in Japan and 7 in Malaysia [8]. About 40000 telephone exchanges most of which are the obsolete Strowger and Crossbar type which use a combination of electromechanical moving parts for switching telephone calls. These exchanges experience more wear and tear than the digital electronic exchanges, and are affected by temperature, moisture and other climatic conditions.

5.1.1 An Historic Perspective

As in other countries, the oldest telecommunication service in India is telegraphic service, introduced in 1851. Within four years 7000 kilometers of telegraph lines were created connecting the northeast and south. The British empire in India realized the advantages of telegraph in administering the vast continent. The telegraph network expanded rapidly and covered not only the

Indian subcontinent but also provided links to Malaysia, Tibet and Europe via Iraq and Iran.

Telephone service was introduced in the country in 1882, barely six years after it was invented by Alexander Graham Bell. Unlike the development of telegraph, the initial development of telephones in India was entrusted to private enterprises and limited to a few cities. When India and Pakistan separated in 1947 the independent state of India had 321 telephone exchanges with a capacity of 100,000 lines, 86,000 working connections, 426 long distance voice circuits, 338 long distance public call offices (PCO) and 3324 public telegraph offices. The progress of telecommunication since India's is shown in Table 5.1 [8].

Parameter		1948	1990
Local Exchange		321	14,285
Equipped Capacity (Million Lines)		0.1	5.3
Working Connections (Millions)		0.08	4.6
Long Distance Public Telephones		338	30,800
Public Telegraph Offices		3,324	36,400
Toll Exchanges		-	325
Telex Capacity (Lines)		-	52,000
Telex Connections		-	42,480
Toll Exchange Capacity (Million Lines)		-	0.7
Direct Distance Dialing (DDD) Station		-	867
Coaxial Cable (Route Km)	Analog	-	30,047
	Digital	-	2,540
UHF (Route Km)	Analog	-	12,747
	Digital	-	1,335
Fiber (Route Km)		-	1,615
Satellite Earth Station	Fixed	-	67
	Mobile	-	17

Table 5. 1: Progress of Telecommunication Since Independence

Although telecommunication service was launched in India in the middle of nineteenth century, progress was slow compared to developed countries, until India attained independence. When India became independent, the national government gave high priority to revitalization of the economy. As economic activity picked up, the need for rapid and effective communication facilities grew, leading to expansion of telecommunication facilities. The growth rate of telephones in India is depicted in **Table 5.2** [8].

Years	Number of Telephones (in Millions)
1980	2.8
1985	3.5
1990	5
1992	6.1

Table 5. 2: Growth of Telephones in India

India uses a five year planning cycle and during the eighth five-year plan (1992-1997) the number of telephones are expected to double to 10 million.

5.2 The Telecommunication Network

The telecommunication network handles over 40 million telephone calls daily and links the four metro centers in the north, south, east, and west (i.e.,

Delhi, Madras, Calcutta, Bombay) to the state capitals which in turn are connected to the districts, towns and villages. Nearly 500,000 people operate and maintain this network. Revenues from telecommunication operations exceed Rs. 40 billion (Rs: Indian rupees) per year. The Indian telecommunication network has recorded an average growth rate of 8 percent since independence. Although the growth rate has been good, it is patchy, with diverse types of switching and transmission equipment. Unlike the network of the developed countries, the Indian network is characterized by low telephone density, 0.8 percent compared to 12 percent of population in the rest of the world. The second problem is long waiting time for new applications 2,000,000 million applicants wait listed in two to ten year waiting period, and the third problem is heavy congestion on local and long distance routes during busy hours – two are three attempts per successful long distance call, 15 call units traffic per line per day; eight to ten BHCA/line. The fourth problem is -diverse type switching, signaling and transmission system- over ten types of switching systems and variety of transmission system and terminal apparatus presently functioning. The Indian telecommunication is vast and currently build around nearly 15,000 exchanges for long distance services. Currently 28 percent of these exchanges are digital and only digital switches will be installed in the future. However, digitalization of the networks is bringing about significant improvement in the quality of service and availability of enhanced features to the customer. These include not only access to wide range

of services, but also features such as abbreviated dialing, call forwarding, three-way calling, itemized billing and dynamic control of Direct Distance Dialing (DDD). The transmission network also is undergoing vast changes toward realization of Integrated Digital Network (IDN). Coaxial cables and analog microwave are being increasingly replaced by optical fiber on trunk routes to form the backbone network.

5.2.1 Domestic and Overseas Service

Today in India telephone or voice services constitute over 70 percent of the telecom network by investment and revenue. Major non-voice services include advance data communication services, telex, and telegraph. These services share a common long distance transmission media with voice services. Telex is an important messaging service and is widely used by the business community on a very large scale. The telex network has over 50,000 customers and carries over 40 crore (4 billion) messages. For carrying urgent messages to rural and remote areas, telegraph networks services play a vital role. A number of new services, including e-mail, video text, cellular telephones, paging, video conferencing and audio conferencing are being demanded by business, academic

customers. Some of these services are being offered from metropolitan centers and will become widespread throughout the country very soon.

5.2.2 Today's Environment

The telecommunication networks has an equipped capacity of 5,932 million working connections as of September 1991. At the same time, the registered waiting list stood at 2,223 million. However, registered waiting list is not the true indication of the demand for telephone. The present status is given in **Table 5.3** [8].

Objectives	Parameters	Status (March 1991)
Availability	Switching capacity	5.825 mn
	Working connection	5.075 mn
	Telephone Exchanges	15,091
	Waiting list	1.961 mn
	Telephone density	6.0 per 100
Accessibility	Urban PCOs	70,770
	Rural LDPTs	26,952
	STD PCOs	11,175
Connectivity	DHO to state capacity	417
	Digital connectivity	18%
Business Network	Satellite Based Remote Commissioned July 1991	
	Area Business Message Network	
	Packet Switched Data	
	Network (I-Net) Commissioned Oct 1991	
Messaging Facility	Telegraph Offices	38,520
	Bureau Fax (Stations)	315

Table 5. 3: Present Status of Telecommunication Networks in India

At present Indian telecommunication owns few satellites which operate in the C-band (4 to 6 GHz) using analog FDM (frequency division multiplexing) /FMN/FDMA (frequency division multiple access) and SPCA techniques. This was ideal for a few hundred earth stations in the country. But unfortunately in these bands, the parking space in the sky is now too crowded for new geostationary satellites so instead of two or three huge multipurpose satellites, it is better to have 10 to 15 small single purpose satellites (micro satellites with Ka-band around 20-3 GHz) [8]. Recently VSAT (Very Small Aperture) with SSMA techniques have been introduced with low bit rate data transmission.

In the end user equipment major developments are taking place. In recent years a significant change that took place in India was opening up of the PBAX market. Instead of PABX being provided by DOT on MTNL (Mahanagar Telephone Nigam Limited), these have become end-user equipment and are offered by third parties. Currently the urban areas are well covered but it will not be anywhere near saturation point in the near future.

5.3 Technology

Throughout the world, telecommunication has changed significantly in the last two decades, not only in technology but also in services offered. India has kept pace with some of these fundamental developments to take advantage of new technology and associated benefits.

In switching there is an age old electromechanical technology in use called Strowger. In 1950s there was a new technology developed called crossbar. The most recent technology that got developed abroad in the 1970s but came in India in the 1980s is the stored program controlled (SPC) exchanges. There are two variations in this: SPC Analog switches and SPC Digital switches. In India there mostly are Analog switches. The most common one which the industry manufactures through the CIT-Alcatel collaboration is called E-10-B. It is a stored program control but the switching parts are analog. E-10-B exchanges are able to deliver a plethora of services which the earlier exchanges could not deliver. Conditions in India being such that we still continue to use pulse phones. One graduation that has taken place is that rotary dial phones are making way for push button phones. Recently Alcatel, a French telecommunication company signed a technology transfer deal with ITI to replace E-10-B exchange in metros

to OCB-283 level to enhance the ISDN (Integrated Services Digital Network) in India.

5.4 Transmission

In transmission, India is making good advances, passing from analog to digital transmission. India has adopted the 32 channel European, CCITT standard. In India there is a very large mix of transmission technologies. Transmission at the inter-city level occurs through coaxial, microwave (2,4,6, and 11GHz band) and satellite. Microwave is graduating to digital microwave, coaxial has graduated and satellite is also digital. However, UHF system (300 MHz to 1 GHz) are used for connecting rural exchange to district head quarters. Presently 18% of the long distance transmission networks are analog, but all new transmission equipment introduced in the networks from 1991 are all digital. Optical fiber is now serving as main transmission technology in all terrestrial routes, linking all 321 secondary switching areas. Current fiber line equipment operates at 34 and 140 Mbps.

5.5 Development of Data Communications in India

For more than 30 years the developed world has been going ahead with data transmission networks and created large databases, nationally and internationally, but India woke up to such a necessity less than a decade ago.

5.5.1 An Historic Perspective

The efforts in India regarding development of online networks have not been very satisfactory so far. By the time India finally decided to set up data transmission networks, including an international switch, the developed world had already undergone a revolution in connected technology with advents of methods of transmits, data of high speed in bulk. Though the plan for developing national network was planned earlier, it came as a reality only in 1980s.

India set up NICNET (National Informatics Centre, Planning Commission of India), INET (National Packet switching Network, I-Net phase I was commissioned by DOT, Department of Telecommunication), RABMN (Remote Area Business Message Network) etc, which are the most sophisticated computer

based networks. The first attempt towards networking was made by the CSIR in 1975 by considering the setting up of CSIR-INFONET, which was visualized to be functional at three levels of nodes- the apex node, the regional node, the laboratory node. But so far the conceptual framework for the CSIR network has not been put into practice. The second attempt for setting up a National Information Network was made with the assistance of UNESCO experts who laid the foundation of NISSAT. NISSAT organized online remote database search in Bombay in 1976, and in New Delhi and Bangalore in 1981 which was a milestone in the history of information technology of India. At present there are at least twenty networks of various types functioning in India. The existing networks in India are categorized in **Table 5.4**.

Wide Area Network (WAN)	Metropolitan Area Network (MAN)
NICNET, INDONET, I-NET, INFLIBNET, ERNET, VIDYANET, BTNET, SAILNET, BANKNET, NETWORK INDIAN AIRLINES, RAILNET, SIRNET, SBINET, OILCOMNET, RABMN	CALIBNET, DELNET

Table 5. 4: Categorization of Existing Networks in India

The emerging information networking services, globally, demand a large bandwidth from telecommunication networks for large volumes of data to be

transmitted within a very short interval. This makes increasing the bandwidth critical in developing countries (DCs). We can make better use of an existing leased line. Data compression can be a savior as it minimizes the need for expansion of the bandwidth.

Chapter 6

6. DATA COMPRESSION

One of the hottest imaging technologies is compression. In the early stages, compression was limited to JPEG, GIF and Group 3 fax. Compression technologies have multiplied. Now there are also Group 4, PNG, wavelet and fractal compression. Compression is applied to still images, audio and video. Three reasons to compress: save storage space, conserve bandwidth, and speed up application software.

Data compression is often referred to as source coding, where coding is a general term encompassing any special representation of data that satisfies a given need. Data compression may be viewed as a branch of information theory in

which the primary objective is to minimize the amount of data to be transmitted [40].

Data compression has important application in the areas of data transmission and data storage. Many data processing applications require storage of large volumes of data, and the number of such applications is constantly increasing as the use of computers extends to new disciplines. At the same time, the proliferation of computer communication networks is resulting in massive transfer of data over communication links. Compressing data to be stored or transmitted reduces storage and/or communication costs. When the amount of data to be transmitted is reduced, the effect is that of increasing the capacity of communication channel. Similarly, compressing a file to half of its original size is equivalent to doubling the capacity of the storage medium. It may then become feasible to store the data at a higher, thus faster, level of the storage, hierarchy and reduce the load on the input/output channels of the computer system.

Image compression and decompression operations are used to reduce the data-content size of a digital image. The goal of these operations is to represent an image, with some required quality level, in a more compact form. Image compression operations seek to extract essential information from an image so that the image can be accurately reconstructed. Nonessential information is discarded.

Two most common applications: **(1) Data storage:** A body of data is compressed before it is stored on some digital storage device, for example, a computer disk or tape. This process allows more data to be placed on a given device. When data is retrieved from the device, it is decompressed; and **(2) Data communications:** Communication lines that are commonly used to transmit digital data include cables between a computer and storage devices, phone lines, and satellite channels. A sender can compress data before transmitting it and the receiver can decompress the data after receiving it.

Image compression is necessary for both electronic storage and transport. Both require more storage space and bandwidth. Image storage refers to the electronic storage of data, for example, on magnetic or other permanent media. Image transport refers to the electronic transfer of data over a data link.

Image compression techniques can be grouped into two major categories: **lossless and lossy methods shown in Figure 6.1 [22].**

Image Compression	
Lossless Compression	Lossy Compression

Figure 6. 1: Image Compression Methods

6.1 Lossless Compression

Lossless image compression is devoid of losses, or degradations, to the data. When the compressed data is decompressed the original form is recovered. Many different techniques come directly from the digital data-compression world and have been adapted for use with digital image data.

Lossless data compression is the process of transforming a body of data to a smaller one, from which it is possible to recover exactly the original data at some point later in time. Lossless compression is appropriate for textual data (book, database information, numerical data, e-mail, etc.) where it may be unacceptable to lose even a single bit of information.

6.1.1 Run-Length Coding

Run-length coding image compression takes advantage of the fact that several nearby pixels in an image will, statistically, tend to have the same

brightness value. This form of redundancy can be reduced by grouping pixels of identical brightness into single codes.

The original image is evaluated by starting at the first pixel in the upper left corner (first line). The scheme determines how many following pixels have the same brightness value by looking at the first pixel and its following neighbors across the line. If the neighbors also have the same value as the pixel, all of them are represented by a new code with two values: brightness value followed by the number of pixels (run-length) that have the same value. The process continues until the entire image is run-length coded [56]. This is depicted in **Figure 6.2**.

56 5					58 1		59 4		60 2		
56	56	56	56	56	58	59	59	59	59	60	60

Figure 6. 2: Run-Length Coding

A run-length compressed image is decompressed by expanding each code in the compressed image. The brightness of each code is replicated across the decompressed image line for the number of pixels indicated by the adjoining length value.

Run-length coding is particularly prone to data explosion problems. For example, with noisy images the brightness in an image change between every

pixel in the image. Each pixel of the image will be a distinct run, and will therefore be represented by its own run-length code. Then each 1-byte pixel in the original image will be represented by 2-byte run-lengths in compressed image, a brightness value and a run-length. The compressed image will be twice as large as the original.

6.1.2 Huffman Coding

Huffman coding converts the pixel brightness values in the original image to a new variable-length codes, based on their frequency of occurrence in the image.

Huffman coding is a statistical data-compression technique. This technique will reduce the average code length used to represent the symbols of data-brightness values. The Huffman coding ensures that the longest codes get assigned to the least frequent brightness and vice versa.

The brightness values are first listed in descending order of frequency of occurrence. The two at the bottom of the list (least frequent) are paired together into a node with a joint probability- the probabilities are combined. This new

node is labeled 0 and 1. The next two lowest frequencies of occurrences are determined and paired. The new node gets the joint probability and is labeled 0 and 1. This continues until all brightness are paired. The result is a binary tree as shown in **Figure 6.3**. The base of the tree is root node and the point at which a branch divides is branch node. The termination point of a branch is leaf node to which the symbols being encoded are assigned. The tree indicates the new Huffman code for each brightness in the image.

Example:

A string of characters to be transmitted is 's₁ through s₈'. The relative frequency of each character is as follows:

$$s_1, s_2 = 0.25, \quad s_3, s_4 = 0.14, \quad s_5, s_6, s_7, s_8 = 0.055$$

According to Shannon's formula, the minimum average number of bits/character (entropy H [22]) is

$$H = - \sum_{i=1}^8 p(s_i) \log_2 p(s_i) \text{ bits per codeword}$$

$$\begin{aligned} H &= - (2 * 0.25 \log_2 0.25) + 2 * 0.14 \log_2 0.14 + 4 * 0.055 \log_2 0.055) \\ &= 1 + 0.794 + 0.921 = 2.715 \text{ bits/codeword} \end{aligned}$$

Codeword Generation:

Huffman Code generation is presented in **Table 6.1**.

s_1 0.25	s_1 0.25	s_1 0.25	s_1 0.25	s_{34} 0.28	s_{25678} 0.47	s_{134} 0.53 (1)	root $s_{12345678}$ 1.00
s_2 0.25	s_2 0.25	s_2 0.25	s_2 0.25	s_1 0.25	s_{34} 0.28 (1)	s_{25678} 0.47 (0)	
s_3 0.14	s_3 0.14	s_3 0.14	s_{5678} 0.22	s_2 0.25 (1)	s_1 0.25 (0)		
s_4 0.14	s_4 0.14	s_4 0.14	s_3 0.14 (1)	s_{5678} 0.22 (0)			
s_5 0.14	s_{78} 0.11	s_{78} 0.11 (1)	s_4 0.14 (0)				
s_6 0.055	s_5 0.14 (1)	s_{56} 0.11 (0)					
s_7 0.055 (1)	s_6 0.055 (0)						
s_8 0.055 (0)							

Table 6. 1: Huffman Code Generation

Weight order = 0.055 0.055 0.055 0.055 0.11 0.11 0.14 0.14 0.22
0.25 0.28 0.47 0.53

Huffman Code:

Symbol	s_1	s_2	s_3	s_4	s_5	s_6	s_7	s_8
$p(s_i)$	0.25	0.25	0.14	0.14	0.055	0.055	0.055	0.055
Codewords	10	01	111	110	0001	0000	0011	0010
#bits	2	2	3	3	4	4	4	4

Average number of bits/codeword using Huffman coding is:

$2(2 * 0.25) + 2(3 * 0.14) + 4(4 * 0.055) = 2.72$ bits/codeword that is, 99.8% of Shannon's value.

Huffman Code Tree (Binary Tree):

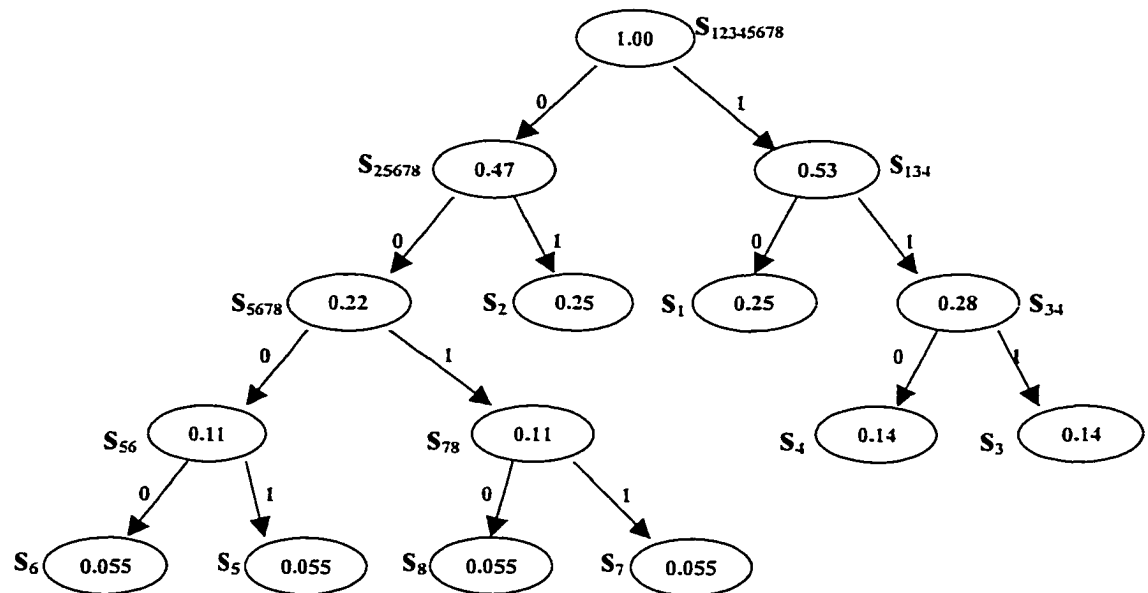


Figure 6. 3: Huffman Binary Tree

Huffman image decompression reverses the compression process by substituting the original fixed length 1-byte-long brightness values for the variable-length Huffman-coded values restoring the original image. Huffman image compression will generally provide compression ratios around 1.5: 1 to 2:1.

6.2 Lossless Predictive Coding

The fundamental form of predictive image coding is differential pulse code modulation (DPCM). The DPCM system was developed at Bell Laboratories a few years after World War II [14]. It is most popular as a speech-encoding system and is widely used in telephone communication.

The difference between pixel brightness is the quantity that is coded. This scheme operates on the entire image, pixel by pixel. Consider the first pixel in the upper left corner of the image. The first pixel remains unchanged. It is coded exactly as its original brightness. The process moves to the second pixel in the line. The preceding pixel's brightness value is subtracted from the current pixel's brightness. The result of the subtraction is the new coded value for the second pixel value in the image. This process repeats across the line. At the start of the

next line, the process begins over. This process continues until the entire image is DPCM coded.

Example:

The five original pixel brightness are compressed from 5x8-bit values (40 bits) to 5x6-bit difference values (30 bits).

$$\text{Compression Ratio} = 8/6 = 1.333:1$$

If all difference values were under 16, only 4-bit numbers would be necessary yielding 2:1 compression ratio.

The DPCM operation works with the assumption that neighboring pixels will be similar or highly correlated. As a result, their difference will usually be very small values. Looking at the values in the above example, none are greater than 31 or less than -32. These difference values can be coded using 6-bit numbers instead of 8-bit numbers, yielding a compression ratio of $8/6 = 1.333:1$. If all difference values were under 16, only 4-bit numbers would be necessary, yielding a 2:1 compression ratio.

The DPCM compression technique relies on the statistics of the pixel brightness in the image to be 'normal' – images do not have inordinate number of large brightness swings between adjoining pixels.

6.3 Lossless Block Coding

Pixel block compression techniques code a group of pixels at a time, searching for image redundancy in the form of repeated patterns. Block coding is based on the statistical likelihood that brightness pattern will probably repeat throughout an image. A block of pixels can be defined in the form of a 2-D array of pixels or a 1-D line of pixels.

Block coding is implemented using a table different pixel brightness patterns, called a codebook. The original image is evaluated pixel by pixel across each line beginning in the upper left of the image. The evaluation searches for brightness patterns that match one of the patterns in the codebook. When a matching pattern is found, the block of pixel brightness is represented, in the compressed image, as a code pointing to the entry in the codebook. The codebook size is kept small enough to ensure that the number of code bits necessary to point a code book entry is smaller than the average number of bits replaced in the original image's pattern.

Lossless pixel block-coding techniques can generally provide between 2:1 and 3:1 compression ratios. The computational requirements for block coding, however, are higher than for other techniques like run-length and DPCM coding. A common lossless block-coding method for images is the Lempel-Ziv-Welch (LZW) scheme [Baxes, Gregory A (1994), Digital Image Processing, Wiley].

6.4 Lossy Image Compression

Lossy data compression is the process of transforming a body of data to a smaller body from which an approximation to the original can be constructed. An important application of lossy compression is the compression of digital sampled analog data (DSAD), such as speech, music, black and white or color images, video, and satellite data.

Lossy image compression schemes are not inferior to lossless schemes. They are far superior for certain applications. All forms of lossy image compression involve discarding image data. The greatest advantage to lossy compression scheme is their ability to compress an image to a much smaller data form than lossless schemes. Lossy schemes can push the compression ratios to 10:1 without visually noticeable degradations [22]. They can reach 100:1 and

greater compression ratios with some noticeable degradations. For example, television broadcasters do not need to maintain absolute image data integrity when transporting video programming to viewers. Certain degradations introduced by lossy compression will not degrade the visual quality beyond its acceptance limits by the end-user, viewing consumer. Lossless removes redundancy and lossy removes irrelevancy.

6.5 Lossy Predictive Coding

Lossy DPCM compression and decompression schemes are identical to the lossless form of DPCM, except that the code overload condition is not handled as a special case. The maximum difference code is used for successive pixels until the coder catches up with the actual brightness value of the original image.

6.6 Lossy Block Coding

Lossy pixel block-coding schemes are similar to the lossless forms, except that blocks are not perfectly represented by the codebook blocks. When a compressed image is decompressed, some errors will occur. Lossy block-coding

techniques do not attempt to find identical pixel block matches in the codebook. They look for good representative matches, where the error between the block being coded and the codebook block is minimized. These techniques measure the error between pixel blocks and the codebook blocks using a measure that is based on the intended use of the decompressed image [59].

$$\text{Compression Ratio} = \frac{\text{Length of original data } D}{\text{Length of compressed data } \delta(D)}$$

The higher the compression ratio, the more effective the compression technique employed. Suppose storing an image made up a square array of 256x256 pixels requires 65,536 bytes. The image is compressed and the compressed version requires 16,384 bytes. The compression ratio is 4:1.

$$\begin{aligned} \text{Figure of merit [59]} &= \frac{\text{Length of compressed data } \delta(D)}{\text{Length of original data } D} \\ &= 1/\text{Compression Ratio} < 1 \text{ for compression} \\ &\quad \text{to be effective} \end{aligned}$$

$$\text{Original data } D \leftrightarrow \text{Data Compression} \leftrightarrow \text{Compressed data } \delta(D) \text{ [59]}$$

6.7 Categories of Image Compression Systems

The end result of the image scanning process is a data file containing the digitized imaged information. Most documents contain large regions of continuous dark or light objects. Significant savings in storage as well as network transmission time can be achieved by compressing the image file.

Compression codes are sophisticated algorithms that act on the digitized image file to reduce redundancy in the bit stream. The compression techniques are a major factor in the traffic engineering algorithms needed to design an appropriate network architecture for transmitting the image files among distributed peripherals.

Without image compression and decompression, most imaging systems would operate much more slowly and would require substantially higher bandwidth as well as more storage capacity. For example, a typical 8.5x11 inch document scanned at 300 dpi without gray-scale may be compressed by a ratio of 15:1 from a file of slightly over a megabyte down to a file of about 75kb. This compressed file is then transmitted and stored on the optical disk, thereby

increasing the effective storage capacity of the disk by a ratio of 15:1 and reducing the time for transmitting the document by a similar amount.

Sequential Coding: Encodes the image in a single scan. The decoder necessarily follows the same order in decoding the data; **Progressive Coding:** Encodes the image in two or more scans, first enclosing an approximation of the entire image and then encoding finer details for the whole image with each succeeding scan. The decoder follows the same order in decoding, first reconstructing the approximation of the image, and then adding finer detail with each succeeding scan; and **Hierarchical or Pyramidal Coding:** Uses a set of successively smaller images that are created by “downsampling” the preceding large image in the set. Then, starting with the smallest image in the set, the image set is coded with increasing resolution. After the first stage, each lower resolution image is scaled up to the next resolution and used as a prediction for the following state.

Motion Compression

Motion compression is a particular case of image compression where image redundancy between sequential image frames is removed. It is applicable only for a sequence of related images, like those encountered in motion picture

and video production, medical image sequence, and teleconferencing applications.

This form of image compression is referred to as interframe coding.

6.8 Standard Image Compression Schemes

A number of standardized image compression techniques have evolved to support the requirements of different industries. A good example includes the compression standards created for facsimile (fax) machine to transmit a document's image.

CCITT Group 3/4 and Joint Bi-Level Image Experts (JBIG) Group (JBIG)

The most widely used standardized image compression schemes were established by the International Telegraph and Telephone Consultative Committee (CCITT) for the transmission of binary images. These compression standards are lossless and are used by all common fax machines to transmit text and line-drawing document images.

The Group 3 and 4 compression standards are based on run-length and Huffman coding techniques. First, the runs of black and white pixels are compressed to run-lengths. The run-lengths are then Huffman coded. The Group 4 standard is simplified version of the Group 3 scheme that provides improved compression ratios. The data explosion problem occurs in Group 3 and 4 schemes.

The Joint Bi-Level Image Expert (JBIG) improves the compression ratios of binary images. The JBIG standard was established by a joint effort of ISO (International Standard Organization), IEC (International Electrotechnical Commission), and CCITT. JBIG-compressed images progressively decompress so that intermediate, lower-resolution images can be displayed without decompressing the entire compressed image data file [56].

Joint Photographic Experts Group (JPEG)

The JPEG standard, established jointly by the ISO/IEC and CCITT organizations, is one of the most important image data compression standards of the 1990s. The JPEG image data compression standard handles gray-scale and color images of varying resolution and size. It is intended to support many industries that need to transport and archive images. The JPEG standard is used

in graphics arts, desktop publishing, medical imaging, color fax, and countless other applications. This standard is commonly used in a lossy mode. However, there is also lossless mode with reduced compression performance [56].

The JPEG image compression scheme uses several cascaded compression modes. First, an image is transformed to the frequency domain using the discrete cosine transform (DCT). Then, resulting smaller-valued frequency components are discarded, leaving only the larger-valued components. The remaining frequency components are DPCM coded and then Huffman coded. The lossless version of JPEG uses only the DPCM and Huffman coding portions of the standard.

Moving Picture Experts Group (MPEG)

The **MPEG** standard, established jointly by the **ISO/IEC** and CCITT organizations, is intended for the mass distribution of motion video sequences, such as motion pictures and television programming. The MPEG scheme is compression-asymmetric- the compression algorithm requires more processing time and computing power than the decompression algorithm. This way, MPEG minimizes decompression hardware by placing a heavier burden on the compression side.

MPEG is divided into different efforts: **MPEG1**, **MPEG2** and **MPEG4**. MPEG1 (usually for storage) is intended for low-resolution image sequences like those composed of 320 pixel x 240 line images. It performs well using 1.5 Mbits/second transport links. MPEG2 (transport) is intended for higher resolution applications, such as full-resolution, 640 pixel x 480 line video image sequences [56]. The MPEG2 standard requires transport links between 4 and 10 Mbits/second, depending on the required decompressed image quality.

MPEG motion image compression standard also uses discrete cosine transform (DCT) and Huffman coding techniques. Additionally, interframe coding methods are used to make MPEG's compression performance better than schemes intended solely for still-image compression.

6.9 Image Synthesis

Image synthesis operations create images from other images or non-image data. The goal of these operations is to synthesize digitally, through computed methods, a resulting image that did not exist before.

Image synthesis operations see daily use in both the medical diagnostic imaging and computer-aided design fields. There have been many advancements in medical diagnostic imaging since the creation of computed tomography (CT) [26]. These techniques can create physically unobtainable cross-sectional images of the human body. Communication of diagnostic information is a high priority in applications of the intelligent network for medical applications. Administration is also a major application where emergency communications, diagnostic dialogs and billing can be coordinated among cooperating institutions over a multimedia network.

A telemedicine network is a vehicle for providing health care services to remotely located and underprivileged sections of society. Many physicians use telecommunication links for central analysis of electrocardiogram readings. Having a medical alert button or life support system in a home is a growing reality [1]. So routine medical care is amenable to computer and network processing. Machines can enhance the productivity and efficiency of medical environments. The medical industry can benefit from the availability of new services in a public telecommunication network, such as image transport at variable bit rates, image processing, image consultation, and imaging network management.

Chapter 7

7. DATA COMPRESSION APPLICATIONS TO NETWORKS

Two most common applications: (1) **Data storage:** A body of data is compressed before it is stored on some digital storage device, for example, a computer disk or tape. This process allows more data to be placed on a given device. When data is retrieved from the device, it is decompressed; and (2) **Data communications:** Communication lines that are commonly used to transmit digital data include cables between a computer and storage devices, phone lines, and satellite channels. A sender can compress data before transmitting it and the receiver can decompress the data after receiving it.

Image compression is necessary for both electronic image storage or transport. Both are very sensitive to the amount of data in an image. Image storage refers to the electronic storage of an image's data, for example, on

magnetic or other permanent media. Image transport refers to the electronic transfer of an image's data over a data link.

7.1. Applications

There are a host of known applications for data compression techniques and systems, and new applications are being proposed each day, creating a demand for modifications of existing compression methods or entirely new approaches. Existing applications of compression methods include facsimile, voice mail, telephony, digital cellular mobile radio, personal communications systems, CD-quality audio, still-image archival, videoconferencing, and video and movie distribution, just to name a few. These applications have led to a plethora of compression standards – agreed-upon techniques and systems-level specifications that are adopted by interested industry representatives to allow the manufacture of compatible equipment.

Compression is applied to text, audio, images, video, and HDTV. Three reasons to compress are to: save storage space, conserve bandwidth, and speed up application software. Data compression has important application in the areas of data transmission and data storage. Many data processing applications require

storage of large volumes of data, and the number of such applications is constantly increasing as the use of computers extends to new disciplines. At the same time, the proliferation of computer communication networks is resulting in massive transfer of data over communication links. Compressing data to be stored or transmitted reduces storage and/or communication costs. When the amount of data to be transmitted is reduced, the effect is that of increasing the capacity of communication channel. Similarly, compressing a file to half of its original size is equivalent to doubling the capacity of the storage medium. It may then become feasible to store the data at a higher, thus faster, level of the storage hierarchy and reduce the load on the input/output channels of the computer system.

7.1.1 E-mail

Along with the Web, electronic mail is one of the most popular Internet applications. E-mail is asynchronous- people send and read messages when it is convenient for them, without having to coordinate with other peoples' schedules. E-mail requires fully reliable data transfer, that is, no data loss. E-mail can make use of as much or as little bandwidth as happens to be available. Development of data compression algorithms has come close on the heels of the popularity of e-

mail application. Two compression algorithms, Huffman Coding and Lempel-Ziv-Welch (LZW) are used in this study to simulate e-mail application (Ch 9).

7.1.2 Audio

Sound is another rich arena for compression. Music is a commodity sold on digital plastic disks by the recording industry. CD-quality audio data is very suitable for compression. The audio signal for each stereo channel is sampled at 44.1kHz, and each sample is represented by 16 bits. This means that the amount of data stored on one CD is enormous. If we want to transmit this data, the amount of channel capacity required would be significant. Compression is definitely useful in this case. MPEG-1 with two simulation runs is used in this study to simulate audio application (Ch 9). Audio compression promises to be one of the most important areas of research in the early part of the 21st century because it will form the basis for how major industries blend with the Internet.

7.1.3 Image

Image is another class of data that is very suitable for compression. One frame of a television signal consists of 480 lines with 640 pixels/line. This frame will produce 4800 8 by 8 pixel blocks for the luminance signal for one frame. Compression is definitely useful in this case. MPEG-1 has been chosen in this study to simulate image application (Ch 9). MPEG-1 standard provides a lossy compression algorithm that can be adjusted to balance image quality versus file size. Clearly, in the case of images, we either make do with lower resolution and/or lower quality images or we procure higher speed communications. Compression would reduce the image size by the compression ratio while preserving the integrity of the image. Lossy compression technique removes redundant information and thus improves the efficiency/cost of transmission.

7.1.4 Video

Digital images are another area for compression algorithms. Moving pictures are a popular medium for expression. The amount of information is staggering even after the relentless success of the magnetic storage industry.

Good video signals change the image 20 to 30 times per second, which means that even a minute of video is made up of 1200 still images. There is redundant information in the images that can be excluded to compress the file. More importantly, the eye is often quite forgiving, so the reconstruction does not need to be perfect. The video algorithms are all lossy, and this allows them to save substantially more storage space or bandwidth.

There are a number of different video compression standards. Most of them are variations of the work of two different committees: The International Telephony Union's Specialist Group on Coding for Visual Telephony and the International Standards Organization's motion picture expert group, known as MPEG. The most popular standards are the ISO's MPEG-2 and the ITU's H.261. MPEG-2 has been chosen in this study to simulate video application (Ch 9).

7.1.5 HDTV

HDTV (High-Definition TV) is not the same as DTV (digital TV). The high-definition was initially achieved by an analog signal, but now is standardized as a digital format. Initially HDTV was IDTV (I was for improved), but the 1000-line threshold of vertical resolution, as high as 1280 lines, produced twice

the resolution. HDTV standards began evolving in the early 1980s and were finalized in the early 1990s. Similar in nature to the video standards HDTV standards increase the number of pixels per line and the number of lines in the image frame, 1920 and 1080, respectively. HDTV standards put the screen width at 16:9 aspect ratio to emulate the wider motion picture screens. HDTV is the maximum resolution and the highest quality of video and sound. The huge bit rate prevents HDTV transmission unless signals are squeezed into a smaller space. In the nineties, advances in digital compression allowed the huge bit stream necessary for HDTV to fit into the 19 Mbps range possible within the confines of a 6-MHz channel. Compressed digital HDTV requires the full bandwidth of a television channel (6 MHz) to deliver the maximum pixel count of 2,073,600 (1920 x 1080).

Although it will take years for the commercial broadcast television industry to fully exploit HDTV, industry is already making widespread use of HDTV for applications requiring higher spatial resolution. The equipment costs, however, are still high. As the commercial television broadcaster and consumer demands increase over time, HDTV costs will drop.

MPEG-2 was originally designed for compressing broadcast quality video into 4 to 6 Mbps, so it could fit in a NTSC (National Television Standard

Committee) or PAL (Phase Alternating Line) broadcast channel. Later, MPEG-2 was expanded to support higher resolution, including HDTV. Thus, MPEG-2 has been chosen in this study to simulate HDTV application (Ch 9).

Text application uses small message size while multimedia applications demand large size. So the message size varies in this study from 100 bytes for text to 93.225MB for video applications. Bandwidth is determined according to the type of application. Smaller bandwidth for text and larger bandwidth for multimedia applications is required (Ch 9).

An important facet of the information explosion is the computer access to libraries and databases. The amount of storage (and bandwidth) generated by the Internet has yet to be estimated and will create countless new storage management and application developments. Beyond the Internet, entertainment may be the largest single driver of storage demand on a global basis as we demonstrate an endless appetite for any form of digital entertainment in the home, away from home and even at work. Knowledge is the full range of facts, information, and principles accumulated by mankind over time. The millennium has promised to be the information age, not the data age, however. Even with the growth of data expanding at unprecedented levels, the path to convert data to information and

then to knowledge, promise to make any historical growth rates for pure data storage seem insignificant.

Typical compression rates (see Ch.6, Sec.6.6 for definition) used for simulation for different applications are shown in **Table 7.1** below.

Algorithms	Huffman Coding	LZW	MPEG-1	MPEG-2
Applications				
E-mail	compression rate: 40%	compression rate: 60%		
Audio Simulation Run1			compression rate: 50%	
Audio Simulation Run 2			compression rate: 50%	
Image			compression rate: 30%	
Video				compression rate: 30%
HDTV				compression rate: 30%

Table 7. 1: Applications, Algorithms and Compression Rates

Chapter 8

8. SIMULATION

Network simulation is a valuable tool because it provides a means for modeling a network to determine its performance characteristics. Performance assessment includes accuracy, reliability, queuing, traffic, versatility, etc. Network simulation provides a means of testing proposed changes prior to placing them into effect, performing what-if analysis concerning the reliability of key components in a network and the effects of losing a component, planning for future growth, and planning the initial design of a proposed network. The cost associated with the building and operating of a network make simulation a viable option in making choices in the building, modification, and performance analysis of a network.

8.1. Simulation Tools

Systems designers and network planners are increasingly looking for support tools to help them decide on the best design. They use the results from these tools to make their cases to clients and management. Companies like Compuware (formerly CACI) provide analysis tools for this purpose. COMNET III (a Compuware product), used in this study is a graphical software package. It is designed to accurately estimate the overall performance characteristics of computer and communication networks. A network description is created graphically using a window interface. The application was formulated primarily for the modeling of both Wide Area Networks (WANs) and Local Area Networks (LANs).

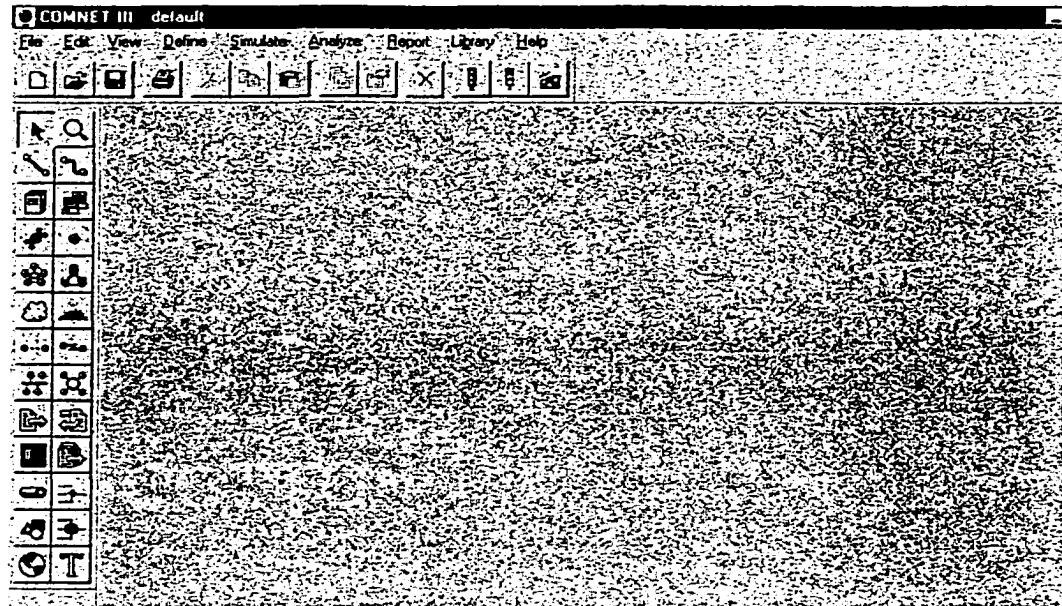


Figure 8. 1: View of COMNET III Graphical User Interface

Figure 8.1 displays the view of COMNET III. The standard menu format of most windowed applications appears across the top of the window. Clicking on a menu activates pull-down menus. The toolbar on the left-hand side of the screen allows for a simple method of creating objects in a model. The area to the right of the toolbar is the work area where models are built.

The topology of a network is defined by three basic components: (a) **nodes** to represent hardware (computers or switches), (b) **links** that carry traffic between nodes, and (c) **arcs** to associate nodes to links. In addition to the basic nodes and links, we have Subnet and Transit Net for modeling independent

routing domains and hierarchical topology, and the WAN Cloud for modeling wide-area network services.

8.1.1. Nodes

COMNET III has three basic node types: (1) **Processing Node**, (2) **Computer Group Node**, and (3) **Network Device Node**. A Processing Node is used for the modeling of end systems such as computers, printers, facsimile machines, or any general piece of network hardware (such as pads and packet switches). A processing node acts as the origin or destination for message traffic, runs applications, and/or acts as a switching point within a network. Some of the attributes of a processing node in a model are as follows:

- Input buffers for each link connected to the node for accepting packets transmitted to the node.
- Output buffers for each link connected to the node through which the node may route packets.
- Local disk storage for modeling disk read and write processes.
- A command list that defines how application commands are to be executed on the node.

- A received message list for saving received messages used to trigger an application or traffic source.
- A list of files which reside in storage on the local disk.

A processing node contains an internal processor. The processor is used for processing of application and traffic sources, and for processing of packets which the node receives. Each node maintains a list of all applications and traffic sources that may run on that node.

Computer Group Nodes model computers. They can generate or receive traffic like processing nodes. Computer groups are restricted to modeling end systems because they can only be sources or sinks for traffic.

Network Device Nodes model hardware used for routing traffic, including routers, hubs, and switches. The network device node is similar to processing node in that it can source or sink traffic and run applications that utilize the internal processor and internal storage. However, it adds a model for an internal bus structure for moving traffic between input and output ports.

All three types of nodes, depicted in **Figure 8.2**, can be sources of traffic and resources for workload- they can accept sources of traffic and applications.

To model applications, these nodes include a command repertoire as well as a parameter set.

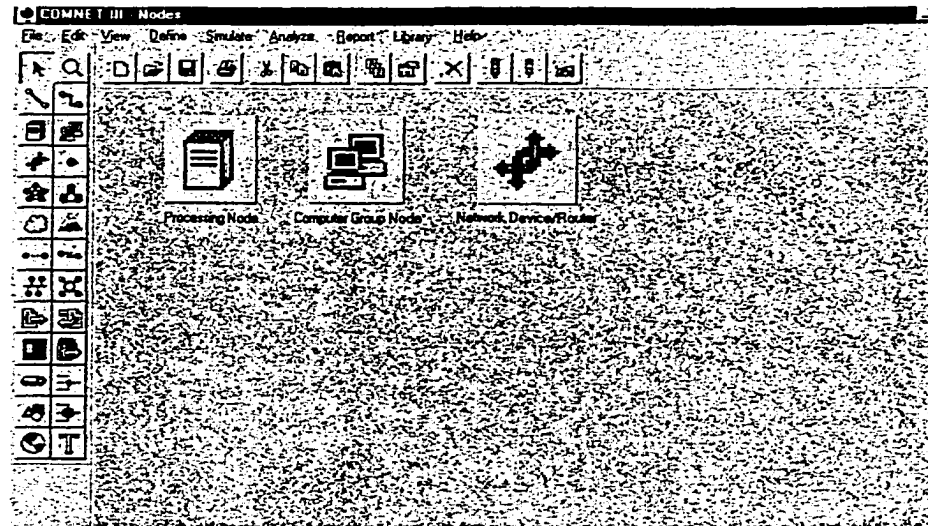


Figure 8.2: COMNET III Processing Nodes

8.1.2. Links

There are two classes of link models available in COMNET III: (1) **Point-To-Point** to represent a channel between only two nodes. These links model communication channels between any two nodes. Typically, these connections are dedicated lines such as a serial line or a dedicated line connection, between routers in wide-area or multihop networks. WAN Cloud model provides access links to the WAN service by using a variation of point-to-point link

available inside the cloud; and (2) **Multiaccess** links model various protocols for sharing a single medium with multiple nodes. They include Carrier-Sense Multiple-Access with Carrier Detection (CSMA/CD or Ethernet), CSMA/CA (collision avoidance), Aloha, Token Passing (token ring, FDDI, token bus), Polling, FDDI, Priority Token Ring, STK[®] (registered trademark of Analytical Graphics Incorporated) for modeling satellite communications, and Link Group for modeling switched ethernet, ISDN and SONET parameter sets. **Figure 8.3** displays some of these links.

WAN Clouds are used to model WAN services such as frame relay, cell relay, and packet switching. A WAN Cloud has an internal topology consisting of Access Links and Virtual Circuits (VCs). Virtual Circuits are abstract models of sets of switches and links and connect to Access Links only. Access Links are special case of a Point-To-Point link that works with the WAN Cloud.

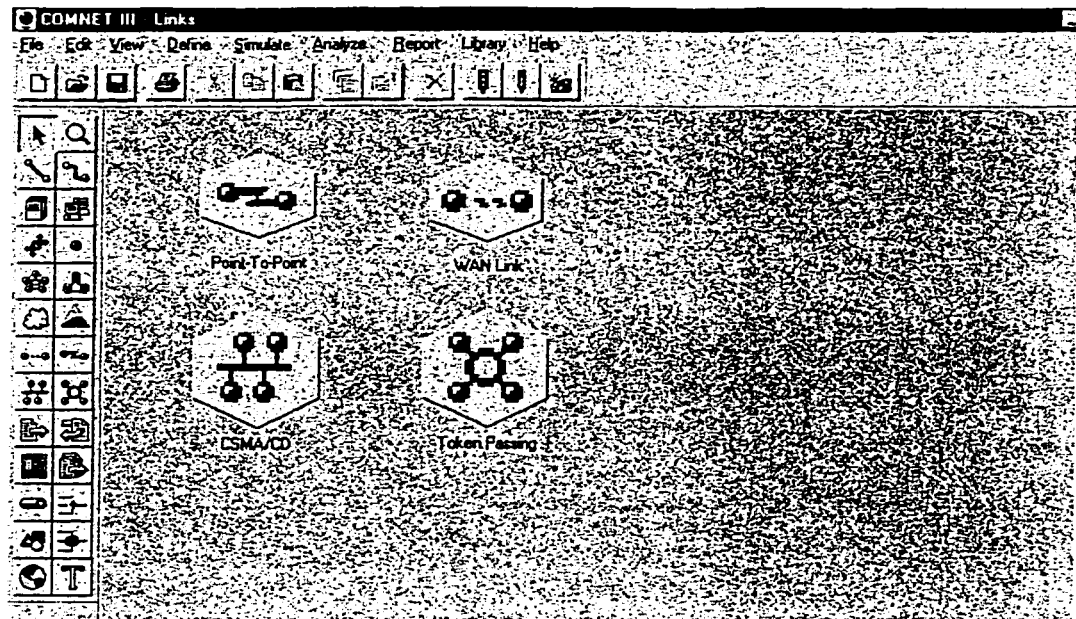


Figure 8. 3: Types of Links in COMNET III

8.1.3. Arcs

Two types of arcs are available in COMNET III: (1) **Diagonal Arc** and (2) **Horizontal/Vertical Arc**. The arcs show which nodes use links and they model the node's port connection to the link. In the backbone and subnet topologies, the arcs connecting the links to nodes represent the connections at the nodes that interface to the link media. The arcs are editable to set parameters for the port such as delay, input and output buffers and for the link (routing penalties). The two arcs are shown in **Figure 8.4**.

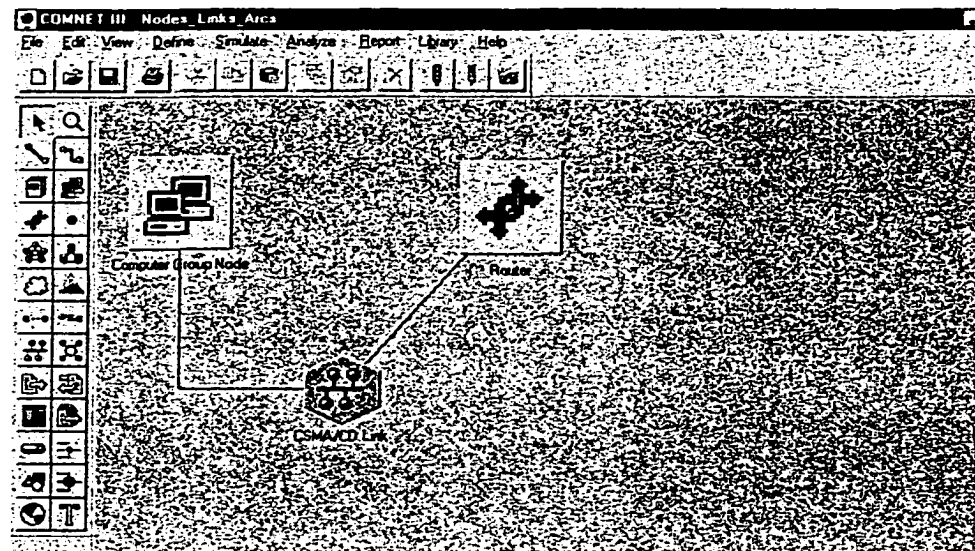


Figure 8. 4: Types of Arcs in COMNET III

Simulation Environment

The hardware and software environment are listed below:

Hardware: **PC:** Dell Dimension XPS
 Processor: Pentium III, 450MHz
 SDRAM: 512 MB
 Hard Drive: 13.6 GB (5400RPM)

Software: **Operating System:** Windows 98.

Simulation Software: COMNET III™, CACI Products Company, University
 Edition, Release 2.1.

8.2. Simulation Topology

We have two simulation models to consider each with a different set of nodes. The first simulation model contains 6 processing nodes and the second contains 12 nodes. The first and second simulation topologies are presented in section 8.2.1 and 8.2.2, respectively.

8.2.1. First Simulation Topology

First simulation model contains six processing nodes and links each (see **Figure 8.5**). The total number of pairs we can have with six nodes is $n(n-1)/2$, where n is the number of nodes. Each pair has two virtual circuits (VCs). So the model contains 15 pairs and 30 VCs. Each node represents a gateway node that connects a regional network to others.

The nodes are clustered in North America and Europe, where each cluster represents a regional network. The nodes in North America are at CUNY, New York and Princeton, New Jersey. These are major gateway nodes to other regional networks in North America and other continents. The nodes in Europe are found in Montpellier, France; Milano, Italy; Bern, Switzerland; and Bonn,

Germany. Each of these nodes is a major gateway and plays an important role in connecting other European, Middle Eastern and North African countries to the global network.

The links in the model are high bandwidth links connecting major gateways. The links from New York to Montpelier and Princeton to Bern are with high bandwidth. Other links connect gateway nodes of regional networks to each other.

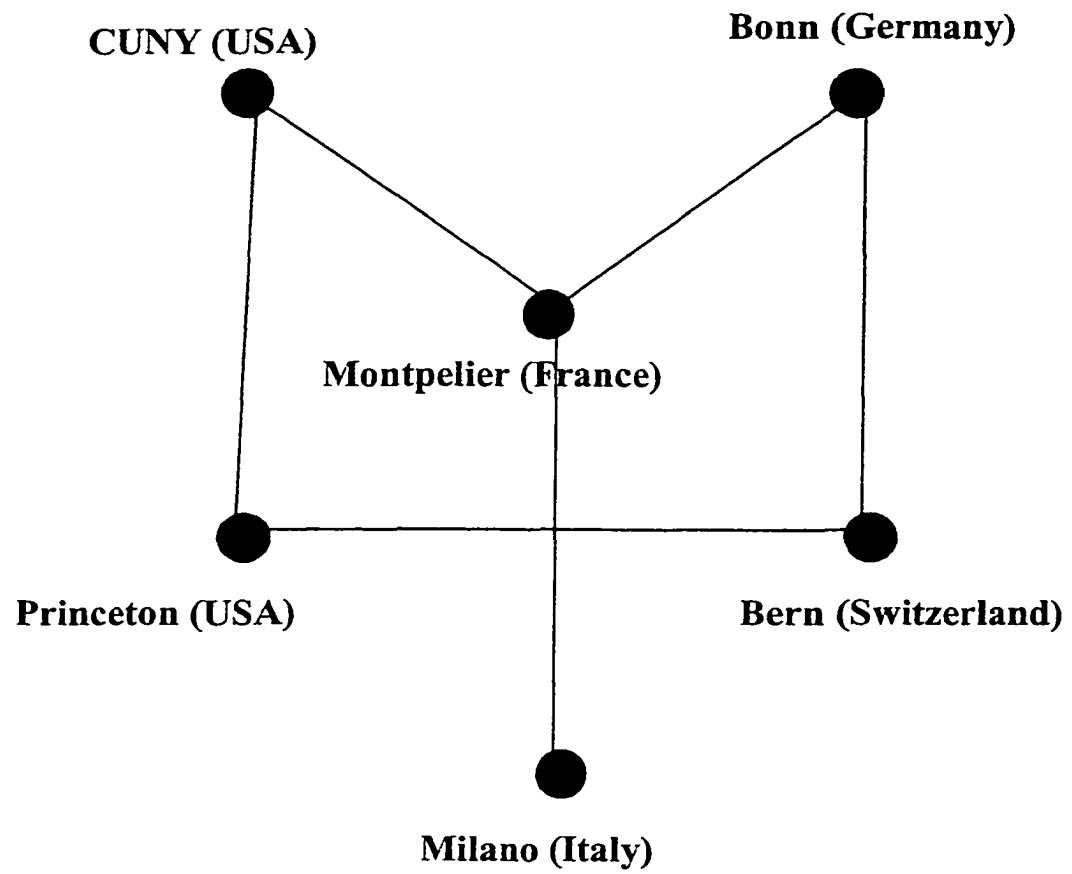


Figure 8. 5: First Simulation Model Topology

8.2.2. Second Simulation Topology

The second simulation model topology is more complex than the first one with 6 nodes. This model contains 12 processing nodes, connecting networks in the Far East, Middle East, South America and North America. There are 36 links in this model, giving more paths between any two nodes. The total number of pairs and VCs in the second model are 66 and 132, respectively. The net effect is a more robust topology and a more reliable network

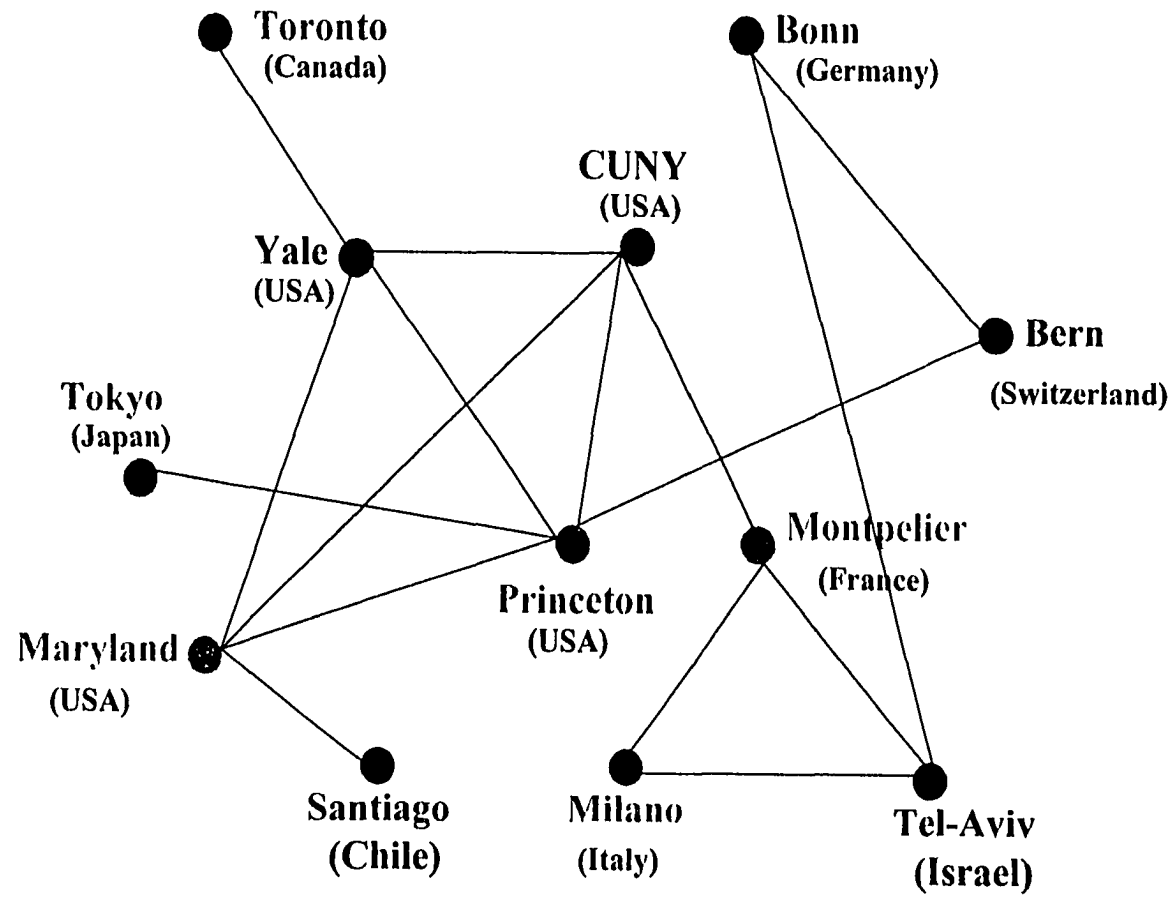


Figure 8. 6: Second Simulation Model Topology

8.3. Simulation Methodology

The model used for study is the 6 node-model presented in 8.2.1. We study link utilization, WAN clouds, and message sources for compressed data and uncompressed data. The study includes five different applications: E-mail, Audio, Image, Video, and HDTV. The bandwidth used for simulation runs depends on the type of application under study. The bandwidth is divided into five percentages, 50%, 60%, 70%, 80%, and 90% (traffic intensity) for the purpose of study. So we have 5 simulation runs for compressed data and uncompressed data each. Compression algorithms used for simulation are Huffman Coding, Lempel, Ziv, and Welch (LZW), MPEG-1, and MPEG-2 [9, 21, 31, 47, 49, 56]. COMNET III simulation network model is presented in **Figure 8.7.**

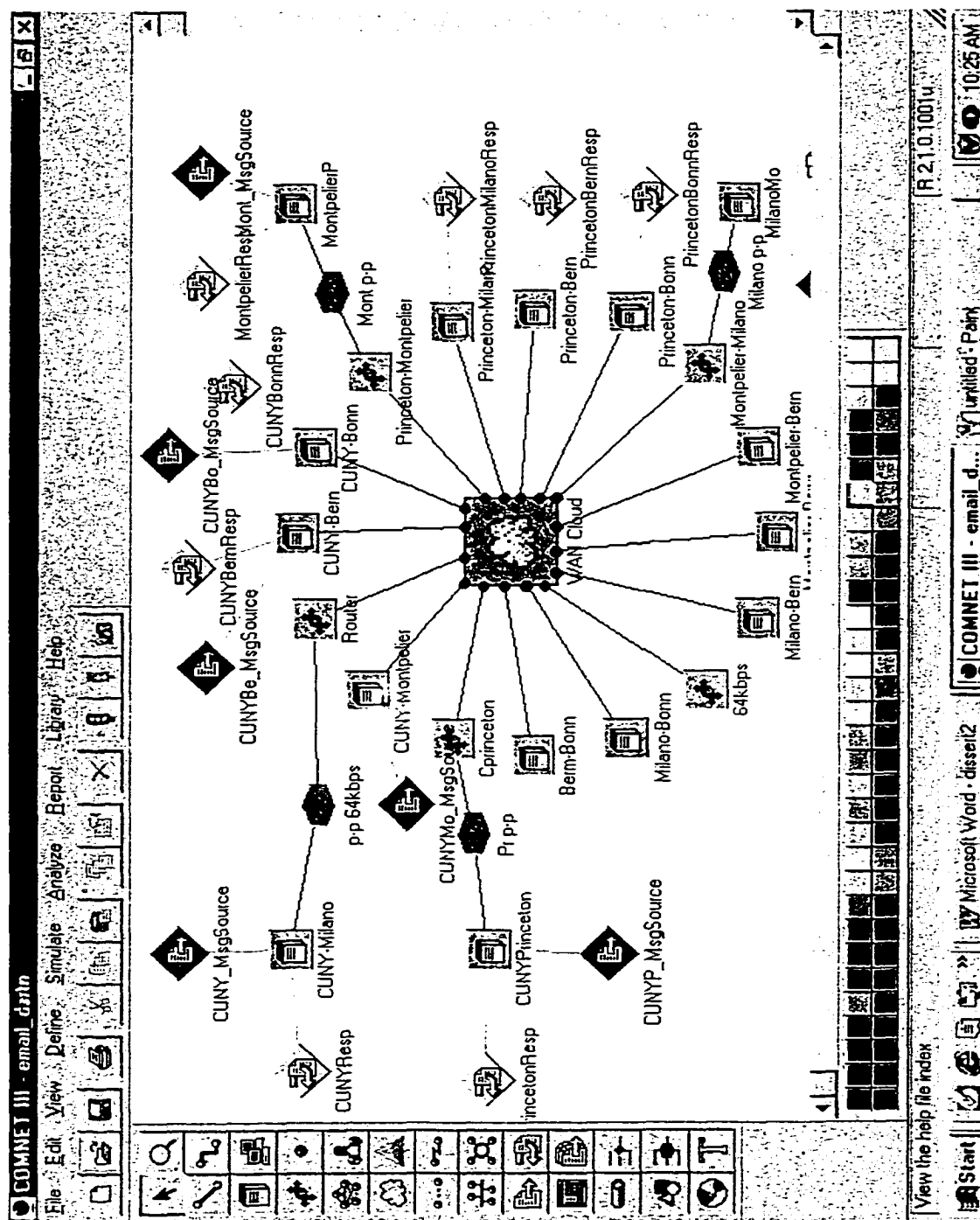


Figure 8. 7: COMNET III Simulation Network Model

8.3.1. **Link Utilization**

We have three categories in links: (1) **Channel Utilization**; (2) **Utilization by Application**; and (3) **Utilization by Protocol**.

Channel Utilization

The channel utilization provides the breakdown of the utilization rates for links used to carry connectionless and VC messages. It presents the number of link layer frames delivered and resent due to error (rst/err), and transmission delays and link utilization.

- **Frames Delivered**

are the number of frames removed from the output buffer at the transmitting node on the link and subsequently placed in the input buffer of the receiving node. Frames that are in transmission are not included as part of the data generated because of transmission delay and propagation delay.

- **Frames Resent/Error (rst/err)**

are the number of retransmitted frames.

- **Transmission Delay**

is the time between when the frame (contains part of a packet/several packets) is created at the input to the link and when the frame is delivered at the end of the link. It includes transmission and propagation time and the time is measured in milliseconds. The average, standard deviation, and maximum delay are observed for any packet across the link.

- **Link Utilization**

The transmission time for a frame is calculated from its size divided by the link speed. The link is in use for this time. Utilization is the total usage time divided by the simulation run length.

Utilization by Application

The utilization by application provides a breakdown of link usage by application for connectionless and virtual circuit messages. In this category the number of packets delivered, packets/second, bytes delivered, kilobits/second (KBPS) delivered, the percentage of bytes in the category and the link utilization percentage produced by packets, are observed.

- **Packets Delivered**

The number of packets per application routed by the link.

- **Packets/Second**

The number of packets/second (pps), per application routed by the link.

- **Bytes Delivered**

Number of bytes delivered per application by the link.

- **KBPS Delivered**

Number of kilobits per second delivered by the link per application

- **% Bytes**

Percentage of total bytes delivered by the link for each application using the link.

- **Utilization %**

Percent utilization of the link per application

Utilization by Protocol

The utilization by protocol report provides a breakdown of link usage by protocol. The statistics in this category are identical to those in Utilization by Application.

8.3.2. Wide Area Networks (WAN)

WAN Cloud is available in COMNET III for modeling WAN services in terms of Access Links and Virtual Circuits (VCs). WAN Clouds are used to model frame relay, cell relay, and packet switching services. WAN Cloud behaves similarly to the link objects in that it delivers frames and models a delay

of these frames across the network. The cloud's internal structure is defined using Access Links and VCs. VCs are abstract models of sets of switches and links and connect to Access Links only. The Access Link is a special case of a Point-To-Point link that works with the WAN Cloud. They provide the point-of-presence to the WAN cloud. Outside the WAN Cloud (backbone) the connection between a processing node and the WAN Cloud is through an Access Point. An Access Point represents an Access Link in the cloud.

WAN Clouds

We have three types in WAN clouds: (1) **Frame Delay by VC**; (2) **Frame Counts by VC**; and (3) **Access Link Statistics at Entry and Exit**.

(1) Frame Delay by VC

The frame delay by VC presents data for the frame delay and burst size for each VC in a WAN cloud. The frame delay data are for all the frames that the WAN cloud successfully delivers. The burst size is monitored for each frame that the virtual circuit accepts. Thus, it captures instantaneous peak values.

- **Frame Delay**

is the delay, measured in milliseconds, that each frame will experience in a VC. The average, standard deviation and maximum frame delay incurred by a link layer frame are calculated.

- **Burst Size**

The average and maximum burst size, measured in kilobits, of packets transmitted across the virtual circuit (VC).

(2) Frame Count by VC

The frame count by VC provides the throughput measure for each virtual circuit both in terms of the number of frames that the virtual circuit accepted or dropped and the number of kilobits accepted or dropped. These numbers are further divided between normal and discard eligible (DE) frames.

- **Virtual Circuit (VC)**

The name of the virtual circuit defined inside the WAN cloud. For each VC, there are two rows of data. The first row gives the frames and the second row gives the kilobits. For each row there are separate counts for normal and DE frames for frames accepted and dropped. Frames that are dropped due to the excess burst size being exceeded will be counted as DE frames. The number of total kilobits transmitted increases with bandwidth as well as with increase in pps (packets/second).

(3) Access Link Statistics

The access link statistics are data/statistics for each access link in a WAN cloud. Each access link has two rows: the first is for the entry (source) link and

the second is for exit (destination) link. No buffer is modeled for the entry link.

Hence, those values are N/A.

- **Access Link**

Is the name of the access link in the WAN cloud. Each access link has entry and exit rows.

- **Frames Accepted**

The number of frames accepted by the entry and exit access link of the WAN cloud. For entry link, this count reflects the frames that found a path to the required destination. If a frame arrives that must go to a destination for which no virtual circuit is defined and the cloud does not allow transmission through non-VCs, then that frame will count as being dropped by the entry access link even though that frame's transmission will have utilized the link.

- **Frames Dropped**

The number of frames dropped by the entry and exit access links of the WAN cloud. For exit links, the count reflects the number of frames accepted or blocked by the exit buffer.

- **Buffer**

The maximum, average and standard deviation of the buffer size for the entry and exit access link.

- **Utilization %**

The entry and exit access link utilization in terms of percentage. The utilization is time-averaged over the replication, where at each event, the utilization is the number of busy circuits divided by the total number of circuits.

8.3.3. Message and Response Sources Delay

We have three categories in sources: (1) **Message Delay**; (2) **Messages Delivered**; and (3) **Packet Delay**.

(1) Message Delay

Message delay is the time between creating the first packet of the message on the originating node and the time that the originating node is notified that the message has been assembled by the destination.

- **Messages Assembled:** For each destination, the number of messages that have been completely assembled at the destination. Messages are broken into packets at the source node according to the transport protocol characteristics and then each packet is sent to the destination.

(2) Messages Delivered

The Message Delivery data show on the delay before the message is reassembled by the destination; that is the time between creating the first packet of the message on the originating node and the time of receiving the last packet on the destination node. The message may still be worked on by the source after the message is delivered because of retransmissions, waiting for acknowledgements, or for modeling the close-sequence of the connection, but these values are not factored into the message delivered delay.

(3) Packet Delay

The Packet Delay data are a summary of the number of packets created, delivered, resent, or dropped for each message and response source. It also provides the average and maximum packet delay. Packet Delay is the time between creating a packet on the originating node and the time of receiving the packet at the destination node.

- **Packets Created:** The number of packets created to send to the destinations listed in the message sources.
- **Packets Delivered:** the number of packets received at the destination. This may differ from the number of packets created by the number of packets that are in transit at the instant the report is written.
- **Packets Dropped:** The number of packets dropped, when a node or link fails.

8.4. Network Analysis

The network analysis involves two types of variables: dynamic and static. The dynamic variables, such as traffic characteristics, are expressed in terms of statistical probabilities and probability distribution functions. For traffic attributes, which are a time dependent component of the analysis, we rely upon queuing theory concepts so that we may perform an efficient analysis of the effect of different traffic patterns on a network. The static variables, such as characteristics of nodes and links, account for the major components of the network and must be enumerated for proper analysis. The queuing theory plays a crucial role in the analysis of networks regardless of the switching method. It enables us to identify probability distributions for events under varying conditions.

In a telecommunication network, the number and the size of messages arriving at a node may vary with time resulting in different probability distributions at different times. This type of statistical behavior is called a stochastic process. A type of stochastic process that is especially useful when modeling networks is the Poisson process. We have an approximate Poisson process with parameter $\lambda > 0$ if the following are satisfied:

- The numbers of change occurring in non-overlapping intervals are independent.
- The probability of exactly one change in a sufficiently short interval of length h is approximately λh .
- The probability of two or more changes in a sufficiently short interval is essentially zero.

The above definition implies that the Poisson process has the Markov property, which is sometimes called the memory-less property. That is to say that the probability that an event takes place does not depend on the events that have already taken place. Instead this probability depends on the time interval between the two events. Consequently, if a process is a Poisson process with rate λ , then it has a Poisson distribution which has the probability of x events (arrivals) taking place in any time interval of length $t > 0$ as the Poisson p.d.f. (probability density function)

$$f(x) = e^{-\lambda t} (\lambda t)^x / x!, \quad x = 0, 1, 2, \dots$$

with the mean and variance each equal to λ . The random variables associated with a Poisson distribution are discrete random variables. For modeling and analysis of mathematical models it more helpful to work with continuous random

variables. Fortunately, if we consider the time between two consecutive arrivals (interarrival time, IAT) rather than number of arrivals within a time interval, we find that it has continuous random variables with a negative exponential distribution. This distribution has the probability function

$$\lambda e^{-\lambda x}$$

where $x > 0$ is the waiting time between two arrivals, with the mean equal to $1/\lambda$ and variance equal to $1/\lambda^2$. The exponential function also preserves the Markov property, that is to say that the distribution of the wait time does not depend on the amount of time that has already passed.

The Markov property of the Poisson process is very useful when modeling arrivals to a queue at a node. The arriving messages may come from many different nodes, therefore are independent of each other. Hence the number of arrivals within a time interval is independent of other arrivals, depending only on the length of the time interval.

As with all statistical models, it must be pointed out that the empirical data obtained from observations may not exactly follow the Poisson process. However, the nice mathematical properties of the Poisson process enables one to

build and analyze a queuing system model easily. Fortunately, the assumption of a Poisson process can be justified on the grounds other than those of analytical convenience. The Poisson distribution, like the normal distribution, turns up as the limiting distribution of large sums.

Chapter 9

9. SIMULATION RESULTS

Overview

This chapter presents different types of applications and algorithms used for data compression. For each type of application the results of the simulation are presented for both compressed and uncompressed data types. The results are analyzed for each data type and are then compared one with another. Five different types of applications are used for compression: (a) **E-mail**, (b) **Audio**, (c) **Image**, (d) **Video**, and (e) **HDTV** (High-Definition TV). The compression algorithms applied are (1) **Huffman Coding**, (2) **Lempel, Ziv and Welch (LZW)**, (3) **MPEG-1**, and (4) **MPEG-2** [9, 21, 31, 47, 49, 56].

Five different percentages of bandwidth are used for simulation of compressed and uncompressed data types. The percentages of the bandwidth are 50%, 60%, 70%, 80%, and 90%. One simulation run of each percentage for each data type is performed. Thus, ten simulation runs are performed in each application type for both compressed and uncompressed data types. The total simulation runs for all five applications are 50.

Simulation Run Statistics

Different message sizes and bandwidths are used for each application. Interarrival time (IAT, the time from start of one message to the start of next) is determined using the message size and bandwidth. The number of packets per second is computed using IAT. The number of bits per second is equal to the bandwidth used for simulation.

Example: E-mail

Message Size = 100 bytes

Bandwidth = 64kbps (kilobits per second)

IAT at 50% bandwidth = $1/(\text{bandwidth}/\text{bits} * 0.5)$
 $= 1/(64000/800 * 0.5) = 0.025 \text{ seconds}$

Packets/Seconds = $1/\text{IAT} = 1/0.025 = 40$

Bits/Second = $64000 * 0.5 = 32000$

Simulation Run Time for each percentage of bandwidth- Length of real time to run the simulation.

For each application and data type the simulation results are presented below. The results for Links and EAN clouds as well as Message & Response Sources fluctuate over variation in bandwidth. This can be attributed to network delay and/or link/node failure.

- **Links: Channel Utilization/Utilization by Application**

For each application utilizing the link:

Frames/Packets Delivered- The number of packets per application routed by the link.

Packets/Second- The number of packets per second, per application routed by the link.

Bytes Delivered- The total number of bytes delivered per application by the link.

KBPS- The number of kilobits per second delivered by the link per application.

Channel Utilization- The percent utilization of the link per application.

- **WAN Clouds: Access Link Statistics**

Frames Accepted- The number of frames accepted by the entry access link of the WAN cloud.

Frames Dropped- The number of frames accepted by the entry access link of the WAN cloud.

Calculation of Drop Rate and Difference between the two data types.

Link Utilization (%)- The entry access link utilization in terms of percentage.

- **Message & Response Sources: Packets Delivered/Packet Delay**

Packets Created- The number of packets created to send to the listed destination. Not all packets need to be delivered before they appear on the report.

Packets Delivered- The number of packets received at the destination. This may differ from the number of packets created by the number of packets that are in transit at the instant the report is written.

Packets Dropped- The number of packets dropped.

Calculation of Drop Rate

Packet Delay (in milliseconds)- Average, Maximum
Average and maximum packet delay.

9.1. E-Mail

Along with the Web, electronic mail is one of the most popular Internet applications. E-mail is asynchronous- people send and read messages when it is convenient for them, without having to coordinate with other peoples' schedules. E-mail requires fully reliable data transfer, that is, no data loss. E-mail can make use of as much or as little bandwidth as happens to be available. Of course, the higher the bandwidth the better. Two compression algorithms, Huffman Coding and Lempel-Ziv-Welch (LZW) are used to simulate e-mail application.

The compression rates for Huffman coding and LZW are 40% and 60%, respectively. In each case, the message size is 100 bytes, warmup length

(specifies the time in the simulation when the application will begin to collect data) is 10 seconds and replication length (the number of seconds of simulated time during which statistics are collected for each report) is 50 seconds. The bandwidth used for simulation run is 64kbps for each compression algorithm. Telecommunication traffic is often described as a Poisson process. The number of messages in successive time intervals has been observed. The distribution of the number of observations in an interval is Poisson distributed. The parameter entered for the Poisson distribution is the interarrival time (IAT, the time from the start of one message to the start of the next). The total number of simulation runs for both algorithms are 20. Each simulation ran for approximately 2 minutes. The simulation run statistics for compressed and uncompressed data for the first algorithm, Huffman Coding are shown in **Tables 9.1** and **9.2**, respectively.

Examination of the data in **Tables 9.1** and **9.2** show that as the bandwidth increases:

1. IAT decreases
2. Packets/Second increase (see **Figures 9.1** and **9.2**)
3. Bits/Second increase
4. Simulation run time is the same for bandwidth runs and the message size used for E-mail is small.

5. The rate of transmission for compressed data at each bandwidth is less than for uncompressed data.

As the tables show, when LAT decreases, the number of packets/second increases linearly in both data types. This is depicted in **Figures 9.1** and **9.2**.

Bandwidth: 64kbps, Replication Length: 50s					
Message Size: 100 bytes = 800 bits					
Huffman Code (40%)	Bandwidth Used for Simulation				
	50%	60%	70%	80%	90%
InterArrival Time (IAT)	0.0425	0.0354	0.0303	0.0263	0.0238
Poisson Distribution (poi)					
Packets/Second(pps)	24	28	33	38	43
Bits/Second(bps)	19200	22400	26400	30400	34400
Simulation Time (min)	2	2	2	2	2

Table 9. 1: E-mail (Huffman) Simulation Run Statistics for Compressed Data

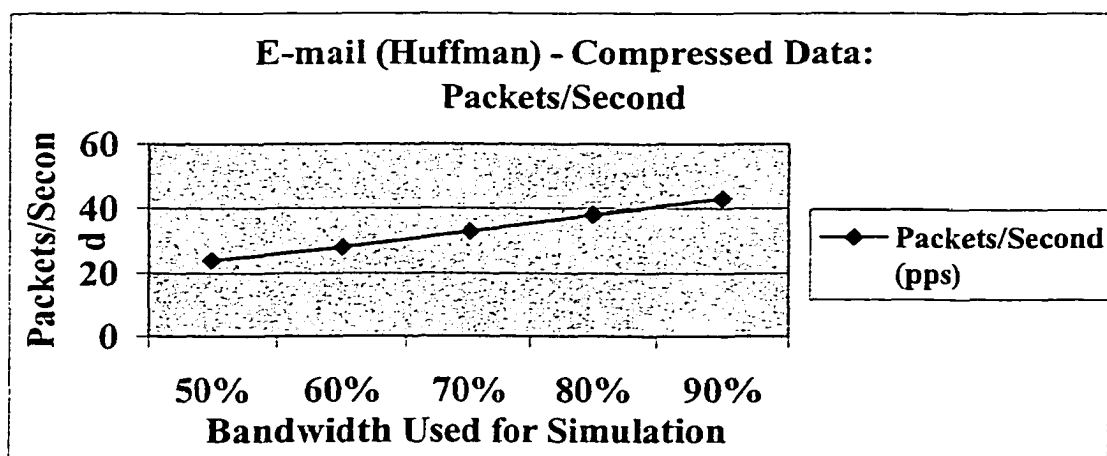


Figure 9. 1: E-mail (Huffman) - Compressed Data: Packets/Second

Bandwidth: 64kbps, Replication Length: 50s					
Message Size: 100 bytes = 800 bits					
Huffman Code (40%)	Bandwidth Used for Simulation				
	50%	60%	70%	80%	90%
InterArrival Time (IAT)	0.025	0.021	0.018	0.016	0.014
Poisson Distribution (poi)					
Packets/Second(pps)	40	48	56	64	72
Bits/Second(bps)	32000	38400	44800	51200	57600
Simulation Time (min)	2	2	2	2	2

Table 9. 2: E-mail (Huffman) Simulation Run Statistics for Uncompressed Data

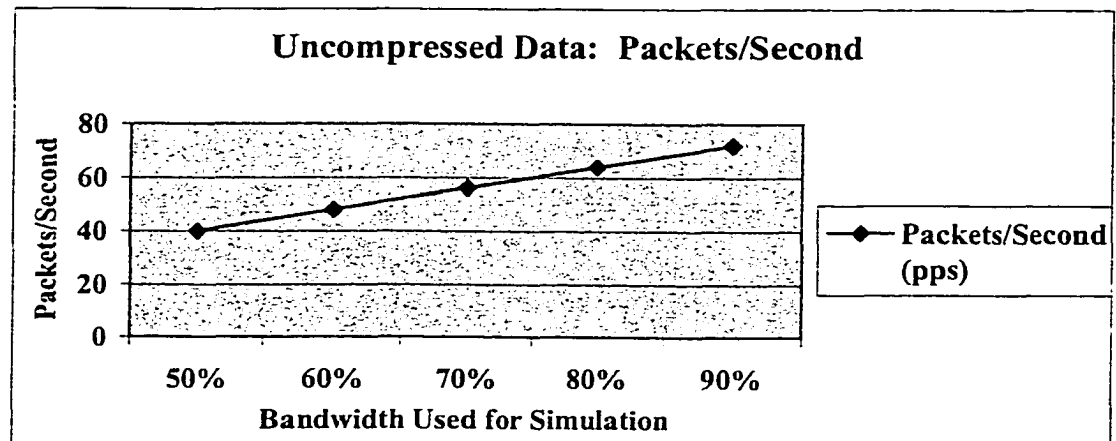


Figure 9. 2: E-mail (Huffman) - Uncompressed Data: Packets/Second

Huffman Coding

Huffman Coding is chosen for E-mail compression because text compression seems natural for it. In text, we have a discrete alphabet that has relatively stationary probabilities. For example, the probability model for a particular message will not differ significantly from that of another message.

Reports are produced automatically at the end of each simulation run. COMNET III generates a statistical report of Link Utilization for various processing nodes as shown in **Tables 9.3** and **9.4** for compressed and uncompressed data, respectively. The number of frames delivered is equal to the number of packets delivered from each link for E-mail application.

Analysis

- **Compressed Data**

The number of packets delivered from CUNY to Milano is 1223 at 50% bandwidth, but at 60% bandwidth, it is 1191, a decrease of 3 percent. As shown in **Table 9.3**, the link utilization rate went down as well by 0.03% at 60%, from 30.58% to 29.78%. The number of packets delivered at 70% increased to 1567, up by 31.5% from 60% bandwidth, but decreased to 1224 at 80% and again increased to 1613 at 90%. From CUNY to Princeton link, notice a gradual

increase from 50% to 60% bandwidth, but a decrease at 70% and an increase at 80% and 90%. From Montpelier to Princeton and Milano to Montpelier, however, the number of packets delivered increases as bandwidth increases. The link utilization also goes up for both links.

Application: E-mail (64kbps), Huffman Code (40%)					
Message Size = 100 bytes, Replication Length = 50s,					
Simulation Time = 2min					
Compressed Data	Bandwidth Used for Simulation				
Link	50%	60%	70%	80%	90%
	19.2kbps	22.4kbps	26.4kbps	30.4kbps	34.4kbps
From CUNY-Milano					
Frames/Packets Delivered	1223	1191	1567	1224	1613
Packets/second	24.460	23.820	31.340	24.480	32.260
Bytes Delivered	122300	119100	156700	122400	161300
KBPS Delivered	19.568	19.056	25.072	19.584	25.808
Utilization (%)	30.58	29.78	39.18	30.60	40.33
From CUNYPrinceton					
Frames/Packets Delivered	1592	1629	1593	1707	1867
Packets/second	31.840	32.580	31.860	34.140	37.340
Bytes Delivered	159200	162900	159300	170700	186700
KBPS Delivered	25.472	26.064	25.488	27.312	29.872
Utilization (%)	39.83	40.73	39.85	42.70	46.68
From MontpelierPri					
Frames/Packets Delivered	1347	1365	1545	2257	2463
Packets/second	26.940	27.300	30.900	45.140	49.260
Bytes Delivered	134700	136500	154500	225700	246300
KBPS Delivered	21.552	21.840	24.720	36.112	39.408
Utilization (%)	33.68	34.13	38.63	56.43	61.60
From Milano-Mont					
Frames/Packets Delivered	1308	1433	1611	1889	2321
Packets/second	26.160	28.660	32.220	37.780	46.420
Bytes Delivered	130800	143300	161100	188900	232100
KBPS Delivered	20.928	22.928	25.776	30.224	37.136
Utilization (%)	32.70	35.83	40.28	47.23	58.03

Table 9. 3: E-mail (Huffman) - Links: Channel Utilization/Utilization by Application for Compressed Data

- **Uncompressed Data**

There is an increase in the number of packets delivered from CUNY to Milano from 1515 at 50% bandwidth to 3275 at 70% bandwidth, as shown in **Table 9.4**. The number of packets delivered at 80% (3049) and 90% (2863) bandwidths decreases by 7% and 6%, respectively. The utilization of CUNY to Milano link increases from 50% to 70% and decreases at 80% and 90% bandwidths. From CUNY to Princeton link notice a gradual increase in the number of packets delivered from 1747 (at 50%), to 2738 (at 60%). However, the number of packets delivered at 70% is 2441, a decrease of 11%. At 80% (3032) and 90% (3364) bandwidths the number of packets delivered increases again by 24% and 11%, respectively. Similarly, an increase in the number of packets delivered from Montpelier to Princeton link is observed, 2272 at 50%, 2664 at 60%, 2118 at 70%, 3014 at 80% and 3655 at 90%. The utilization rate of both CUNY to Princeton and Montpelier to Princeton links decreases at 70%. From Milano to Montpelier link the number of packets delivered and utilization rate increase linearly from 50% bandwidth to 90% bandwidth.

Application: E-mail (64kbps), Huffman Code (40%)					
Message Size = 100 bytes, Replication Length = 50s,					
Simulation Time = 2min					
Uncompressed Data	Bandwidth Used for Simulation				
Link	50%	60%	70%	80%	90%
	32kbps	38.4kbps	44.8kbps	51.2kbps	57.6kbps
From CUNY-Milano					
Frames/Packets Delivered	1515	2299	3275	3049	2863
Packets/second	30.300	45.980	65.500	60.980	57.260
Bytes Delivered	151500	229900	327500	304900	286300
KBPS Delivered	24.240	36.784	52.400	48.874	45.808
Utilization (%)	37.88	57.50	81.88	76.25	71.60
From CUNYPrinceton					
Frames/Packets Delivered	1747	2738	2441	3032	3364
Packets/second	34.940	54.760	48.820	60.640	67.280
Bytes Delivered	174700	273800	244100	303200	336400
KBPS Delivered	27.952	43.808	39.056	48.512	53.824
Utilization (%)	43.68	68.45	61.03	75.83	84.13
From MontpelierPri					
Frames/Packets Delivered	2272	2664	2118	3014	3655
Packets/second	45.440	53.280	42.360	60.280	73.100
Bytes Delivered	227200	266400	211800	301400	365500
KBPS Delivered	36.352	42.624	33.888	48.224	58.480
Utilization (%)	56.80	66.60	52.95	75.35	91.40
From Milano-Mont					
Frames/Packets Delivered	1933	2294	2641	2806	3523
Packets/second	38.660	45.880	52.820	56.120	70.460
Bytes Delivered	193300	229400	264100	280600	352300
KBPS Delivered	30.928	36.704	42.256	44.896	56.368
Utilization (%)	48.33	57.33	66.05	70.15	88.08

Table 9. 4: E-mail (Huffman) - Links: Channel Utilization/Utilization by Application for Uncompressed Data

First row in **Tables 9.3** and **9.4** represents packets delivered during the simulation. They fluctuate for some links and increase as the bandwidth increases for others. This fluctuation could be a function of network delay and/or link/node failure.

Fluctuations in:

- Packets/Second (Packets Delivered/Replication Length, 50s);
- Bytes Delivered (Packets Delivered * Message Size);
- KBPS (Packets/Second * Message Size * 8bits); and
- Link Utilization

can be seen as reflections of the fluctuations in packets delivered.

Comparison of the results shown in **Table 9.5** demonstrates that the utilization rate for compressed data is consistently lower than for uncompressed data. This implies that compression permits transmission of larger volume of data in shorter intervals.

E-mail (Huffman)	Link Utilization (%)	
Link	Compressed Data	Uncompressed Data
From CUNY-Milano		
50%	30.58	37.88
60%	29.78	57.50
70%	39.18	81.88
80%	30.60	76.25
90%	40.33	71.60
From CUNYPrinceton		
50%	39.83	43.68
60%	40.73	68.45
70%	39.85	61.03
80%	42.70	75.83
90%	46.68	84.13
From MontpelierPri		
50%	33.68	56.80
60%	34.13	66.60
70%	38.63	52.95
80%	56.43	75.35
90%	61.60	91.40
From Milano-Mont		
50%	32.70	48.33
60%	35.83	57.33
70%	40.28	66.05
80%	47.23	70.15
90%	58.03	88.08

Table 9. 5: E-mail (Huffman) - Comparison of Utilization (%) for Compressed & Uncompressed Data

WAN Clouds: Access Link Statistics Analysis

The data shown in this section are based on the first simulation model (6-node) topology (see **Figure 8.5**). For the 6-node topology, WAN cloud contains 30 virtual circuits (VCs), a pair connecting each source/destination processing nodes. Each VC is connected to an access link with a dashed line indicating the flow of the traffic. The statistics for only 7 different access links (**Tables 9.6, 9.7, 9.8, and 9.9**) are presented since there would be an enormous amount of data for all 15 access links of both compressed and uncompressed data for all five different bandwidth percentages. The 7 access links are (1) CUNY-Montpelier, (2) CUNY-Milano, (3) CUNY-Bern, (4) CUNY-Bonn, (5) Princeton-Montpelier, (6) CUNY-Princeton, and (7) Montpelier-Milano. Only three of these links are chosen for analysis, namely CUNY-Montpelier, CUNY-Bonn, and Montpelier-Milano to demonstrate the pattern of variation over access links.

Compressed Data	Frames Accepted				
Access Link	50%	60%	70%	80%	90%
CUNY-Montpelier	979	1275	1158	1573	1743
CUNY-Milano	1096	1069	1425	1102	1458
CUNY-Bern	1113	1482	1427	1457	2643
CUNY-Bonn	850	1101	1272	1492	1338
Princeton-Montpelier	1217	1206	1363	2053	2185
CUNY-Princeton	1381	1474	1429	1522	1712
Montpelier-Milano	1172	1281	1438	1714	2095

Table 9. 6: E-mail (Huffman) - WAN Cloud Access Link Statistics: Frames Accepted (Compressed Data)

Compressed Data	Frames Dropped				
Access Link	50%	60%	70%	80%	90%
CUNY-Montpelier	91	107	120	200	206
CUNY-Milano	127	124	142	122	182
CUNY-Bern	130	165	143	179	288
CUNY-Bonn	95	120	140	175	155
Princeton-Montpelier	130	159	182	210	269
CUNY-Princeton	185	155	165	176	174
Montpelier-Milano	136	152	172	175	226

Table 9. 7: E-mail (Huffman) - WAN Cloud Access Link Statistics: Frames Dropped (Compressed Data)

Uncompressed Data	Frames Accepted				
Access Link	50%	60%	70%	80%	90%
CUNY-Montpelier	2057	2180	2890	2931	2997
CUNY-Milano	1371	2081	2915	2709	2357
CUNY-Bern	1452	1723	2121	3013	2790
CUNY-Bonn	1167	1874	2155	3075	3109
Princeton-Montpelier	2059	2399	1880	2741	3121
CUNY-Princeton	1587	2483	2179	2778	3013
Montpelier-Milano	1721	2156	2352	2541	3110

Table 9. 8: E-mail (Huffman) - WAN Cloud Access Link Statistics: Frames Accepted (Uncompressed Data)

Uncompressed Data	Frames Dropped				
Access Link	50%	60%	70%	80%	90%
CUNY-Montpelier	213	277	341	322	365
CUNY-Milano	138	253	355	307	288
CUNY-Bern	189	187	230	343	353
CUNY-Bonn	136	225	198	362	350
Princeton-Montpelier	223	265	220	300	355
CUNY-Princeton	160	265	246	324	342
Montpelier-Milano	213	215	271	316	347

Table 9. 9: E-mail (Huffman) - WAN Cloud Access Link Statistics: Frames Dropped (Uncompressed Data)

- **Compressed Data**

CUNY-Montpelier access link accepted 979 frames at 50% and 1275 frames at 60%, an increase of 30 percent. At 70% the number of frames accepted by the link went down to 1158 frames, a decrease of 9%. The number of frames accepted at 80% went up by 415 frames, an increase of 35% and at 90% the frames accepted are 1743, an increase of 11% as illustrated in **Figure 9.3**. The number of frames dropped by CUNY-Montpelier access link at different bandwidth simulation runs are 91, 107, 120, 200, and 206, as shown in **Figure 9.4**.

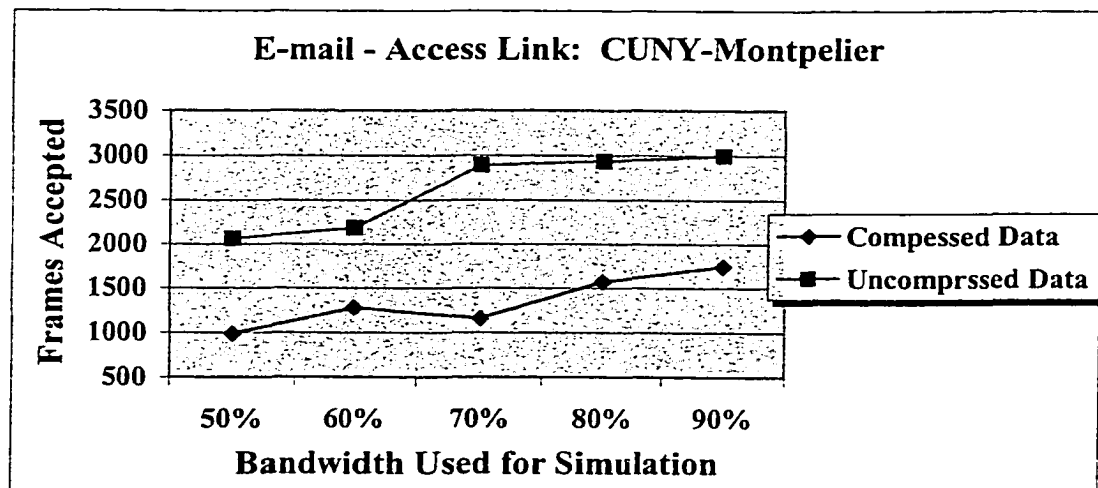


Figure 9. 3: E-mail (Huffman) – Selected Access Link: CUNY-Montpelier (Frames Accepted)

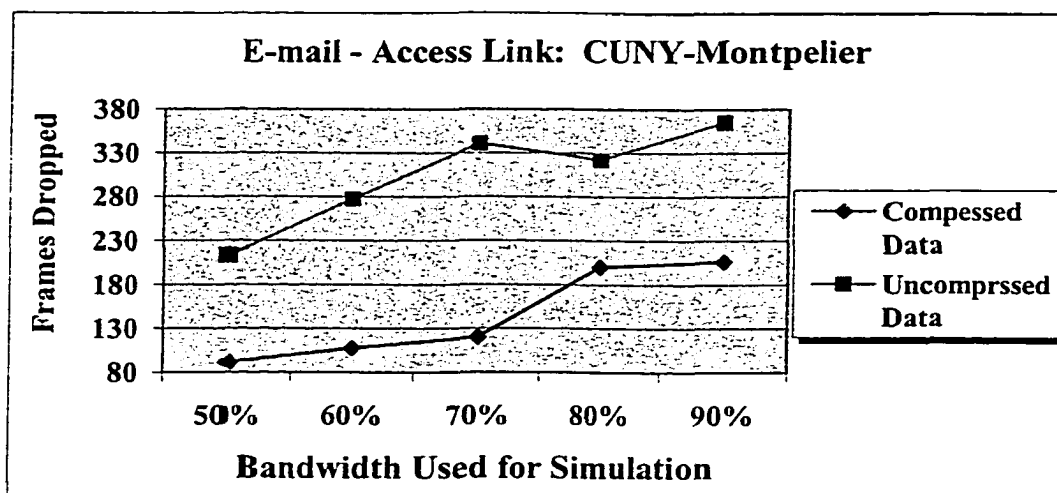


Figure 9. 4: E-mail (Huffman) – Selected Access Link: CUNY-Montpelier (Frames Dropped)

CUNY-Bonn access link accepted 850 frames at 50% and 1101 frames at 60%, an increase of 23 percent. At 70% the number of frames accepted by the link went up to 1272, an increase of 13%. The number of frames accepted at 80% went up to 1492, 14% increase, and at 90% the frames accepted are 1338, a decrease of 12%. CUNY-Bonn access link dropped frames at each simulation run. The number of dropped frames at each bandwidth percentage is 95 at 50%, 120 at 60%, 140 at 70%, 175 at 80% and 155 at 90%. The drop rate ranges from 9% to 12%.

The number of frames accepted by Montpelier-Milano access link for compressed data at different bandwidth simulation runs are 1172, 1281, 1438, 1714, and 2095. The numbers indicate that the increase from one percentage bandwidth to another is linear. Montpelier-Milano access link dropped frames at each simulation run. The number of dropped frames at each bandwidth percentage is 136 at 50%, 152 at 60%, 172 at 70%, 175 at 80% and 226 at 90%. The drop rate ranges from 10% to 12%.

- **Uncompressed Data**

CUNY-Montpelier access link accepted 2057 frames at 50% and 2180 frames at 60%, an increase of 6 percent. At 70% the number of frames accepted by the link went up to 2890 frames, an increase of 32%. The number of frames accepted at 80% went up by only 41 frames and at 90% the frames accepted are 2997, an increase of 2%. The frames accepted by CUNY-Montpelier access link are graphically shown in **Figure 9.3**. CUNY-Montpelier access link dropped frames at each simulation run, as shown in **Figure 9.4**. The number of dropped frames at each bandwidth percentage is 213 at 50%, 277 at 60%, 341 at 70%, 322 at 80% and 365 at 90%. The drop rate ranges from 10% to 13%.

CUNY-Bonn access link accepted 1167 frames at 50% and 1874 frames at 60%, an increase of 60 percent. At 70% the number of frames accepted by the link went up to 2155 frames, an increase of 15%. The number of frames accepted at 80% went up to 3075 frames and at 90% the frames accepted are 3109, an increase of 1%. CUNY-Bonn access link dropped frames at each simulation run. The number of dropped frames at each bandwidth percentage is 136 at 50%, 225 at 60%, 198 at 70%, 362 at 80% and 350 at 90%. The drop rate ranges from 9% to 12%.

The number of frames accepted by Montpelier-Milano access link for uncompressed data at different bandwidth simulation runs are 1721, 2156, 2352, 2541, and 3110. The numbers indicate that the increase from one percentage bandwidth to another is linear. Montpelier-Milano access link dropped frames at each simulation run. The number of dropped frames at each bandwidth percentage is 213 at 50%, 215 at 60%, 271 at 70%, 316 at 80% and 347 at 90%. The drop rate ranges from 10% to 12%.

Comparison of Frames-Dropped

The drop rate of compressed and uncompressed data and their differences are shown in **Table 9.10**. Also shown in the table is the comparison of the link utilization rate for both data types.

The drop rate of CUNY-Montpelier access link at 50% bandwidth for compressed and uncompressed data is 9% and 10%, respectively. The difference between the two is $\pm 1\%$ falling within the margin of error $\pm 5\%$. Similarly, the difference for the remaining bandwidth percentages, 60%-90%, is $\pm 4\%$, $\pm 1\%$, $\pm 2\%$, and 0%. The link utilization rate is twice as much for uncompressed data nearly in all cases.

The drop rate of CUNY-Bonn access link at 50% bandwidth for compressed and uncompressed data is 11% and 12%, respectively. The difference between the two is $\pm 1\%$ falling within the margin of error $\pm 5\%$. Similarly, the difference for the remaining bandwidth percentages, 60%-90%, is $\pm 1\%$, $\pm 2\%$, $\pm 0\%$, and 1%. The link utilization rate is 26.44% for compressed data compared to 36.48% for uncompressed data at 50% bandwidth, substantially less. We observe that the link utilization rate is less for compressed data at 60%-90% bandwidth simulation runs: at 60% 34.19% compared to 58.80%; at 70% 39.54% compared to 65.86%; at 80% 46.68% compared to 96.23%; and at 90% 41.83% compared to 96.86%.

The drop rate of Montpelier-Milano access link at 50% bandwidth is equal to 12% for both compressed data and uncompressed data and the difference between the two is zero. Similarly, the difference for the remaining bandwidth

percentages, 60%-90%, is $\pm 2\%$, $\pm 0\%$, $\pm 2\%$, and 0% . The link utilization rate at 50% bandwidth is 36.62% for compressed data compared to 54.14% for uncompressed data. We observe that the link utilization rate is less for compressed data at 60%-90% bandwidth simulation runs: at 60% 40.12% compared to 66.36%; at 70% 45.09% compared to 73.47%; at 80% 52.89% compared to 80.01%; and at 90% 64.99% compared to 96.80%.

This demonstrates that while the drop rate is more or less the same for both compressed and uncompressed data, compressing data has a clear and substantive advantage in terms of the link utilization rate. Thus, data compression allows for transmission of substantially larger volume of data in shorter intervals.

Huffman Code (40%)	WAN CLOUDS: ACCESS LINK STATS (ENTRY)									
Message Size: 100 bytes	Comparison of Compressed & Uncompressed E-mail Data									
Replication Length: 50s	Frames Accepted/Dropped/Link Utilization									
Cloud: Access Link	Cmp 50%	Uncmp 50%	Cmp 60%	Uncmp 60%	Cmp 70%	Uncmp 70%	Cmp 80%	Uncmp 80%	Cmp 90%	Uncmp 90%
CUNY-Montpelier										
Frames Accepted	979	2057	1275	2180	1158	2890	1573	2931	1743	2997
Frames Dropped	91	213	107	277	120	341	200	322	206	365
Drop Rate	0.0929	0.10354	0.08392	0.12706	0.1036	0.11799	0.1271	0.10986	0.11818	0.12178
Drop Rate (%)	9%	10%	8%	13%	10%	12%	13%	11%	12%	12%
Difference	-1%		-4%		-1%		2%		-0%	
Link Utilization (%)	29.96	63.54	38.70	68.81	35.78	90.46	49.64	91.07	54.57	94.14
CUNY-Milano										
Frames Accepted	1096	1371	1069	2081	1425	2915	1102	2709	1458	2357
Frames Dropped	127	138	124	253	142	355	122	307	182	288
Drop Rate	0.1158	0.10065	0.11599	0.12157	0.0996	0.12178	0.1107	0.11332	0.12482	0.12218
Drop Rate (%)	12%	10%	12%	12%	10%	12%	11%	11%	12%	12%
Difference	2%		-1%		-2%		-0%		0%	
Link Utilization (%)	34.24	42.27	33.39	65.36	43.88	91.55	34.27	84.44	45.91	74.07

	Frames Accepted/Dropped/Link Utilization (Cont'd.)									
Cloud: Access Link	Cmp 50%	Uncmp 50%	Cmp 60%	Uncmp 60%	Cmp 70%	Uncmp 70%	Cmp 80%	Uncmp 80%	Cmp 90%	Uncmp 90%
CUNY-Bern										
Frames Accepted	1113	1452	1482	1723	1427	2121	1457	3013	2643	2790
Frames Dropped	130	189	165	187	143	230	179	343	288	353
Drop Rate	0.1168	0.13016	0.11133	0.10853	0.1002	0.10843	0.1228	0.11384	0.10896	0.12652
Drop Rate (%)	12%	13%	11%	11%	10%	11%	12%	11%	11%	13%
Difference	-1%		0%		-1%		1%		-2%	
Link Utilization (%)	34.80	45.95	46.12	53.50	43.96	65.81	45.79	93.97	82.06	88.03
CUNY-Bonn										
Frames Accepted	850	1167	1101	1874	1272	2155	1492	3075	1338	3109
Frames Dropped	95	136	120	225	140	198	175	362	155	350
Drop Rate	0.1117	0.11653	0.10899	0.12006	0.1100	0.09187	0.1172	0.11772	0.11584	0.11257
Drop Rate (%)	11%	12%	11%	12%	11%	9%	12%	12%	12%	11%
Difference	-0%		-1%		2%		-0%		0%	
Link Utilization (%)	26.44	36.48	34.19	58.80	39.54	65.86	46.68	96.23	41.83	96.86
Princeton-Montpelier										
Frames Accepted	1217	2059	1206	2399	1363	1880	2053	2741	2185	3121
Frames Dropped	130	223	159	265	182	220	210	300	269	355
Drop Rate	0.1068	0.10830	0.13184	0.11046	0.1335	0.11702	0.1022	0.10944	0.12311	0.11374
Drop Rate (%)	11%	11%	13%	11%	13%	12%	10%	11%	12%	11%
Difference	-0%		2%		2%		-1%		1%	
Link Utilization (%)	37.72	63.88	38.22	74.59	43.26	58.81	63.35	85.15	68.73	97.33

	Frames Accepted/Dropped/Link Utilization (Cont'd.)									
Cloud: Access Link	Cmp 50%	Uncmp 50%	Cmp 60%	Uncmp 60%	Cmp 70%	Uncmp 70%	Cmp 80%	Uncmp 80%	Cmp 90%	Uncmp 90%
CUNY-Princeton										
Frames Accepted	1381	1587	1474	2483	1429	2179	1522	2778	1712	3013
Frames Dropped	185	160	155	265	165	246	176	324	174	342
Drop Rate	0.1339	0.10081	0.10515	0.10672	0.1154	0.11289	0.1156	0.11663	0.10163	0.11350
Drop Rate (%)	13%	10%	11%	11%	12%	11%	12%	12%	10%	11%
Difference	3%		-0%		0%		-0%		-1%	
Link Utilization (%)	43.86	48.92	45.61	76.93	44.63	67.93	47.56	86.84	52.78	93.95
Montpelier-Milano										
Frames Accepted	1172	1721	1281	2156	1438	2352	1714	2541	2095	3110
Frames Dropped	136	213	152	215	172	271	175	316	226	347
Drop Rate	0.1160	0.12376	0.11865	0.09972	0.1196	0.11522	0.1021	0.12436	0.10787	0.11157
Drop Rate (%)	12%	12%	12%	10%	12%	12%	10%	12%	11%	11%
Difference	-1%		2%		0%		-2%		-0%	
Link Utilization (%)	36.62	54.14	40.12	66.36	45.09	73.47	52.89	80.01	64.99	96.80

Table 9. 10: E-mail (Huffman) - WAN Clouds: Access Link Statistics- Frames Accepted, Frames Dropped & Link Utilization

Message & Response Sources

Table 9.11 shows data for 6 distinct origin nodes that send messages to 6 different destination nodes. Of these, only two nodes are chosen for analysis that follows:

CUNY-Milano/CUNY-Princeton (Origin/Destination)

Compressed Data

CUNY-Milano, origin node, created 1223, 1191, 1567, 1224, and 1596 to send to CUNY-Princeton, destination node, at 50%, 60%, 70%, 80%, and 90% bandwidth, respectively. The number of packets delivered at the destination is 391, 388, 468, 369, and 467 at the five simulation runs, respectively. And the number of packets dropped similarly is 166, 167, 193, 167, and 222. The number of packets created by the origin node at 60% bandwidth decreased by approximately 3% from the 50% bandwidth and increased by 32% to 1567 packets at 70%. We observe a decrease of 22% and an increase of 30% at 80% and 90% bandwidth simulation runs, respectively. The average packet delay (in milliseconds) is higher when the number of packets created is more and lower for smaller number of packets created.

Uncompressed Data

CUNY-Milano, origin node, created 1515, 2188, 3178, 3164, and 3483 to send to CUNY-Princeton, destination node, at 50%, 60%, 70%, 80%, and 90% bandwidth, respectively. The number of packets delivered at the destination is 397, 531, 521, 463, and 373 at the five simulation runs, respectively. And the number of packets dropped similarly is 178, 311, 415, 345, and 329. The number of packets created by the origin node at 80% bandwidth decreased by 14 packets from the 70% bandwidth and increased by 10% to 3483 packets at 90%. The average packet delay (in milliseconds) at 70% bandwidth is 28621.252ms, less than that of 60% bandwidth, 30544.959. However, the number of packets delivered is more at 70% than at 60%. Similarly, the average packet delay is more at 80% than at 90% because the number of packets delivered is higher at 80% than at 90%.

Comparison of Compressed and Uncompressed Data

As can be expected, the number of packets created for compressed data is lower than that of uncompressed data for all 5 bandwidth simulation runs. The number of packets delivered for compressed data is less for 50%-80% bandwidths than for uncompressed data. The number of packets dropped similarly is less for compressed than for uncompressed data for all simulation runs. The ratio of packets dropped over packets delivered (packets-dropped/packets-delivered) is 42%, 43%, 41%, 45%, and 48% for compressed data and it is 45%, 59%, 80%,

75%, and 88% for uncompressed data. The high ratios at 70%-90% bandwidths for uncompressed data is due to the increase in packets dropped.

MontpelierP/Montpelier-Milano (Origin/Destination)

Compressed Data

MontpelierP, origin node, created 1347, 1365, 1545, 2182, and 2465 to send to Montpelier-Milano, destination node, at 50%, 60%, 70%, 80%, and 90% bandwidth, respectively. The number of packets delivered at the destination is 464, 514, 466, 748, and 486 at the five simulation runs, respectively. And the number of packets dropped similarly is 179, 221, 223, 288, and 339. The number of packets created and the number of packets dropped increase linearly. The variation in the number of packets delivered over the range of the bandwidths observed in **Table 9.11** may be attributable to network delay and/or node/link failure.

Uncompressed Data

MontpelierP, origin node, created 2197, 2664, 2118, 3014, and 3867 to send to Montpelier-Milano, destination node, at 50%, 60%, 70%, 80%, and 90% bandwidth, respectively. The number of packets delivered at the destination is 612, 434, 394, 490, and 520 at the five simulation runs, respectively. And the number of packets dropped similarly is 303, 303, 280, 351, and 435. The number

of packets created at 70% bandwidth decreased by 546 packets or 20% from the 60% bandwidth number and increased by 42% to 3014 packets from 2118 packets. The average packet delay (in milliseconds) at 70% bandwidth is 26877.438ms, more than that of 60% bandwidth. However, the number of packets created is less at 70% than at 60%.

Comparison of Compressed and Uncompressed Data

As expected, the number of packets created for compressed data is less than that of uncompressed data for all 5 bandwidth simulation runs, a function of compression. The number of packets delivered for compressed data is more for 70% and 80% bandwidths (466 and 748), respectively, than for uncompressed data. The number of packets dropped for compressed data is also less for all simulation runs than for uncompressed data. The ratio of packets dropped over packets delivered ($\text{packets dropped} / \text{packets delivered}$) is 39%, 43%, 48%, 39%, and 70% for compressed data and it is 50%, 70%, 71%, 72%, and 84% for uncompressed data. We see high percentages at 90% bandwidth for compressed data (70%) and at 60%-90% bandwidths for uncompressed data. This is due to the increase in packets dropped.

Application: E-mail (64kbps), Huffman Code (40%)										
Message Size = 100bytes, Replication Length = 50s, Simulation Time = 2min										
	Message & Response Sources: Packets Delivered/Packet Delay									
	Compressed Data					Uncompressed Data				
	Bandwidth Used For Simulation									
Origin/Destination	50%	60%	70%	80%	90%	50%	60%	70%	80%	90%
CUNY-Milano/ CUNY-Princeton										
Packets Created	1223	1191	1567	1224	1596	1515	2188	3178	3164	3483
Packets Delivered	391	388	468	369	467	397	531	521	463	373
Packets Dropped	166	167	193	167	222	178	311	415	345	329
Dropped/Created	14%	14%	12%	14%	14%	12%	14%	13%	11%	9%
Dropped/Delivered	42%	43%	41%	45%	48%	45%	59%	80%	75%	88%
Packet Delay (MS)										
Average	22123.158	20396.967	25587.939	23435.029	29973.043	22330.287	30544.959	28621.252	28872.840	24603.479
Maximum	33976.503	38892.503	40332.503	44234.503	45990.503	41204.503	49976.503	49920.503	49962.503	46680.503
MontpelierP/ Montpelier-Milano										
Packets Created	1347	1365	1545	2182	2465	2197	2664	2118	3014	3867
Packets Delivered	464	514	466	748	486	612	434	394	490	520
Packets Dropped	179	221	223	288	339	303	303	280	351	435
Dropped/Created	13%	16%	14%	13%	14%	14%	11%	13%	12%	11%
Dropped/Delivered	39%	43%	48%	39%	70%	50%	70%	71%	72%	84%
Packet Delay (MS)										
Average	23302.976	26389.418	21797.645	28008.536	29177.816	27942.616	25532.075	26877.438	27956.487	31580.186
Maximum	33558.002	38956.002	40992.002	45718.002	46992.002	45964.002	47992.002	46936.002	48922.002	52964.002

	Message & Response Sources: Packets Delivered/Packet Delay (Cont'd.)									
	Compressed Data					Uncompressed Data				
	Bandwidth Used For Simulation									
Origin/Destination	50%	60%	70%	80%	90%	50%	60%	70%	80%	90%
CUNY-Bern/ Bern-Bonn										
Packets Created	1243	1647	1570	1628	2926	1641	1939	2468	3381	3573
Packets Delivered	387	724	443	445	467	426	225	411	401	267
Packets Dropped	174	261	183	223	347	244	213	277	395	378
Dropped/Created	14%	16%	12%	14%	12%	15%	11%	11%	12%	11%
Dropped/Delivered	45%	36%	41%	50%	74%	57%	95%	67%	99%	142%
Packet Delay (MS)										
Average	19603.092	23940.847	28004.602	22776.554	26231.462	22947.757	24652.286	30012.483	30004.575	26497.088
Maximum	33004.002	39348.002	40684.002	43978.002	43262.002	42900.002	46936.002	46978.002	49978.002	47256.002
CUNY-Bonn/ Milano-Bonn										
Packets Created	931	1221	1412	1667	1525	1303	2102	2124	3475	3384
Packets Delivered	459	502	475	499	370	253	347	603	527	519
Packets Dropped	137	183	219	230	190	161	262	262	422	419
Dropped/Created	15%	15%	16%	14%	12%	12%	12%	12%	12%	12%
Dropped/Delivered	30%	36%	46%	46%	51%	64%	76%	43%	80%	81%
Packet Delay (MS)										
Average	21542.995	20558.185	27260.608	24774.731	22286.726	19326.784	30056.405	29382.104	30057.273	31262.441
Maximum	33592.002	37514.002	40606.002	44922.002	43730.002	41388.002	47390.002	50922.002	52992.002	53936.002

	Message & Response Sources: Packets Delivered/Packet Delay (Cont'd.)									
	Compressed Data					Uncompressed Data				
	Bandwidth Used For Simulation									
Origin/Destination	50%	60%	70%	80%	90%	50%	60%	70%	80%	90%
CUNY-Montpelier/ Montpelier-Milano										
Packets Created	1070	1382	1278	1773	1949	2183	2324	3388	4019	3177
Packets Delivered	330	332	384	305	489	595	590	530	318	544
Packets Dropped	129	139	158	228	271	286	358	400	368	420
Dropped/Created	12%	10%	12%	13%	14%	13%	15%	12%	9%	13%
Dropped/Delivered	39%	42%	41%	75%	55%	48%	61%	75%	116%	77%
Packet Delay (MS)										
Average	18022.171	25053.616	19532.84	28173.038	23068.251	26360.600	29519.463	28544.605	30580.052	30885.248
Maximum	33460.002	38908.002	38814.002	43992.002	44578.002	43142.002	49908.002	48782.002	48936.002	52978.002
CUNYPrinceton/ Princeton-Mont										
Packets Created	1605	1629	1594	1741	1861	1747	2714	2441	2918	3399
Packets Delivered	674	428	421	370	456	527	485	359	528	520
Packets Dropped	266	223	202	220	217	218	307	294	369	400
Dropped/Created	17%	14%	13%	13%	12%	12%	11%	12%	13%	12%
Dropped/Delivered	39%	52%	48%	59%	48%	41%	63%	82%	70%	77%
Packet Delay (MS)										
Average	20175.23	21152.642	23247.622	31068.953	27806.598	26516.745	28486.175	24852.620	29751.767	29513.84
Maximum	33978.002	39474.002	39732.002	43950.002	44788.002	43936.002	47880.002	46950.002	51796.002	49992.002

Table 9. 11: E-mail (Huffman) - Message & Response Sources: Packets Delivered/Delay

LZW (Lempel-Ziv-Welch)

The simulation run statistics for compressed and uncompressed data for the second algorithm, LZW are shown in **Tables 9.12** and **9.13**, respectively.

Examination of the data in **Tables 9.12** and **9.13** show that as the bandwidth increases:

1. IAT decreases
2. Packets/Second increase (see **Figures 9.5** and **9.6**)
3. Bits/Second increase
4. Simulation run time is the same for bandwidth runs and the message size used for E-mail is small.
5. The rate of transmission for compressed data at each bandwidth is less than for uncompressed data.

As the tables show, when IAT decreases, the number of packets/second increases linearly in both data types. This is depicted in **Figures 9.5** and **9.6**.

Bandwidth: 64kbps, Replication Length: 50s					
Message Size: 100 bytes = 800 bits					
LZW (60%)	Bandwidth Used for Simulation				
	50%	60%	70%	80%	90%
InterArrival Time (IAT)	0.0625	0.0521	0.0446	0.04	0.035
Poisson Distribution (poi)					
Packets/Second(pps)	16	19	22	25	29
Bits/Second(bps)	12800	15200	17600	20000	23200
Simulation Time (min)	2	2	2	2	2

Table 9. 12: E-mail (LZW) Simulation Run Statistics for Compressed Data

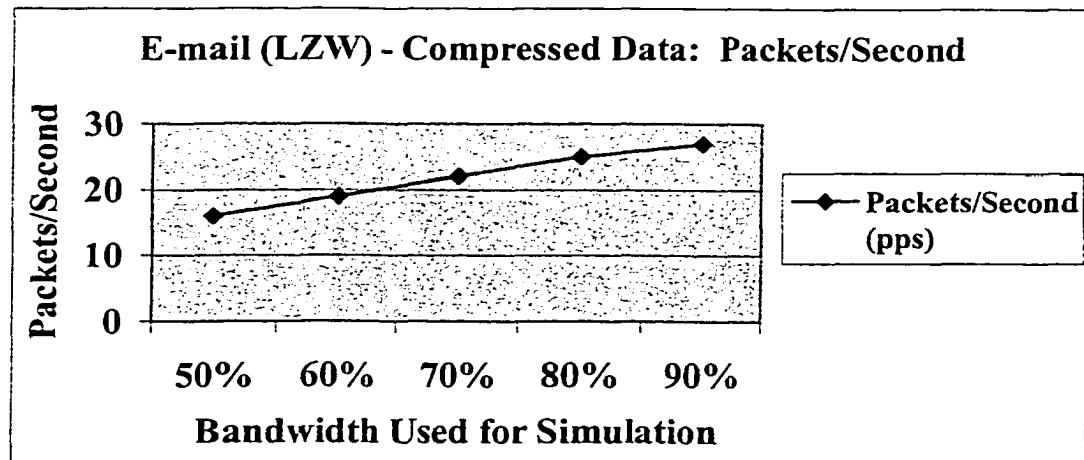


Figure 9. 5: E-mail (LZW) - Compressed Data: Packets/Second

Bandwidth: 64kbps, Replication Length: 50s					
Message Size: 100 bytes = 800 bits					
LZW (60%)	Bandwidth Used for Simulation				
	50%	60%	70%	80%	90%
InterArrival Time (IAT)	0.025	0.021	0.018	0.016	0.014
Poisson Distribution (poi)					
Packets/Second(pps)	40	48	56	63	71
Bits/Second(bps)	32000	38400	44800	50400	56800
Simulation Time (min)	2	2	2	2	2

Table 9. 13: E-mail (LZW) Simulation Run Statistics for Uncompressed Data

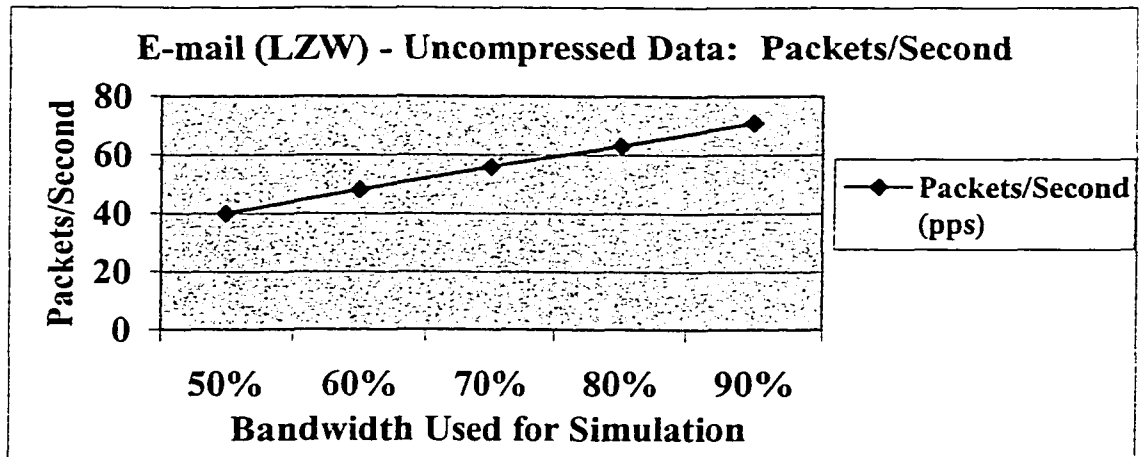


Figure 9. 6: E-mail (LZW) - Uncompressed Data: Packets/Second

Reports are produced automatically at the end of each simulation run. COMNET III generates a statistical report of Link Utilization for various processing nodes as shown in **Tables 9.14** for compressed and uncompressed data. The number of frames delivered is equal to the number of packets delivered from each link for E-mail application.

First row in **Table 9.14** represents packets delivered during the simulation. They fluctuate for some links and increase as the bandwidth increases for others. We ascribe such variation to the duration of the simulation run and also whether steady saturation conditions have been reached as the results are scanned.

Fluctuation in:

- Packets/Second (Packets Delivered/Replication Length, 50s);
- Bytes Delivered (Packets Delivered * Message Size);
- KBPS (kilobits/second, Packets/Second * Message Size * 8bits); and
- Link Utilization

can be seen as reflections of the variations in packets delivered.

Comparison of the results shown in **Table 9.14** demonstrates that the utilization rate for compressed data is consistently lower than for uncompressed data. This implies that compression permits transmission of larger volume of data in shorter intervals.

Analysis

- **Compressed Data**

The number of packets delivered from CUNY to Milano is a steady upward increase, 746, 939, 1137, 1214 and 1286 at 50%, 60%, 70%, 80% and 90% bandwidths, respectively. From 50% to 60% bandwidth we see a 26% increase in packets delivered. From 60% to 70% the increase is 21%, from 70% to 80% it is nearly 7% and at 90% the increase is 6%. As shown in **Table 9.14**, the link utilization rate is 18.65% at 50% and 60% bandwidths and then increases from 60% bandwidth to 90% bandwidths. From CUNY to Princeton link, notice a gradual increase from 50% to 70% bandwidth, but a decrease at 80% and an increase at 90%. This fluctuation reflects in link utilization rate as well. From Montpelier to Princeton the number of packets delivered increases linearly from 50% to 80% bandwidth and decreases at 90% bandwidth by nearly 4%. The link utilization rate also increases from 50% to 80% and decreases at 90% bandwidth. From Milano to Montpelier the number of packets delivered increases as bandwidth increases from 50% to 70%, however, decreases at 80% bandwidth by 24% and again increases at 90% bandwidth by 41%. The link utilization goes up from 16.43% to 37.80% from 50% to 70% bandwidth, goes down to 28.60% at 80% bandwidth, and again increases to 40.30% at 90% bandwidth.

- **Uncompressed Data**

In case of uncompressed data, the number of packets delivered from CUNY to Milano is 1515, 2299, 3275, 3049, and 2863 at 50%, 60%, 70%, 80%, and 90% bandwidths, respectively. We see an increase of 52% and 42% in frames/packets delivered from 50% to 60% and from 60% to 70%, respectively. However, the number of packets delivered decreases at 80% and 90% bandwidths by 7% and 6%, respectively. The slope of each category, packets/second, bytes delivered, kilobits per second (kbps), and link utilization percentage for 50%-70% bandwidths is upward and for 80% and 90% bandwidth it is downward. The numbers in the table for CUNY to Princeton and Montpelier to Princeton links indicate that the slopes of number of packets delivered is nonlinear. In case of Milano to Montpelier link, the number of packets delivered goes up linearly from 50% bandwidth to 90% bandwidth. These changes are also reflected in other categories, packets/second, bytes delivered, kbps delivered, and utilization percentage.

E-mail (64kbps)	Links: Channel Utilization/Utilization by Application									
LZW (60%)	Compressed Data					Uncompressed Data				
Message Size = 100B	Bandwidth Used For Simulation									
Link	50%	60%	70%	80%	90%	50%	60%	70%	80%	90%
From CUNY-Milano										
Packets Delivered	746	939	1137	1214	1286	1515	2299	3275	3049	2863
Packets/second	14.920	18.780	22.740	24.280	25.720	30.300	45.980	65.500	60.980	57.260
Bytes Delivered	74600	93900	113700	121400	128600	151500	229900	327500	304900	286300
KBPS Delivered	11.936	15.024	18.192	19.424	20.576	24.240	36.784	52.400	48.874	45.808
Utilization (%)	18.65	18.65	28.43	30.35	32.15	37.88	57.50	81.88	76.25	71.60
From CUNYPrinceton										
Packets Delivered	712	1085	1278	854	1468	1747	2738	2441	3032	3364
Packets/second	14.240	21.700	25.560	17.080	29.360	34.940	54.760	48.820	60.640	67.280
Bytes Delivered	71200	108500	127800	85400	146800	174700	273800	244100	303200	336400
KBPS Delivered	11.392	17.36	20.448	13.664	23.488	27.952	43.808	39.056	48.512	53.824
Utilization (%)	17.8	27.13	31.95	21.35	36.7	43.68	68.45	61.03	75.83	84.13

		Links: Channel Utilization/Utilization by Application								
		Compressed Data				Uncompressed Data				
		Bandwidth Used For Simulation								
Link	50%	60%	70%	80%	90%	50%	60%	70%	80%	90%
From MontpelierPri										
Packets Delivered	782	806	1008	1500	1447	2272	2664	2118	3014	3655
Packets/second	15.640	16.120	20.160	30.000	28.940	45.440	53.280	42.360	60.280	73.100
Bytes Delivered	78200	80600	100800	150000	144700	227200	266400	211800	301400	365500
KBPS Delivered	12.512	12.896	16.128	24	23.152	36.352	42.624	33.888	48.224	58.480
Utilization (%)	19.55	20.15	25.2	37.5	36.18	56.80	66.60	52.95	75.35	91.40
From Milano-Mont										
Packets Delivered	657	1114	1512	1144	1612	1933	2294	2641	2806	3523
Packets/second	13.140	22.280	30.240	22.880	32.240	38.660	45.880	52.820	56.120	70.460
Bytes Delivered	65700	111400	151200	114400	161200	193300	229400	264100	280600	352300
KBPS Delivered	10.512	17.824	24.192	18.304	25.792	30.928	36.704	42.256	44.896	56.368
Utilization (%)	16.43	27.83	37.8	28.6	40.3	48.33	57.33	66.05	70.15	88.08

Table 9. 14: E-Mail (LZW) - Links: Channel Utilization/Utilization by Application for Compressed & Uncompressed Data

WAN Clouds: Access Link Statistics Analysis

Tables 9.15, 9.16, 9.17 and 9.18 present statistics for 7 access links: (1) CUNY-Montpelier, (2) CUNY-Milano, (3) CUNY-Bern, (4) CUNY-Bonn, (5) Princeton-Montpelier, (6) CUNY-Princeton, and (7) Montpelier-Milano. Only three of these links are chosen for analysis to demonstrate the pattern of variation over access links.

Compressed Data	Frames Accepted				
Access Link	50%	60%	70%	80%	90%
CUNY-Montpelier	653	664	1042	1061	1352
CUNY-Milano	661	844	1021	1102	1144
CUNY-Bern	775	869	1193	902	1158
CUNY-Bonn	726	807	1031	1287	1238
Princeton-Montpelier	709	720	906	1351	1320
CUNY-Princeton	633	971	1140	775	1315
Montpelier-Milano	593	1024	1379	1032	1452

Table 9. 15: E-mail (LZW) - WAN Cloud Access Link Statistics: Frames Accepted (Compressed Data)

Compressed Data	Frames Dropped				
Access Link	50%	60%	70%	80%	90%
CUNY-Montpelier	66	89	132	139	159
CUNY-Milano	85	95	116	112	142
CUNY-Bern	86	105	135	91	119
CUNY-Bonn	84	106	105	150	147
Princeton-Montpelier	73	86	112	149	127
CUNY-Princeton	79	114	138	79	153
Montpelier-Milano	64	99	133	112	160

Table 9. 16: E-mail (LZW) - WAN Cloud Access Link Statistics: Frames Dropped (Compressed Data)

Uncompressed Data	Frames Accepted				
Access Link	50%	60%	70%	80%	90%
CUNY-Montpelier	2057	2180	2890	2931	2997
CUNY-Milano	1371	2081	2915	2709	2357
CUNY-Bern	1452	1723	2121	3013	2790
CUNY-Bonn	1167	1874	2155	3075	3109
Princeton-Montpelier	2059	2399	1880	2741	3121
CUNY-Princeton	1587	2483	2179	2778	3013
Montpelier-Milano	1721	2156	2352	2541	3110

Table 9. 17: E-mail (LZW) WAN Cloud Access Link Statistics: Frames Accepted (Uncompressed Data)

Uncompressed Data	Frames Dropped				
Access Link	50%	60%	70%	80%	90%
CUNY-Montpelier	213	277	341	322	365
CUNY-Milano	138	253	355	307	288
CUNY-Bern	189	187	230	343	353
CUNY-Bonn	136	225	198	362	350
Princeton-Montpelier	223	265	220	300	355
CUNY-Princeton	160	265	246	324	342
Montpelier-Milano	213	215	271	316	347

Table 9. 18: E-mail (LZW) - WAN Cloud Access Link Statistics: Frames Dropped (Uncompressed Data)

- **Compressed Data**

CUNY-Montpelier access link accepted 653 frames at 50% and 664 frames at 60%, an increase of 2 percent. At 70% the number of frames accepted by the link goes up to 1042 frames, an increase of 57%. The number of frames accepted at 80% differ from 70% by 19 frames and at 90% the frames accepted are 1352, an increase of 27%, as depicted in **Figure 9.7**. CUNY-Montpelier access link dropped frames at each simulation run. The number of dropped frames at each bandwidth percentage is 66 at 50%, 89 at 60%, 132 at 70%, 139 at 80% and 159 at 90%, as shown in **Figure 9.8**. The drop rate is 10%, 13%, 13%, 13%, and 12% for all five bandwidth simulation runs, 50%, 60%, 70%, 80%, and 90%, respectively.

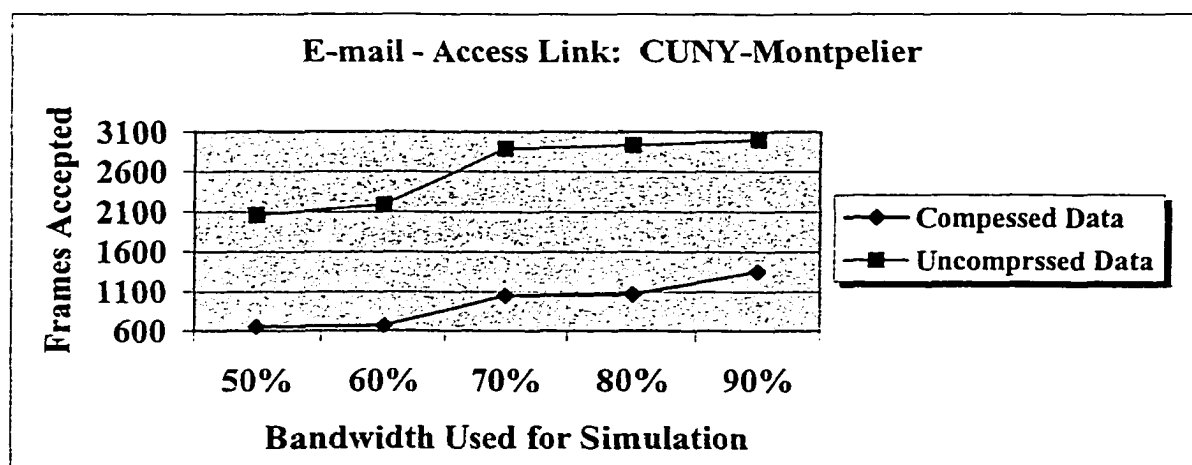


Figure 9. 7: E-mail (LZW) – Selected Access Link: CUNY-Montpelier (Frames Accepted)

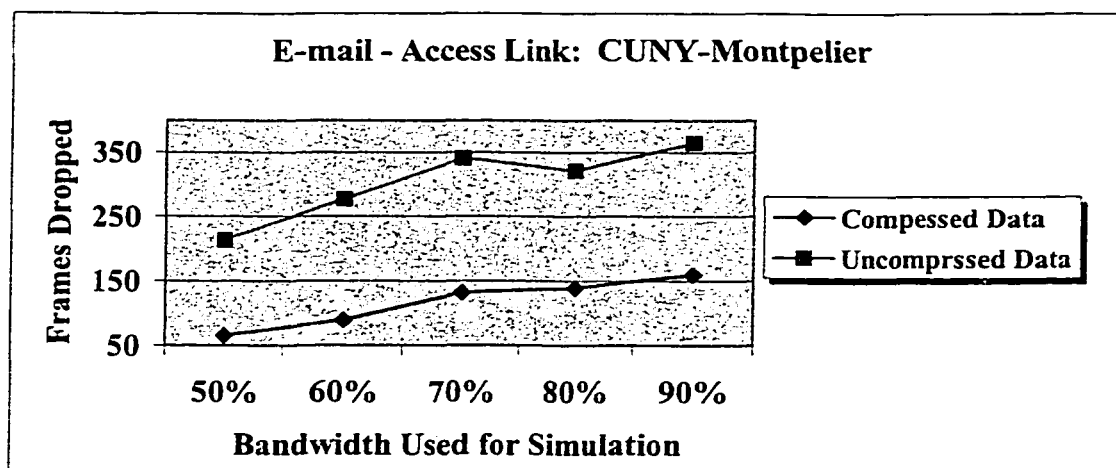


Figure 9. 8: E-mail (LZW) – Selected Access Link: CUNY-Montpelier (Frames Dropped)

CUNY-Bonn access link accepted 726 frames at 50% and 807 frames at 60%, an increase of 11 percent. At 70% the number of frames accepted by the link goes up to 1193, an increase of 37%. The number of frames accepted at 80% goes down to 902, a 24% decrease, and at 90% the frames accepted are 1158, an increase of 28%. CUNY-Bonn access link dropped frames at each simulation run. The number of dropped frames at each bandwidth percentage is 84 at 50%, 106 at 60%, 105 at 70%, 150 at 80% and 147 at 90%. The drop rate is 12%, 13%, 10%, 12%, and 12% for all five bandwidth simulation runs, 50%, 60%, 70%, 80%, and 90%, respectively.

The number of frames accepted by Montpelier-Milano access link for compressed data are 593, 1024, 1379, 1032, and 1452 at 50%, 60%, 70%, 80%, and 90% bandwidths, respectively. The numbers indicate that the increase from one percentage bandwidth to another is not linear. Montpelier-Milano access link dropped frames at each simulation run. The number of dropped frames at each bandwidth percentage is 64 at 50%, 99 at 60%, 133 at 70%, 112 at 80% and 160 at 90%. The drop rate ranges from 10% to 11%.

- **Uncompressed Data**

The number of accepted frames for CUNY-Montpelier access link are similar to that of compressed data, in the sense, at 50% bandwidth the number is 2057 and we see at 60% an increase of 6%. At 70% the number of frames accepted by the link goes up to 2890 frames, an increase of 33%. The number of frames accepted at 80% differ from 70% by 41 frames or 1% and at 90% the frames accepted are 2997, an increase of 2%. CUNY-Montpelier access link dropped frames at each simulation run. The number of dropped frames at each bandwidth percentage is 213 at 50%, 277 at 60%, 341 at 70%, 322 at 80% and 365 at 90%. The drop rate is 10%, 13%, 12%, 11%, and 12% for all five bandwidth simulation runs, 50%, 60%, 70%, 80%, and 90%, respectively. **Figures 9.7 and 9.8** depict the number of frames accepted and dropped, respectively.

The number of frames accepted by CUNY-Bonn access link are 1167, 1874, 2155, 3075, and 3109 at 50%, 60%, 70%, 80%, and 90% bandwidth simulation runs, respectively. These numbers indicate that the increase is upward from one simulation run to another. CUNY-Bonn access link dropped frames at each simulation run. The number of dropped frames at each bandwidth percentage is 136 at 50%, 225 at 60%, 198 at 70%, 362 at 80% and 350 at 90%. The drop rate is 12%, 12%, 9%, 12%, and 11% for all five bandwidth simulation runs, 50%, 60%, 70%, 80%, and 90%, respectively.

The number of frames accepted by Montpelier-Milano access link for uncompressed data at different bandwidth simulation runs are 1721, 2156, 2352, 2541, and 3110. The numbers indicate that the number of frames accepted increase linearly from one bandwidth to another. Montpelier-Milano access link dropped frames at each simulation run. The number of dropped frames at each bandwidth percentage is 213 at 50%, 215 at 60%, 271 at 70%, 316 at 80% and 347 at 90%. The drop rate ranges from 10% to 12%.

Comparison of Frames-Dropped

The drop rate of compressed and uncompressed data and their differences are shown in **Table 9.19**. Also shown in the table is the comparison of the link utilization rate for both data types.

The drop rate of CUNY-Montpelier access link at 50% bandwidth for compressed and uncompressed data is 10%. Similarly, the drop rates for the remaining bandwidth percentages are equal or close to each other. Their differences are well above the margin of error, $\pm 5\%$. The link utilization rate is less for compressed data than for uncompressed data for all bandwidths as shown in **Table 9.19**.

The drop rate of CUNY-Bonn access link at 50% bandwidth for compressed and uncompressed data is 12%. Similarly, the drop rates for the remaining bandwidth percentages, 60%-90%, differ by 1% or 2% or equal to each other. The link utilization rate is less for compressed compared to uncompressed data for all bandwidths.

The drop rate of Montpelier-Milano access link at 50% bandwidth is 11% and 12% for compressed and uncompressed data, respectively and the difference between the two is $\pm 1\%$. Similarly, the difference for the remaining bandwidth

percentages, 60%-90%, is less than or equal 2%. The link utilization rate is less for compressed compared to uncompressed data for all bandwidths.

This demonstrates that while the drop rate is more or less the same for both compressed and uncompressed data, compressing data has a clear and substantive advantage in terms of the link utilization rate. Thus, data compression allows for transmission of substantially larger volume of data in shorter intervals.

E-mail (64kbps)	WAN CLOUDS: ACCESS LINK STATS (ENTRY)									
LZW (60%)	Comparison of Compressed & Uncompressed E-mail (LZW) Data									
Message Size = 100B	Frames Accepted/Dropped/Link Utilization (%)									
Cloud: Access Link	Cmp 50%	Uncmp 50%	Cmp 60%	Uncmp 60%	Cmp 70%	Uncmp 70%	Cmp 80%	Uncmp 80%	Cmp 90%	Uncmp 90%
CUNY-Montpelier										
Frames Accepted	653	2057	664	2180	1042	2890	1061	2931	1352	2997
Frames Dropped	66	213	89	277	132	341	139	322	159	365
Drop Rate	0.1010	0.10354	0.1340	0.1270	0.12667	0.11799	0.13100	0.10986	0.11760	0.121788
Drop Rate (%)	10%	10%	13%	13%	13%	12%	13%	11%	12%	12%
Difference	0%		0%		1%		2%		0%	
Link Utilization (%)	20.13	63.54	21.08	68.81	32.87	90.46	33.60	91.07	42.31	94.14
CUNY-Milano										
Frames Accepted	661	1371	844	2081	1021	2915	1102	2709	1144	2357
Frames Dropped	85	138	95	253	116	355	112	307	142	288
Drop Rate	0.1285	0.10065	0.1125	0.1215	0.11361	0.12178	0.10163	0.11332	0.12412	0.122189
Drop Rate (%)	13%	10%	11%	12%	11%	12%	10%	11%	12%	12%
Difference	3%		-1%		-1%		-1%		0%	
Link Utilization (%)	20.89	42.27	26.29	65.36	31.84	91.55	33.99	84.44	36.01	74.07

	Frames Accepted/Dropped/Link Utilization (%) (Cont'd.)									
Cloud: Access Link	Cmp 50%	Uncmp 50%	Cmp 60%	Uncmp 60%	Cmp 70%	Uncmp 70%	Cmp 80%	Uncmp 80%	Cmp 90%	Uncmp 90%
CUNY-Bern										
Frames Accepted	775	1452	869	1723	1193	2121	902	3013	1158	2790
Frames Dropped	86	189	105	187	135	230	91	343	119	353
Drop Rate	0.1109	0.13016	0.1208	0.1085	0.11316	0.10843	0.10088	0.11384	0.10276	0.126523
Drop Rate (%)	11%	13%	12%	11%	11%	11%	10%	11%	10%	13%
Difference	-2%		1%		0%		-1%		-2%	
Link Utilization (%)	24.11	45.95	27.27	53.50	37.18	65.81	27.80	93.97	35.76	88.03
CUNY-Bonn										
Frames Accepted	726	1167	807	1874	1031	2155	1287	3075	1238	3109
Frames Dropped	84	136	106	225	105	198	150	362	147	350
Drop Rate	0.1157	0.11653	0.1313	0.1200	0.10184	0.09187	0.11655	0.11772	0.11873	0.112576
Drop Rate (%)	12%	12%	13%	12%	10%	9%	12%	12%	12%	11%
Difference	-0%		1%		1%		-0%		1%	
Link Utilization (%)	22.68	36.48	25.56	58.80	31.81	65.86	40.26	96.23	38.79	96.86
Princeton-Montpelier										
Frames Accepted	709	2059	720	2399	906	1880	1351	2741	1320	3121
Frames Dropped	73	223	86	265	112	220	149	300	127	355
Drop Rate	0.1029	0.10830	0.1194	0.1104	0.12362	0.11702	0.11028	0.10944	0.09621	0.113745
Drop Rate (%)	10%	11%	12%	11%	12%	12%	11%	11%	10%	11%
Difference	-1%		1%		1%		0%		-2%	
Link Utilization (%)	21.90	63.88	22.57	74.59	28.49	58.81	42.00	85.15	40.52	97.33

	Frames Accepted/Dropped/Link Utilization (%) (Cont'd.)									
Cloud: Access Link	Cmp 50%	Uncmp 50%	Cmp 60%	Uncmp 60%	Cmp 70%	Uncmp 70%	Cmp 80%	Uncmp 80%	Cmp 90%	Uncmp 90%
CUNY-Princeton										
Frames Accepted	633	1587	971	2483	1140	2179	775	2778	1315	3013
Frames Dropped	79	160	114	265	138	246	79	324	153	342
Drop Rate	0.1248	0.10081	0.1174	0.1067	0.12105	0.11289	0.10193	0.11663	0.11634	0.113508
Drop Rate (%)	12%	10%	12%	11%	12%	11%	10%	12%	12%	11%
Difference	2%		1%		1%		-1%		0%	
Link Utilization (%)	19.94	48.92	30.38	76.93	35.78	67.93	23.91	86.84	41.1	93.95
Montpelier-Milano										
Frames Accepted	593	1721	1024	2156	1379	2352	1032	2541	1452	3110
Frames Dropped	64	213	99	215	133	271	112	316	160	347
Drop Rate	0.1079	0.12376	0.0966	0.0997	0.09644	0.11522	0.10852	0.12436	0.11019	0.111575
Drop Rate (%)	11%	12%	10%	10%	10%	12%	11%	12%	11%	11%
Difference	-2%		-0%		-2%		-2%		-0%	
Link Utilization (%)	18.40	54.14	31.43	66.36	42.34	73.47	32.03	80.01	45.14	96.80

Table 9. 19: E-mail (LZW) - WAN Cloud: Access Link Statistics: Frames Accepted, Frames Dropped & Link Utilization

Message & Response Sources

Table 9.20 shows data for 6 distinct origin nodes that send messages to 6 different destination nodes. Of these, only two nodes are chosen for analysis that follows:

CUNY-Milano/CUNY-Princeton (Origin/Destination)

Compressed Data

CUNY-Milano, origin node, created 746, 939, 1137, 1214, and 1286 to send to CUNY-Princeton, destination node, at 50%, 60%, 70%, 80%, and 90% bandwidth, respectively. The number of packets created increases linearly from one bandwidth to another. The number of packets delivered at the destination is 431, 410, 336, 477, and 400 at each of the five simulation runs. These numbers show a fluctuation. And the number of packets dropped is 133, 155, 162, 179, and 185 at each of the five simulation runs. Again, this shows that drop is linear like that of packets created. The average packet delay (in milliseconds) at 80% and 90% is less than the average delay at 50%-70% bandwidths.

Uncompressed Data

CUNY-Milano, origin node, created 1515, 2188, 3178, 3164, and 3483 to send to CUNY-Princeton, destination node, at 50%, 60%, 70%, 80%, and 90%

bandwidth, respectively. The number of packets created increases from one bandwidth to another linearly from 50% bandwidth to 70% bandwidths, decreases at 80% by 0.4% and again increases by 10% at 90%. The number of packets delivered at the destination is 397, 531, 521, 463, and 373 at each of the five simulation runs. These numbers show a decrease from 531 packets delivered at 60% to 521 packets delivered at 70% and the number of packets delivered further decreases at 80% and 90% bandwidth for simulation. And the number of packets dropped is 178, 311, 415, 345, and 329 at each of the five simulation runs. The average packet delay (in milliseconds) increases from 50% bandwidth to 60% bandwidth, decreases at 70%, increases at 80% and again decreases at 90% bandwidth.

Comparison of Compressed and Uncompressed Data

The number of packets created for compressed data is lower than for uncompressed data for all bandwidths. These statistics are given in **Table 9.20**. The number packets delivered for compressed data is more for 50% (431/397), 80% (477/463) and 90% (400/463) bandwidths and less for the remaining bandwidths than for uncompressed data. The number of packets dropped similarly is less for compressed than for uncompressed data for all simulation runs. The ratio of packets dropped over packets delivered (packets

dropped/packets delivered) is 31%, 38%, 48%, 38%, and 46% for compressed data and it is 45%, 59%, 80%, 75%, and 88% for uncompressed data. The high percentages at 70%-90% bandwidths for uncompressed data is due to the increase in packets dropped. The average packet delay (in milliseconds) for compressed data increases from 50% to 70% and then decreases from 80% to 90% and for uncompressed data it increases from 50% to 60%, decreases at 70%, increases at 80% and again decreases at 90% bandwidth.

Microsoft Glitch!

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E-mail (64kbps)	Message & Response Sources: Packets Delivered/Packet Delay									
LZW (60%)	Compressed Data					Uncompressed Data				
Message Size = 100B	Bandwidth Used For Simulation									
Origin/Destination	50%	60%	70%	80%	90%	50%	60%	70%	80%	90%
CUNY-Milano/ CUNY-Princeton										
Packets Created	746	939	1137	1214	1286	1515	2188	3178	3164	3483
Packets Delivered	431	410	336	477	400	397	531	521	463	373
Packets Dropped	133	155	162	179	185	178	311	415	345	329
Dropped/Created	18%	17%	14%	15%	14%	12%	14%	13%	11%	9%
Dropped/Delivered	31%	38%	48%	38%	46%	45%	59%	80%	75%	88%
Packet Delay (MS)										
Average	10263.496	14471.157	21924.706	20033.300	19122.248	22330.287	30544.959	28621.252	28872.840	24603.479
Maximum	15990.503	23568.503	33962.503	34310.503	37920.503	41204.503	49976.503	49920.503	49962.503	46680.503
MontpelierP/ Montpelier-Milano										
Packets Created	782	806	995	1500	1447	2197	2664	2118	3014	3867
Packets Delivered	520	335	420	615	556	612	434	394	490	520
Packets Dropped	126	116	152	214	182	303	303	280	351	435
Dropped/Created	16%	14%	15%	14%	13%	14%	11%	13%	12%	11%
Dropped/Delivered	24%	35%	36%	35%	33%	50%	70%	71%	72%	84%
Packet Delay (MS)										
Average	9623.229	13590.306	17820.916	19792.044	21053.865	27942.616	25532.075	26877.438	27956.487	31580.186
Maximum	16194.002	23124.002	33474.002	33978.002	37992.002	45964.002	47992.002	46936.002	48922.002	52964.002

	Message & Response Sources: Packets Delivered/Packet Delay (Cont'd.)									
E-mail (64kbps)	Compressed Data					Uncompressed Data				
LZW (60%)	Bandwidth Used For Simulation									
Origin/Destination	50%	60%	70%	80%	90%	50%	60%	70%	80%	90%
CUNY-Bern/ Bern-Bonn										
Packets Created	861	974	1328	993	1277	1641	1939	2468	3381	3573
Packets Delivered	501	490	490	429	505	426	225	411	401	267
Packets Dropped	156	161	175	145	182	244	213	277	395	378
Dropped/Created	18%	17%	13%	15%	14%	15%	11%	11%	12%	11%
Dropped/Delivered	31%	33%	36%	34%	36%	57%	95%	67%	99%	142%
Packet Delay (MS)										
Average	9605.175	11831.189	19761.826	20600.337	23722.726	22947.757	24652.286	30012.483	30004.575	26497.088
Maximum	16264.002	23544.002	33922.002	33950.002	37950.002	42900.002	46936.002	46978.002	49978.002	47256.002
CUNY-Bonn/ Milano-Bonn										
Packets Created	810	913	1136	1487	1386	1303	2102	2124	3475	3384
Packets Delivered	448	485	326	512	243	253	347	603	527	519
Packets Dropped	147	164	145	197	185	161	262	262	422	419
Dropped/Created	18%	18%	13%	13%	13%	12%	12%	12%	12%	12%
Dropped/Delivered	33%	34%	44%	38%	76%	64%	76%	43%	80%	81%
Packet Delay (MS)										
Average	9744.872	13693.663	20964.922	20548.502	24239.870	19326.784	30056.405	29382.104	30057.273	31262.441
Maximum	16292.002	23676.002	33978.002	34362.002	37964.002	41388.002	47390.002	50922.002	52992.002	53936.002

	Message & Response Sources: Packets Delivered/Packet Delay (Cont'd.)									
E-mail (64kbps)	Compressed Data					Uncompressed Data				
LZW (60%)	Bandwidth Used For Simulation									
Origin/Destination	50%	60%	70%	80%	90%	50%	60%	70%	80%	90%
CUNY-Montpelier/ Montpelier-Milano										
Packets Created	719	753	1174	1200	1511	2183	2324	3388	4019	3177
Packets Delivered	490	343	479	381	588	595	590	530	318	544
Packets Dropped	114	135	187	189	235	286	358	400	368	420
Dropped/Created	16%	18%	16%	16%	16%	13%	15%	12%	9%	13%
Dropped/Delivered	23%	39%	39%	50%	40%	48%	61%	75%	116%	77%
Packet Delay (MS)										
Average	9949.655	13650.142	19960.92	24103.87	24779.284	26360.600	29519.463	28544.605	30580.052	30885.248
Maximum	15936.002	23558.002	33880.002	34472.002	39978.002	43142.002	49908.002	48782.002	48936.002	52978.002
CUNYPrinceton/ Princeton-Mont										
Packets Created	712	1085	1278	854	1468	1747	2714	2441	2918	3399
Packets Delivered	452	505	633	323	388	527	485	359	528	520
Packets Dropped	127	165	206	113	191	218	307	294	369	400
Dropped/Created	18%	15%	16%	13%	13%	12%	11%	12%	13%	12%
Dropped/Delivered	28%	33%	33%	35%	49%	41%	63%	82%	70%	77%
Packet Delay (MS)										
Average	9692.086	13111.285	19658.814	19663.06	24605.656	26516.745	28486.175	24852.620	29751.767	29513.84
Maximum	16138.002	22978.002	33886.002	33754.002	37754.002	43936.002	47880.002	46950.002	51796.002	49992.002

Table 9. 20: E-mail (LZW) - Message & Response Sources: Packets Delivered/Delay

MontpelierP/Montpelier-Milano (Origin/Destination)

Compressed Data

MontpelierP, origin node, created 782, 806, 995, 1500, and 1447 to send to Montpelier-Milano, destination node, at 50%, 60%, 70%, 80%, and 90% bandwidths, respectively. The number of packets delivered at the destination is 520, 335, 420, 615, and 556 at the five simulation runs, respectively. And the number of packets dropped similarly is 126, 116, 152, 214, and 182. The number of packets created increases linearly from 50% to 80% and decreases at 90%. But, the number of packets delivered and dropped oscillates between 50% and 90% bandwidths. The average packet delay (in milliseconds) increases linearly.

Uncompressed Data

MontpelierP, origin node, created 2197, 2664, 2118, 3014, and 3867 to send to Montpelier-Milano, destination node, at 50%, 60%, 70%, 80%, and 90% bandwidths, respectively. The number of packets delivered at the destination is 612, 434, 394, 490, and 520 at the five simulation runs, respectively. And the number of packets dropped similarly is 303, 303, 280, 351, and 435. The number of packets created from 50% to 60% bandwidths increases, decreases at 70%, and again increases at 80% and 90% bandwidths. The ratio of packets dropped over packets delivered (packets dropped/packets delivered) is 50%, 70%, 71%, 72%, and 84% at 50%-90% bandwidths. The average packet delay (in milliseconds)

decreases at 60% bandwidth from 50% and increases linearly from 70% to 90% bandwidths.

Comparison of Compressed and Uncompressed Data

The number of packets created for compressed data is less than that of uncompressed data for all 5 bandwidth simulation runs, a function of compression. The number of packets dropped for compressed data is also less for all simulation runs than for uncompressed data. The ratio of packets dropped over packets delivered is 24%, 35%, 36%, 35%, and 33% for compressed data and it is 50%, 70%, 71%, 72%, and 84% for uncompressed data.

9.2. Audio

Sound is another rich arena for compression. Music is a commodity sold on digital plastic disks by the recording industry. CD-quality audio data is very suitable for compression. The audio signal for each stereo channel is sampled at 44.1kHz, and each sample is represented by 16 bits. This means that the amount of data stored on one CD is enormous. If we want to transmit this data, the amount of channel capacity required would be significant. Compression is definitely useful in this case. MPEG-1 with two simulation runs is used to simulate audio application.

The compression rate of MPEG-1 is 50% and 75% for Simulation Run1 and Simulation Run2, respectively. The message size in each run is 200 bytes, warmup length (specifies the time in the simulation when the application will begin to collect data) is 10 seconds and replication length (the number of seconds of simulated time during which statistics are collected for each report) is 50 seconds. The bandwidth used for both simulation runs is 128kbps, twice that of e-mail application. Just as in e-mail application, the number of observations in an interval is Poisson distributed. And the parameter entered for the Poisson distribution is the interarrival time (LAT, the time from the start of one message to

the start of the next). The total number of simulation runs for both are 20. Each simulation ran for approximately 1 to 2 minutes.

The traffic intensity is rather high for Simulation Run1. The drop rate is high because the network is saturated. Compressed data from Simulation Run1 is used as uncompressed data for Simulation Run2 and further reduced it by 50% for compressed data for this run to see the effect on link utilization rate and drop rate (**Tables 9.29** and **9.30**).

MPEG-1

The simulation runs statistics for compressed and uncompressed data for both simulations are shown in **Tables 9.21** and **9.22**, respectively.

Examination of data in **Tables 9.21** and **9.22** show that as the bandwidth increases:

1. IAT decreases
2. Packets/Second increase (see **Figures 9.9** and **9.10**)
3. Bits/Second increase
4. Simulation run time is the same for bandwidth runs and the message size used for Video is small.

5. The rate of transmission for compressed data at each bandwidth is less than for uncompressed data.

as was observed in E-mail application. As the tables show, when LAT decreases, the number of packets/second increases linearly in both data types. This is depicted in **Figures 9.9 and 9.10**.

Bandwidth: 128kbps, Replication Length: 50s					
Message Size (bytes): 200 = 1600 bits					
MPEG-1 (50%)	Bandwidth Used for Simulation				
	50%	60%	70%	80%	90%
SimRun1 -InterArrival Time(IAT)	0.05	0.042	0.036	0.032	0.028
SimRun2 -InterArrival Time(IAT)	0.1	0.08	0.07	0.0625	0.06
SimRun1- Packets/Second(pps)	20	24	28	32	36
SimRun2- Packets/Second(pps)	10	12	14	16	18
SimRun1- Bits/Second(bps)	32000	38400	44800	51200	57600
SimRun2- Bits/Second(bps)	16000	19200	22400	25600	28800
SimRun1- Simulation Time (min)	2	2	2	3	3
SimRun2- Simulation Time (min)	1	1	1	1	1

Table 9. 21: Audio - Simulation Run Statistics for Compressed Data

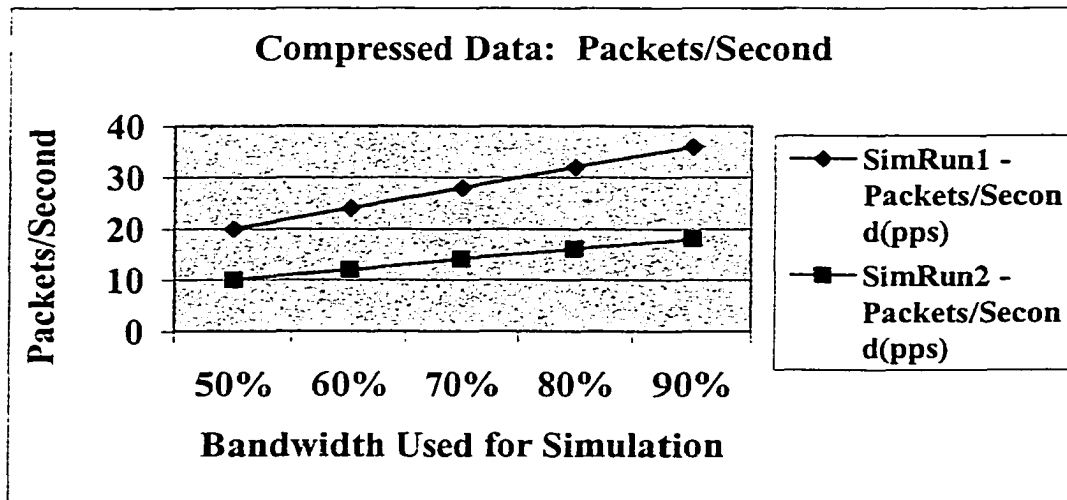


Figure 9. 9: Audio - Compressed Data: Packets/Second

Bandwidth: 128kbps, Replication Length: 50s					
Message Size (bytes): 200 = 1600 bits					
MPEG-1 (50%)	Bandwidth Used for Simulation				
	50%	60%	70%	80%	90%
SimRun1 - InterArrival Time(IAT)	0.025	0.021	0.018	0.016	0.014
SimRun2 - InterArrival Time(IAT)	0.05	0.042	0.036	0.032	0.028
SimRun1 - Packets/Second(pps)	40	48	56	64	72
SimRun2 - Packets/Second(pps)	20	24	28	32	36
SimRun1 - Bits/Second(bps)	64000	76800	89600	102400	115200
SimRun2 - Bits/Second(bps)	32000	38400	44800	51200	57600
SimRun1 - Simulation Time (min)	2	2	3	3	3
SimRun2 - Simulation Time (min)	2	2	2	3	3

Table 9. 22: Audio - Simulation Run Statistics for Uncompressed Data

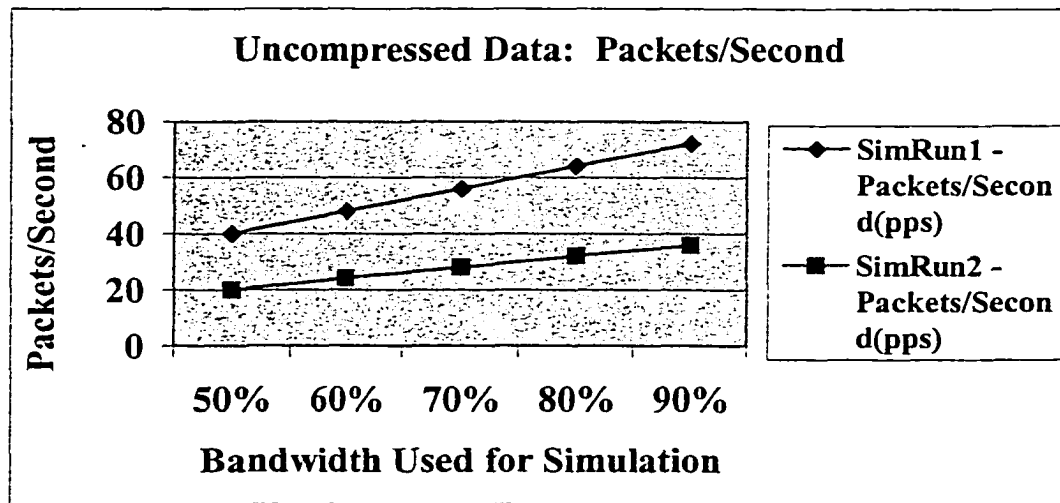


Figure 9. 10: Audio - Uncompressed Data: Packets/Second Analysis

COMNET III generates a statistical report of Link Utilization for various processing nodes as shown in **Tables 9.23** and **9.24** for compressed and uncompressed data for simulation runs 1 and 2, respectively. The number of frames delivered is equal to the number of pockets delivered from each link for audio simulation.

First row in **Tables 9.23** and **9.24** represents packets delivered during the simulation. They fluctuate for some links and increase as the bandwidth increases for others. This fluctuation could be a function of network delay and/or link/node failure.

Fluctuation in:

- Packets/Second (Packets Delivered/Replication Length, 50s);
- Bytes Delivered (Packets Delivered * Message Size);
- KBPS (kilobits/second, Packets/Second * Message Size * 8bits); and
- Link Utilization

can be seen as reflections of the fluctuations in packets delivered.

Comparison of the results shown in **Tables 9.23** and **9.24** demonstrates that the utilization rate for compressed data is consistently lower than for uncompressed data. This implies that compression permits transmission of larger volume of data in shorter intervals.

Compressed Data

- **Simulation Run 1**

The number of packets delivered from CUNY to Milano is 965, 1057, 1390, 1538, and 2195 at 50%, 60%, 70%, 80%, and 90% bandwidths, respectively. From 50% to 60% we see a 10% increase in frames/packets delivered. From 60% to 70%, the increase is 31%; from 70% to 80%, it is 10%; and from 80% to 90%, it is nearly 43%. The increase is upward in each category, packets/second, bytes delivered, kilobits per second (kbps), and link utilization percentage. The numbers in the table indicate that from CUNY to Princeton link is a linear curve. But, for the remaining two links, from Montpelier to Princeton and from Milano to Montpelier the number of packets delivered gradually increases from 50% bandwidth to 70% bandwidth and decreases at 80% by 5% in case of the former link and by 14% in the latter case. The same holds for the other categories.

- **Simulation Run 2**

The number of packets delivered from CUNY to Milano is 632, 721, 643, 1538, and 754 at 50%, 60%, 70%, 80%, and 90% bandwidths, respectively. From 50% to 60% and 60% to 70% we see a 13% increase in frames/packets delivered. However, from 70% to 80% the number frames/packets delivered decreases by 11% and again increases from 80% to 90% by 17%. The increase is upward in

each category, packets/second, bytes delivered, kilobits per second (kbps), and link utilization percentage from 50% to 70%, downward at 80%, and again upward at 90% bandwidths. The numbers in the table show the same pattern to CUNY to Princeton link, i.e., increase from 50% to 70%, decrease at 80%, and again increase at 90% bandwidth for each category. However, for Montpelier to Princeton link the number frames/packets delivered increases from 50% to 80% and decreases at 90% bandwidth. And for Milano to Montpelier the number of packets delivered increases from 50% to 60% bandwidth, decreases at 70% by 6% and again increases from 70% to 90% bandwidths. The same holds for the other categories.

Uncompressed Data

- **Simulation Run 1**

In case of uncompressed data, the number of packets delivered from CUNY to Milano is a steady upward increase, 1553, 2323, 2711, and 3549 at 50%, 60%, 70%, 80% bandwidths, respectively, but decreases from 80% bandwidth to 90% bandwidth by 6%. From 50% to 60% we see a 50% increase in frames/packets delivered. From 60% to 70%, the increase is 17%; from 70% to 80%, it is 31%. The slope of number of packets delivered is upward in each category, packets/second, bytes delivered, kilobits per second (kbps), and link utilization percentage for 50%-80% bandwidths and for 90% bandwidth it is

downward. The numbers in the table indicate that the slopes of number of packets delivered for the other three links is nonlinear. In case of CUNY to Princeton link, the number of packets delivered goes down by 9% at 70% bandwidth. From Montpelier to Princeton link, the number of packets delivered decreases by 14% at 80% bandwidth and from Milano to Montpelier link the decrease is by 1% at 60% bandwidth. These changes are also reflected in other categories, packets/second, bytes delivered, kbps delivered, and utilization percentage.

- **Simulation Run 2**

The data analysis presented in Simulation Run 1 for compressed data applies to uncompressed data in Simulation Run 2.

Application: Audio (128kbps), MPEG-1 (50%) - Simulation Run 1										
Message Size = 200bytes, Replication Length = 50s										
Links: Channel Utilization/Utilization by Application										
	Compressed Data					Uncompressed Data				
	Bandwidth Used For Simulation									
Link	50%	60%	70%	80%	90%	50%	60%	70%	80%	90%
From CUNY-Milano										
Frames/Packets Delivered	965	1057	1390	1538	2195	1553	2323	2711	3549	3342
Packets/second	19.300	21.140	27.800	30.760	43.900	31.060	46.460	54.220	70.980	66.840
Bytes Delivered	193000	211400	278000	307600	439000	310600	464600	542200	709800	668400
KBPS Delivered	30.88	33.824	44.840	49.216	70.240	49.696	74.336	86.752	113.568	106.944
Utilization (%)	24.13	26.43	34.75	38.48	54.88	38.83	58.08	67.80	88.75	83.58
From CUNYPrincet										
Frames/Packets Delivered	1040	1185	1455	1509	1683	1999	2500	2277	2725	3383
Packets/second	20.800	23.700	29.100	30.180	33.660	39.980	50.000	45.540	54.500	67.660
Bytes Delivered	208000	237000	291000	301800	336600	399800	500000	455400	545000	676600
KBPS Delivered	33.280	37.920	46.560	48.288	53.856	63.968	80.000	72.864	87.2	108.256
Utilization (%)	26.00	29.63	36.38	37.73	42.10	49.98	62.53	56.93	68.15	84.58

Links: Channel Utilization/Utilization by Application (Cont'd)										
Audio (128kbps)	Compressed Data					Uncompressed Data				
MPEG-1 (50%)	Bandwidth Used For Simulation									
Link	50%	60%	70%	80%	90%	50%	60%	70%	80%	90%
From MontpelierPri										
Frames/Packets Delivered	985	1121	1360	1297	1781	2160	2298	2871	2468	2957
Packets/second	19.700	22.420	27.200	25.940	35.620	43.200	45.960	57.420	49.360	59.140
Bytes Delivered	197000	224200	272000	259400	356200	432000	459600	574200	493600	591400
KBPS Delivered	31.52	35.872	43.520	41.504	56.992	69.120	73.536	91.872	78.976	94.624
Utilization (%)	24.63	28.03	34.00	32.43	44.55	54.00	57.45	71.80	61.73	73.93
From Milano-Mont										
Frames/Packets Delivered	879	1127	1355	1170	1471	2124	2099	2138	2915	3480
Packets/second	17.580	22.540	27.100	23.400	29.420	42.480	41.980	42.760	58.300	69.600
Bytes Delivered	175800	225400	271000	234000	294200	424800	419800	427600	583000	696000
KBPS Delivered	28.128	36.064	43.360	37.440	47.072	67.968	67.168	68.416	93.28	111.36
Utilization (%)	21.98	28.18	33.88	29.25	36.78	53.12	52.48	53.45	72.85	87.00

Table 9. 23: Audio SimRun1 - Links: Channel Utilization/Utilization by Application

Audio (128kbps)	Links: Channel Utilization/Utilization by Application									
MPEG-1 (75%)	Compressed Data					Uncompressed Data				
MsgSize = 200B	Bandwidth Used For Simulation									
Link	50%	60%	70%	80%	90%	50%	60%	70%	80%	90%
From CUNY-Milano										
Frames/Packets Delivered	562	637	721	643	754	965	1057	1390	1538	2195
Packets/second	11.24	12.74	14.42	12.86	15.08	19.300	21.140	27.800	30.760	43.900
Bytes Delivered	112400	127400	144200	128600	150800	193000	211400	278000	307600	439000
KBPS Delivered	17.984	20.384	23.072	20.576	24.128	30.88	33.824	44.840	49.216	70.240
Utilization (%)	14.05	15.983	18.03	16.08	18.85	24.13	26.43	34.75	38.48	54.88
From CUNYPrincet										
Frames/Packets Delivered	538	584	658	624	860	1040	1185	1455	1509	1683
Packets/second	10.76	11.68	13.16	12.48	17.2	20.800	23.700	29.100	30.180	33.660
Bytes Delivered	107600	116800	131600	124800	172000	208000	237000	291000	301800	336600
KBPS Delivered	17.216	18.688	21.056	19.968	24.128	33.280	37.920	46.560	48.288	53.856
Utilization (%)	13.45	14.6	16.45	15.6	18.85	26.00	29.63	36.38	37.73	42.10

Audio (128kbps)	Links: Channel Utilization/Utilization by Application (Cont'd.)									
MPEG-1 (75%)	Compressed Data					Uncompressed Data				
MsgSize = 200B	Bandwidth Used For Simulation									
Link	50%	60%	70%	80%	90%	50%	60%	70%	80%	90%
From MontpelierPri										
Frames/Packets Delivered	516	589	627	993	854	985	1121	1360	1297	1781
Packets/second	10.32	11.78	12.54	19.86	17.08	19.700	22.420	27.200	25.940	35.620
Bytes Delivered	103200	117800	125400	198600	170800	197000	224200	272000	259400	356200
KBPS Delivered	16.512	18.848	20.064	31.776	27.328	31.52	35.872	43.520	41.504	56.992
Utilization (%)	12.9	14.73	15.68	24.83	21.35	24.63	28.03	34.00	32.43	44.55
From Milano-Mont										
Frames/Packets Delivered	436	603	568	857	869	879	1127	1355	1170	1471
Packets/second	8.72	12.06	11.36	17.14	17.38	17.580	22.540	27.100	23.400	29.420
Bytes Delivered	87200	120600	113600	171400	173800	175800	225400	271000	234000	294200
KBPS Delivered	13.952	19.296	18.176	27.424	27.808	28.128	36.064	43.360	37.440	47.072
Utilization (%)	10.9	15.08	14.2	21.43	21.73	21.98	28.18	33.88	29.25	36.78

Table 9. 24: Audio SimRun2 - Links: Channel Utilization/Utilization by Application

WAN Clouds: Access Link Statistics Analysis

Tables 9.25, 9.26, 9.27 and 9.28 present statistics for 7 access links: (1) CUNY-Montpelier, (2) CUNY-Milano, (3) CUNY-Bern, (4) CUNY-Bonn, (5) Princeton-Montpelier, (6) CUNY-Princeton, and (7) Montpelier-Milano. Only three of these links are chosen for analysis to demonstrate the pattern of variation over access links.

Compressed Data	Frames Accepted				
Access Link	50%	60%	70%	80%	90%
CUNY-Montpelier					
Simulation Run 1	981	900	1295	1294	1408
Simulation Run 2	509	548	615	518	660
CUNY-Milano					
Simulation Run 1	885	976	1250	1283	1577
Simulation Run 2	511	574	651	590	658
CUNY-Bern					
Simulation Run 1	913	993	1052	1448	1614
Simulation Run 2	357	540	664	702	809
CUNY-Bonn					
Simulation Run 1	688	922	1462	1313	1298
Simulation Run 2	441	561	588	734	587
Princeton-Montpelier					
Simulation Run 1	878	1028	1175	1145	1447
Simulation Run 2	467	533	574	909	765
CUNY-Princeton					
Simulation Run 1	921	1035	1201	1233	1441
Simulation Run 2	487	520	600	572	797
Montpelier-Milano					
Simulation Run 1	805	1042	1186	1104	1276
Simulation Run 2	387	540	533	764	776

Table 9. 25: Audio - WAN Cloud Access Link Statistics: Frames Accepted (Compressed Data)

Compressed Data	Frames Dropped				
Access Link	50%	60%	70%	80%	90%
CUNY-Montpelier					
Simulation Run 1	101	76	242	252	350
Simulation Run 2	59	50	62	66	75
CUNY-Milano					
Simulation Run 1	80	81	140	250	641
Simulation Run 2	51	63	70	53	96
CUNY-Bern					
Simulation Run 1	87	115	138	512	541
Simulation Run 2	37	56	65	70	89
CUNY-Bonn					
Simulation Run 1	79	117	281	210	249
Simulation Run 2	53	51	66	70	55
Princeton-Montpelier					
Simulation Run 1	107	93	185	152	329
Simulation Run 2	49	56	53	84	89
CUNY-Princeton					
Simulation Run 1	119	150	254	276	248
Simulation Run 2	51	65	58	52	63
Montpelier-Milano					
Simulation Run 1	74	85	169	66	195
Simulation Run 2	49	63	35	93	93

Table 9. 26: Audio - WAN Cloud Access Link Statistics: Frames Dropped (Compressed Data)

Uncompressed Data	Frames Accepted				
Access Link	50%	60%	70%	80%	90%
CUNY-Montpelier					
Simulation Run 1	1571	1550	1841	1759	1821
Simulation Run 2	981	900	1295	1294	1408
CUNY-Milano					
Simulation Run 1	1296	1634	1730	1859	1860
Simulation Run 2	885	976	1250	1283	1577
CUNY-Bern					
Simulation Run 1	1439	1763	1779	1714	1833
Simulation Run 2	913	993	1052	1448	1614
CUNY-Bonn					
Simulation Run 1	1347	1588	1746	1847	1865
Simulation Run 2	688	922	1462	1313	1298
Princeton-Montpelier					
Simulation Run 1	1566	1621	1751	1660	1676
Simulation Run 2	878	1028	1175	1145	1447
CUNY-Princeton					
Simulation Run 1	1507	1731	1636	1744	1867
Simulation Run 2	921	1035	1201	1233	1441
Montpelier-Milano					
Simulation Run 1	1601	1599	1534	1778	1851
Simulation Run 2	805	1042	1186	1104	1276

Table 9. 27: Audio - WAN Cloud Access Link Statistics: Frames Accepted (Uncompressed Data)

Uncompressed Data	Frames Dropped				
Access Link	50%	60%	70%	80%	90%
CUNY-Montpelier					
Simulation Run 1	401	611	1321	1003	1090
Simulation Run 2	101	76	242	252	350
CUNY-Milano					
Simulation Run 1	257	727	982	1682	1505
Simulation Run 2	80	81	140	250	641
CUNY-Bern					
Simulation Run 1	405	840	859	1277	1358
Simulation Run 2	87	115	138	512	541
CUNY-Bonn					
Simulation Run 1	327	691	732	1386	1381
Simulation Run 2	79	117	281	210	249
Princeton-Montpelier					
Simulation Run 1	594	677	1074	827	1319
Simulation Run 2	107	93	185	152	329
CUNY-Princeton					
Simulation Run 1	503	759	641	945	1482
Simulation Run 2	119	150	254	276	248
Montpelier-Milano					
Simulation Run 1	518	500	604	1164	1584
Simulation Run 2	74	85	169	66	195

Table 9. 28: Audio - WAN Cloud Access Link Statistics: Frames Dropped (Uncompressed Data)

Compressed Data

- **Simulation Run 1**

CUNY-Montpelier access link accepted 981 frames at 50% and 900 frames at 60%, a decrease of 8 percent. At 70% the number of frames accepted by the link went up to 1295 frames, an increase of 44%. The number of frames accepted at 80% differ from 70% by 1 frame and at 90% the frames accepted are 1408, an increase of 9%. CUNY-Montpelier access link dropped frames at each simulation run. The number of dropped frames at each bandwidth percentage is 101 at 50%, 76 at 60%, 242 at 70%, 252 at 80% and 350 at 90%. The drop rate is 10%, 8%, 19%, 19%, and 25% for all five different bandwidth simulation runs, 50%, 60%, 70%, 80%, and 90%, respectively. **Figures 9.11** and **9.12** depict the number of frames accepted and dropped, respectively.

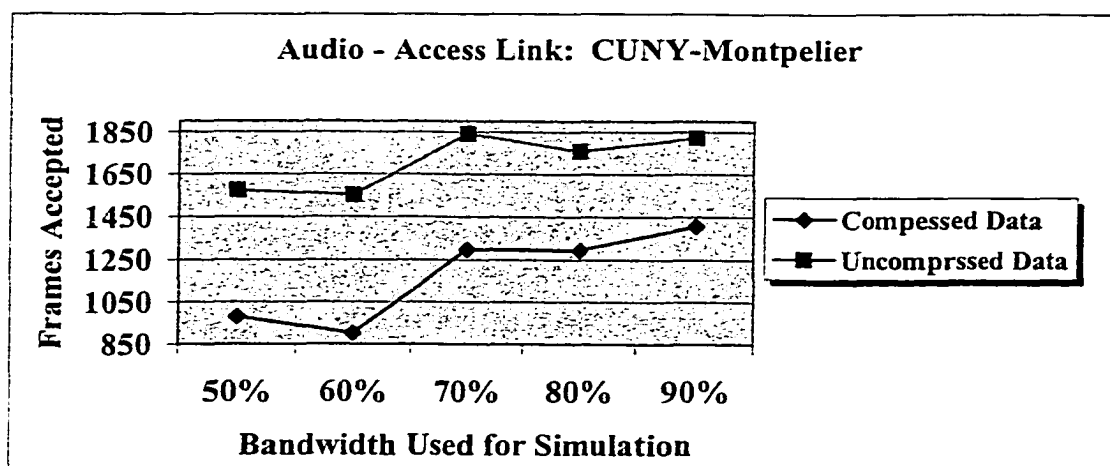


Figure 9. 11: Audio (SimRun1) – Selected Access Link: CUNY-Montpelier (Frames Accepted)

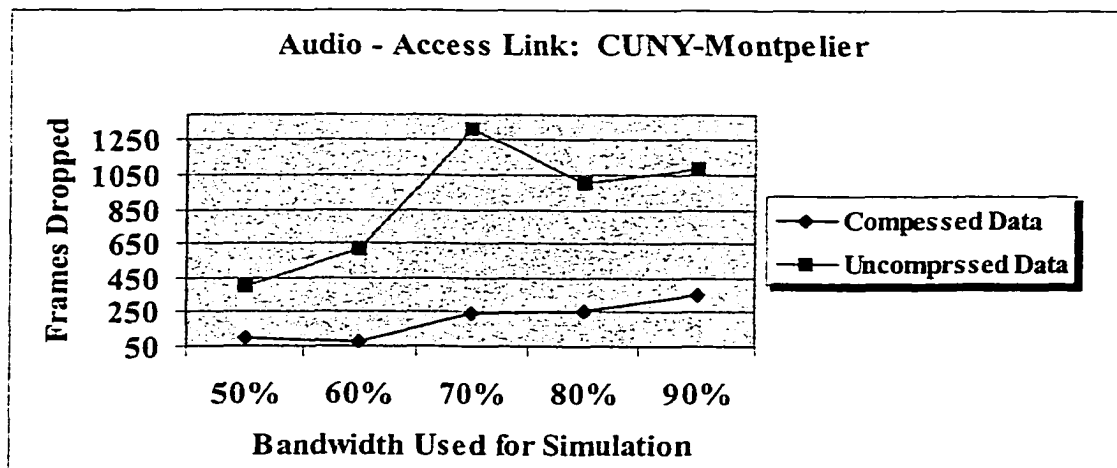


Figure 9. 12: Audio (SimRun1) – Selected Access Link: CUNY-Montpelier (Frames Dropped)

CUNY-Bonn access link accepted 688 frames at 50% and 922 frames at 60%, an increase of 34 percent. At 70% the number of frames accepted by the link went up to 1462, an increase of 59%. The number of frames accepted at 80% went down to 1313, 10% decrease, and at 90% the frames accepted are 1298, a decrease of 1%. CUNY-Bonn access link dropped frames at each simulation run. The number of dropped frames at each bandwidth percentage is 79 at 50%, 117 at 60%, 281 at 70%, 210 at 80% and 249 at 90%. The drop rate is 11%, 13%, 19%, 16%, and 19% for all five different bandwidth simulation runs, 50%, 60%, 70%, 80%, and 90%, respectively.

The number of frames accepted by Montpelier-Milano access link for compressed data at different bandwidth simulation runs are 805, 1042, 1186, 1104, and 1276. The numbers indicate that the increase from one percentage bandwidth to another is not linear. Montpelier-Milano access link dropped frames at each simulation run. The number of dropped frames at each bandwidth percentage is 74 at 50%, 85 at 60%, 169 at 70%, 66 at 80% and 195 at 90%. The drop rate ranges from 6% to 15%.

- **Simulation Run 2**

CUNY-Montpelier access link accepted 509 frames at 50% and 548 frames at 60%, an increase of 8 percent. At 70% the number of frames accepted

by the link went up to 615 frames, an increase of 12%. The number of frames accepted at 80% went down to 518 from 70%, a decrease of nearly 16%, and at 90% the frames accepted are 660, an increase of 27%, as shown in **Figure 9.13**. CUNY-Montpelier access link dropped frames at each simulation run, as shown in **Table 9.26**. The number of dropped frames at each bandwidth percentage is 59 at 50%, 50 at 60%, 62 at 70%, 66 at 80% and 75 at 90%, as shown in **Figure 9.14**. The drop rate is 12%, 9%, 10%, 13%, and 25% for all five different bandwidth simulation runs, respectively.

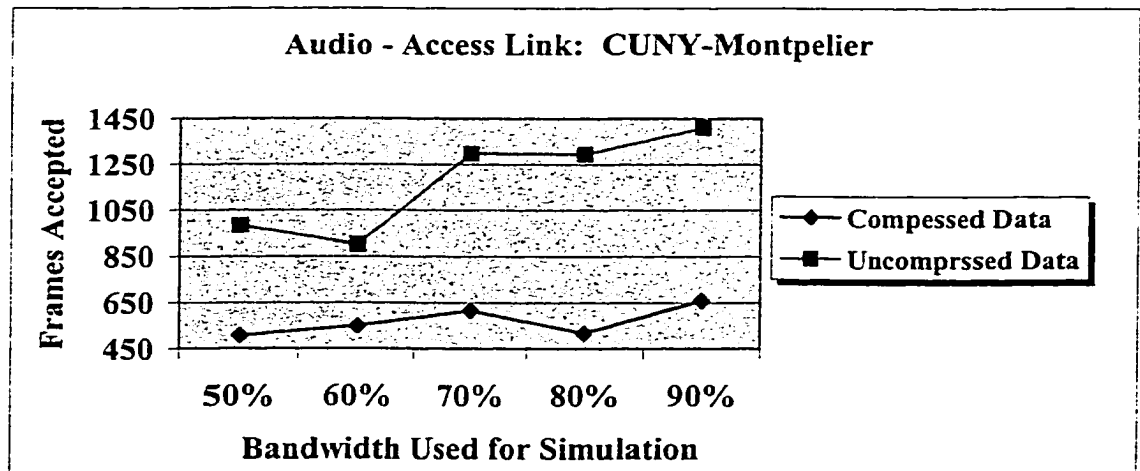


Figure 9. 13: Audio (SimRun2) – Selected Access Link: CUNY-Montpelier (Frames Accepted)

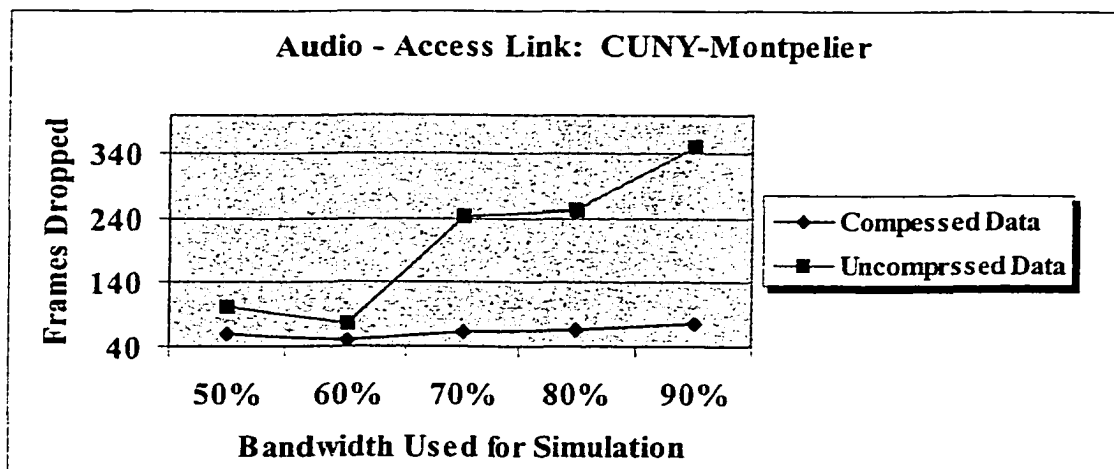


Figure 9. 14: Audio (SimRun2) – Selected Access Link: CUNY-Montpelier (Frames Dropped)

CUNY-Bonn access link accepted 441 frames at 50% and 561 frames at 60%, an increase of 27 percent. At 70% the number of frames accepted by the link went up to 588, an increase of 5%. The number of frames accepted at 80% is 734, 25% increase, and at 90% the frames accepted are 587, a decrease of 20%. CUNY-Bonn access link dropped frames at each simulation run. The number of dropped frames at each bandwidth percentage is 53 at 50%, 51 at 60%, 66 at 70%, 70 at 80% and 55 at 90%. The drop rate is 12%, 9%, 11%, 10%, and 9% for all five different bandwidth simulation runs, respectively.

The number of frames accepted by Montpelier-Milano access link for compressed data at different bandwidth simulation runs are 387, 540, 533, 764, and 776. The numbers indicate that the increase from one percentage bandwidth to another is not linear. Montpelier-Milano access link dropped frames at each simulation run. The number of dropped frames at each bandwidth percentage is 49 at 50%, 63 at 60%, 35 at 70%, 93 at 80% and 93 at 90%. The drop rate ranges from 7% to 12%.

Uncompressed Data

- **Simulation Run 1**

The number of accepted frames for CUNY-Montpelier access link are similar to that of uncompressed data, in the sense, at 50% bandwidth the number is 1571 and we see at 60% a decrease of 21 frames or 1%. At 70% the number of frames accepted by the link went up to 1841 frames, an increase of 19%. The number of frames accepted at 80% differ from 70% by 82 frames or 4% and at 90% the frames accepted are 1821, an increase of 3%. CUNY-Montpelier access link dropped frames at each simulation run. The number of dropped frames at each bandwidth percentage is 401 at 50%, 611 at 60%, 1321 at 70%, 1003 at 80% and 1090 at 90%. The drop rate is 26%, 39%, 72%, 38%, and 60% for all five different bandwidth simulation runs, 50%, 60%, 70%, 80%, and 90%.

respectively. **Figures 9.11 and 9.12** depict the number of frames accepted and dropped, respectively.

The number of frames accepted by CUNY-Bonn access link are 1347, 1588, 1746, 1847, and 1865 at 50%, 60%, 70%, 80%, and 90% bandwidth simulation runs, respectively. These numbers indicate that the increase is upward from one simulation run to another. CUNY-Bonn access link dropped frames at each simulation run. The number of dropped frames at each bandwidth percentage is 327 at 50%, 691 at 60%, 732 at 70%, 1386 at 80% and 1381 at 90%. The drop rate is 24%, 44%, 42%, 75%, and 74% for all five different bandwidth simulation runs, 50%, 60%, 70%, 80%, and 90%, respectively.

The number of frames accepted by Montpelier-Milano access link for uncompressed data at different bandwidth simulation runs are 1601, 1599, 1534, 1778, and 1851. The numbers indicate that the number of frames accepted decreased from 50% bandwidth to 60% and 70% and then they increased from 70% to 80% and 90%. Montpelier-Milano access link dropped frames at each simulation run. The number of dropped frames at each bandwidth percentage is 518 at 50%, 500 at 60%, 604 at 70%, 1164 at 80% and 1584 at 90%. The drop rate ranges from 31% to 86%.

- **Simulation Run 2**

The data analysis presented in Simulation Run 1 for compressed data applies to uncompressed data in Simulation Run 2. **Figures 9.13** and **9.14** depict the number of frames accepted and dropped, respectively.

Comparison of Frames-Dropped

The drop rate of compressed and uncompressed data and their differences are shown in **Table 9.29** and **9.30** for simulation runs 1 and 2, respectively. Also shown in the tables is the comparison of the link utilization rate for both data types.

- **Simulation Run 1**

The drop rate of CUNY-Montpelier access link at 50% bandwidth for compressed and uncompressed data is 10% and 26%, respectively. The difference between the two is $\pm 16\%$. Similarly, the difference for the remaining bandwidth percentages, 60%-90%, is $\pm 31\%$, $\pm 53\%$, $\pm 38\%$, and 35%. These differences are well above the margin of error, $\pm 5\%$. The link utilization rate is less for compressed data at 50% (28.67%), 70% (25.84%), and 90% (40.73%) bandwidths compared to 83.78%, 73.18%, and 77.12% for uncompressed data for the same bandwidths and more than that of uncompressed data at 60% and 80% as shown in **Table 9.29**.

The drop rate of CUNY-Bonn access link at 50% bandwidth for compressed and uncompressed data is 11% and 24%, respectively. The difference between the two is $\pm 13\%$. Similarly, the difference for the remaining bandwidth percentages, 60%-90%, is $\pm 31\%$, $\pm 23\%$, $\pm 59\%$, and $\pm 55\%$. The link utilization rate is less for compressed compared to uncompressed data for all bandwidths.

The drop rate of Montpelier-Milano access link at 50% bandwidth is 9% and 32% for compressed and uncompressed data, respectively and the difference between the two is $\pm 23\%$. Similarly, the difference for the remaining bandwidth percentages, 60%-90%, is $\pm 23\%$, $\pm 25\%$, $\pm 59\%$, and $\pm 70\%$. The link utilization rate is less for compressed compared to uncompressed data for all bandwidths.

- **Simulation Run 2**

The drop rate of CUNY-Montpelier access link at 50% bandwidth for compressed and uncompressed data is 12% and 10%, respectively. The difference between the two is $\pm 2\%$. Similarly, the difference for the remaining bandwidth percentages, 60%-90%, is $\pm 1\%$, $\pm 9\%$, $\pm 7\%$, and 0%. The differences at 70% and 80% bandwidths are above the margin of error, $\pm 5\%$. The link

utilization rate is less for compressed data than for uncompressed data for all bandwidths as shown in **Table 9.30**.

The drop rate of CUNY-Bonn access link at 50% bandwidth for compressed and uncompressed data is 12% and 11%, respectively. The difference between the two is $\pm 1\%$. Similarly, the difference for the remaining bandwidth percentages, 60%-90%, is $\pm 4\%$, $\pm 8\%$, $\pm 6\%$, and $\pm 10\%$. The link utilization rate is less for compressed compared to uncompressed data for all bandwidths.

The drop rate of Montpelier-Milano access link at 50% bandwidth is 13% and 9% for compressed and uncompressed data, respectively and the difference between the two is $\pm 3\%$. Similarly, the difference for the remaining bandwidth percentages, 60%-90%, is $\pm 4\%$, $\pm 8\%$, $\pm 6\%$, and $\pm 3\%$. The link utilization rate is less for compressed compared to uncompressed data for all bandwidths.

This demonstrates that while the drop rate is more or less the same for both compressed and uncompressed data, compressing data has a clear and substantive advantage in terms of the link utilization rate. Thus, data compression allows for transmission of substantially larger volume of data in shorter intervals.

AUDIO (128KBPS)	WAN CLOUDS: ACCESS LINK STATS (ENTRY)									
MPEG-1 (50%)	Comparison of Compressed & Uncompressed Audio Data									
Message Size: 200B	Frames Accepted/Dropped/Link Utilization									
Cloud: Access Link	Cmp 50%	Uncmp 50%	Cmp 60%	Uncmp 60%	Cmp 70%	Uncmp 70%	Cmp 80%	Uncmp 80%	Cmp 90%	Uncmp 90%
CUNY-Montpelier										
Frames Accepted	981	1571	900	1550	1295	1841	1294	1759	1408	1821
Frames Dropped	101	401	76	611	242	1321	252	1003	350	1090
Drop Rate	0.1030	0.25525	0.08444	0.39419	0.18687	0.71754	0.19474	0.57021	0.24857	0.59857
Drop Rate (%)	10%	26%	8%	39%	19%	72%	19%	57%	25%	60%
Difference	-15%		-31%		-53%		-38%		-35%	
Link Utilization (%)	28.67	52.27	25.84	57.26	40.73	83.78	40.97	73.18	46.59	77.12
CUNY-Milano										
Frames Accepted	885	1296	976	1634	1250	1730	1283	1859	1577	1860
Frames Dropped	80	257	81	727	140	982	250	1682	641	1505
Drop Rate	0.0904	0.19830	0.08299	0.44492	0.112	0.56763	0.19485	0.90478	0.40646	0.80913
Drop Rate (%)	9%	20%	8%	44%	11%	57%	19%	90%	41%	81%
Difference	-11%		-36%		-46%		-71%		-40%	
Link Utilization (%)	25.57	41.15	28.01	62.54	36.83	71.87	40.64	93.83	58.77	89.16

	Frames Accepted/Dropped/Link Utilization (Cont'd.)									
Cloud: Access Link	Cmp 50%	Uncmp 50%	Cmp 60%	Uncmp 60%	Cmp 70%	Uncmp 70%	Cmp 80%	Uncmp 80%	Cmp 90%	Uncmp 90%
CUNY-Bern										
Frames Accepted	913	1439	993	1763	1052	1779	1448	1714	1614	1833
Frames Dropped	87	405	115	840	138	859	512	1277	541	1358
Drop Rate	0.0953	0.28144	0.11581	0.47646	0.13117	0.48285	0.35359	0.74504	0.33519	0.74086
Drop Rate (%)	10%	28%	12%	48%	13%	48%	35%	75%	34%	74%
Difference	-19%		-36%		-35%		-39%		-41%	
Link Utilization (%)	26.50	48.87	29.36	68.98	31.53	69.93	51.96	79.24	57.12	84.55
CUNY-Bonn										
Frames Accepted	688	1347	922	1588	1462	1746	1313	1847	1298	1865
Frames Dropped	79	327	117	691	281	732	210	1386	249	1381
Drop Rate	0.1148	0.24276	0.12689	0.43513	0.19220	0.41924	0.15993	0.75040	0.19183	0.74048
Drop Rate (%)	11%	24%	13%	44%	19%	42%	16%	75%	19%	74%
Difference	-13%		-31%		-23%		-59%		-55%	
Link Utilization (%)	20.33	44.36	27.53	60.39	46.19	65.64	40.36	85.69	41.00	86.01
Princeton-Montpelier										
Frames Accepted	878	1566	1028	1621	1175	1751	1145	1660	1447	1676
Frames Dropped	107	594	93	677	185	1074	152	827	329	1319
Drop Rate	0.1219	0.37931	0.09046	0.41764	0.15744	0.61336	0.13275	0.49819	0.22736	0.78699
Drop Rate (%)	12%	38%	9%	42%	16%	61%	13%	50%	23%	79%
Difference	-26%		-33%		-46%		-37%		-56%	
Link Utilization (%)	26.10	57.24	29.71	60.90	36.04	74.88	34.37	65.91	47.08	79.35

	Frames Accepted/Dropped/Link Utilization (Cont'd.)									
Cloud: Access Link	Cmp 50%	Uncmp 50%	Cmp 60%	Uncmp 60%	Cmp 70%	Uncmp 70%	Cmp 80%	Uncmp 80%	Cmp 90%	Uncmp 90%
CUNY-Princeton										
Frames Accepted	921	1507	1035	1731	1201	1636	1233	1744	1441	1867
Frames Dropped	119	503	150	759	254	641	276	945	248	1482
Drop Rate	0.1292	0.33377	0.14492	0.43847	0.21149	0.39180	0.22384	0.54185	0.17210	0.79378
Drop Rate (%)	13%	33%	14%	44%	21%	39%	22%	54%	17%	79%
Difference	-20%		-29%		-18%		-32%		-62%	
Link Utilization (%)	27.56	53.24	31.40	66.01	38.56	60.34	39.99	71.25	44.75	88.75
Montpelier-Milano										
Frames Accepted	805	1601	1042	1599	1186	1534	1104	1778	1276	1851
Frames Dropped	74	518	85	500	169	604	66	1164	195	1584
Drop Rate	0.0919	0.32354	0.08157	0.31269	0.14249	0.39374	0.05978	0.65466	0.15282	0.85575
Drop Rate (%)	9%	32%	8%	31%	14%	39%	6%	65%	15%	86%
Difference	-23%		-23%		-25%		-59%		-70%	
Link Utilization (%)	23.29	56.17	29.87	55.62	35.91	56.66	31.00	77.95	38.98	91.02

Table 9. 29: Audio SimRun1 - WAN Cloud: Frames Accepted, Frames Dropped & Link Utilization

AUDIO (128KBPS)	WAN CLOUDS: ACCESS LINK STATS (ENTRY)									
MPEG-1 (50%)	Comparison of Compressed & Uncompressed Data									
Message Size: 200B	Frames Accepted/Dropped/Link Utilization (%)									
Cloud: Access Link	Cmp 50%	Uncmp 50%	Cmp 60%	Uncmp 60%	Cmp 70%	Uncmp 70%	Cmp 80%	Uncmp 80%	Cmp 90%	Uncmp 90%
CUNY-Montpelier										
Frames Accepted	509	981	548	900	615	1295	518	1294	660	1408
Frames Dropped	59	101	50	76	62	242	66	252	75	350
Drop Rate	0.1159	0.1030	0.0912	0.08444	0.10081	0.18687	0.12741	0.19474	0.11364	0.24857
Drop Rate (%)	12%	10%	9%	8%	10%	19%	13%	19%	25%	25%
Difference	1%		1%		-9%		-7%		0%	
Link Utilization (%)	15.05	28.67	15.85	25.84	17.94	40.73	15.48	40.97	19.48	46.59
CUNY-Milano										
Frames Accepted	511	885	574	976	651	1250	590	1283	658	1577
Frames Dropped	51	80	63	81	70	140	53	250	96	641
Drop Rate	0.0998	0.0904	0.10975	0.08299	0.10752	0.112	0.08983	0.19485	0.14589	0.40646
Drop Rate (%)	10%	9%	11%	8%	11%	11%	9%	19%	15%	41%
Difference	1%		3%		-0%		-11%		-26%	
Link Utilization (%)	14.89	25.57	16.88	28.01	19.11	36.83	17.04	40.64	19.98	58.77

Audio (128kbps)	Frames Accepted/Dropped/Link Utilization (%) (Cont.d.)									
MPEG-1 (50%)	Cmp 50%	Uncmp 50%	Cmp 60%	Uncmp 60%	Cmp 70%	Uncmp 70%	Cmp 80%	Uncmp 80%	Cmp 90%	Uncmp 90%
CUNY-Bern										
Frames Accepted	357	913	540	993	664	1052	702	1448	809	1614
Frames Dropped	37	87	56	115	65	138	70	512	89	541
Drop Rate	0.1036	0.0953	0.10370	0.11581	0.09789	0.13117	0.09971	0.35359	0.11001	0.33519
Drop Rate (%)	10%	10%	10%	12%	10%	13%	10%	35%	11%	34%
Difference	1%		-1%		-3%		-25%		-23%	
Link Utilization (%)	10.44	26.50	15.79	29.36	19.32	31.53	20.46	51.96	23.8	57.12
CUNY-Bonn										
Frames Accepted	441	688	561	922	588	1462	734	1313	587	1298
Frames Dropped	53	79	51	117	66	281	70	210	55	249
Drop Rate	0.1202	0.1148	0.09090	0.12689	0.11224	0.19220	0.09536	0.15993	0.09369	0.19183
Drop Rate (%)	12%	11%	9%	13%	11%	19%	10%	16%	9%	19%
Difference	1%		-4%		-8%		-6%		-10%	
Link Utilization (%)	13.09	20.33	16.22	27.53	17.33	46.19	21.31	40.36	17.01	41.00
Princeton-Montpelier										
Frames Accepted	467	878	533	1028	574	1175	909	1145	765	1447
Frames Dropped	49	107	56	93	53	185	84	152	89	329
Drop Rate	0.1049	0.1219	0.10506	0.09046	0.09233	0.15744	0.09240	0.13275	0.11633	0.22736
Drop Rate (%)	10%	12%	11%	9%	9%	16%	9%	13%	12%	23%
Difference	-2%		1%		-7%		-4%		-11%	
Link Utilization (%)	13.67	26.10	15.61	29.71	16.62	36.04	26.31	34.37	22.63	47.08

Audio (128kbps)	Frames Accepted/Dropped/Link Utilization (%) (Cont.d.)									
MPEG-1 (50%)	Cmp 50%	Uncmp 50%	Cmp 60%	Uncmp 60%	Cmp 70%	Uncmp 70%	Cmp 80%	Uncmp 80%	Cmp 90%	Uncmp 90%
CUNY-Princeton										
Frames Accepted	487	921	520	1035	600	1201	572	1233	797	1441
Frames Dropped	51	119	65	150	58	254	52	276	63	248
Drop Rate	0.1047	0.1292	0.125	0.14492	0.09666	0.21149	0.09090	0.22384	0.07904	0.17210
Drop Rate (%)	10%	13%	13%	14%	10%	21%	9%	22%	8%	17%
Difference	-2%		-2%		-11%		-13%		-9%	
Link Utilization (%)	14.26	27.56	15.50	31.40	17.44	38.56	16.54	39.99	22.79	44.75
Montpelier-Milano										
Frames Accepted	387	805	540	1042	533	1186	764	1104	776	1276
Frames Dropped	49	74	63	85	35	169	93	66	93	195
Drop Rate	0.1266	0.0919	0.11666	0.08157	0.06566	0.14249	0.12172	0.05978	0.11984	0.15282
Drop Rate (%)	13%	9%	12%	8%	7%	14%	12%	6%	12%	15%
Difference	3%		4%		-8%		6%		-3%	
Link Utilization (%)	11.55	23.29	15.98	29.87	15.05	35.91	22.71	31.00	23.03	38.98

Table 9. 30: Audio SimRun2 - WAN Cloud: Frames Accepted, Frames Dropped & Link Utilization

Message & Response Sources

Tables 9.31 and 9.32 for simulation runs 1 and 2 show data for 6 distinct origin nodes that send messages to 6 different destination nodes, respectively. Of these, only two nodes are chosen for analysis that follows:

CUNY-Milano/CUNY-Princeton (Origin/Destination)

Compressed Data

- **Simulation Run 1**

CUNY-Milano, origin node, created 965, 1057, 1390, 1551, and 2195 to send to CUNY-Princeton, destination node, at 50%, 60%, 70%, 80%, and 90% bandwidth, respectively. The number of packets created increases from one bandwidth to another linearly. The number of packets delivered at the destination is 450, 545, 513, 422, and 305 at each of the five simulation runs. These numbers show a decrease at 80% and 90%. And the number of packets dropped is 121, 133, 195, 305, and 667 at each of the five simulation runs. Again, this shows that drop is linear like that of packets created. The average packet delay (in milliseconds) at 70%, 80%, and 90% is less than the average delay at 60%.

- **Simulation Run 2**

CUNY-Milano, origin node, created 562, 637, 721, 643, and 754 to send to CUNY-Princeton, destination node, at 50%, 60%, 70%, 80%, and 90% bandwidth, respectively. The number of packets created increases from 50% to 70% bandwidth linearly, decreases at 80% and again increases at 90% bandwidth. The number of packets delivered at the destination is 436, 449, 533, 488, and 460 at each of the five simulation runs. These numbers show a decrease at 80% and 90%. And the number of packets dropped is 132, 188, 188, 155, and 294 at each of the five simulation runs. Again, this shows that drop is not linear like that of packets created. The average packet delay (in milliseconds) at 70%, 80%, and 90% is less than the average delay at 60%.

Uncompressed Data

- **Simulation Run 1**

CUNY-Milano, origin node, created 1553, 2153, 2628, 3303, and 3325 to send to CUNY-Princeton, destination node, at 50%, 60%, 70%, 80%, and 90% bandwidth, respectively. The number of packets created increased from one bandwidth to another linearly. The number of packets delivered at the destination is 565, 516, 439, 348, and 482 at each of the five simulation runs. These numbers show a decrease from 565 packets delivered at 50% to 516 packets delivered at 60% and the number of packets delivered further decreased at 70% and 80%

bandwidth for simulation. We observe a 39% increase at 90% bandwidth for simulation, from 348 packets delivered to 482 packets delivered. And the number of packets dropped is 308, 790, 1029, 1731, and 1550 at each of the five simulation runs. The average packet delay (in milliseconds) increases from 50% bandwidth to 80% bandwidth and decreases at 90%.

- **Simulation Run 2**

The data for simulation run 2 is the same as for compressed data in simulation run 1. Thus, it will not be repeated here again.

Comparison of Compressed and Uncompressed Data

The number of packets created for compressed data is lower than for uncompressed data for all bandwidths in both simulations. These statistics are given in **Tables 9.31** and **9.32**. The number packets delivered for compressed data is less for 50% (450/565) and 90% (305/482) bandwidths and more for 60%-80% (545/516, 513/439, 422/348) bandwidths than for uncompressed data in the first simulation run. And in the second simulation run, the number of packets delivered for compressed data is less at 50% and 60% bandwidths and more for 70%-90% bandwidths than for uncompressed data. The number of packets dropped similarly is less for compressed than for uncompressed data in simulation run 1. In simulation run 2 the number of packets dropped is more at 50%-60%

bandwidths and less at 70%-90 bandwidths for compressed than for uncompressed data. The ratio of packets dropped over packets delivered (packets dropped/packets delivered) in simulation run 1 is 27%, 24%, 38%, 72%, and 219% for compressed data and it is 55%, 153%, 234%, 497%, and 322% for uncompressed data. In simulation run 2, the ratio for compressed data is 23%, 30%, 26%, 24%, and 39% and for uncompressed data it is the same as for compressed data in simulation run 1, given above. The high percentages at 60%-90% bandwidths for uncompressed data in simulation run 1 is due to the increase in packets dropped. The average packet delay (in milliseconds) increases from 50% to 60% and then decreases from 70% to 90% for compressed data and uncompressed data in simulation runs 1 and 2. In case of simulation run 1 the delay for uncompressed data increases linearly from 50% to 80% bandwidths and decreases at 90% bandwidth.

MontpelierP/Montpelier-Milano (Origin/Destination)

Compressed Data

- **Simulation Run 1**

MontpelierP, origin node, created 985, 1121, 1360, 1297, and 1810 to send to Montpelier-Milano, destination node, at 50%, 60%, 70%, 80%, and 90% bandwidths, respectively. The number of packets delivered at the destination is 508, 365, 548, 449, and 686 at the five simulation runs, respectively. And the

number of packets dropped similarly is 160, 134, 231, 201, and 400. The number of packets created increases linearly from 50% to 70% and decreases at 80% and again increases at 90%. But, the number of packets delivered and dropped oscillates between 50% and 90% bandwidths. The average packet delay (in milliseconds) is less at 90% (5355.856) than that of at 50% (9583.161) when the number of packets delivered (1810) is more.

- **Simulation Run 2**

MontpelierP, origin node, created 516, 589, 627, 993, and 854 to send to Montpelier-Milano, destination node, at 50%, 60%, 70%, 80%, and 90% bandwidths, respectively. The number of packets delivered at the destination is 414, 468, 502, 680, and 607 at the five simulation runs, respectively. And the number of packets dropped similarly is 107, 121, 150, 299, and 290. The number of packets created, delivered, and dropped increases linearly from 50% to 80% and decreases at 90%. The average packet delay (in milliseconds) oscillates between 50% and 90% bandwidths.

Uncompressed Data

- **Simulation Run 1**

MontpelierP, origin node, created 2160, 2298, 3183, 2533, and 2901 to send to Montpelier-Milano, destination node, at 50%, 60%, 70%, 80%, and 90%

bandwidths, respectively. The number of packets delivered at the destination is 539, 394, 523, 633, and 507 at the five simulation runs, respectively. And the number of packets dropped similarly is 651, 717, 1126, 879, and 1369. The number of packets created from 50% to 70% bandwidths increases linearly and decreases at 80% to 2533 and increases by 15% at 90% bandwidth. The ratio of packets dropped over packets delivered (packets dropped/packets delivered) is 121%, 182%, 215%, 139%, and 270% at 50%-90% bandwidths. The average packet delay (in milliseconds) at 60% bandwidth is 969.843ms, less than at 50% bandwidth, 5316.473ms. However, the number of packets created is more at 60% than at 50%. The average packet delay increases linearly from 70% to 90%.

- **Simulation Run 2**

This is similar to the compressed data in simulation run 1.

Comparison of Compressed and Uncompressed Data

The number of packets created for compressed data is less than that of uncompressed data for all 5 bandwidths in simulation run 1, a function of compression, and it is less for compressed data at 50%-80% and more at 90% bandwidth than for uncompressed data in simulation run 2. However, for compressed data the number packets delivered in simulation run 1 is more at 70% and 90% bandwidths and it is more in simulation run 2 at 70% and 80%

bandwidths, respectively. The number of packets dropped for compressed data is also less for all simulation runs than for uncompressed data except in the case of simulation run 2 at 80%. The ratio of packets dropped over packets delivered (packets dropped/packets delivered) is 31%, 37%, 42%, 45%, and 58% for compressed data and it is 121%, 182%, 215%, 139%, and 270% for uncompressed data in simulation run 1. And for compressed data in simulation run 2, it is 26%, 26%, 30%, 44%, and 48%. We observe that the number of packets dropped is more than the number of packets delivered for uncompressed data in simulation run 1.

SimRun1	Message & Response Sources: Packets Delivered/Packet Delay									
Audio (128kbps)	Compressed Data					Uncompressed Data				
MPEG-1 (50%)	Bandwidth Used For Simulation									
Origin/Destination	50%	60%	70%	80%	90%	50%	60%	70%	80%	90%
CUNY-Milano/ CUNY-Princeton										
Packets Created	965	1057	1390	1551	2195	1553	2153	2628	3303	3325
Packets Delivered	450	545	513	422	305	565	516	439	348	482
Packets Dropped	121	133	195	305	667	308	790	1029	1731	1550
Dropped/Created	13%	13%	14%	20%	30%	20%	37%	39%	52%	47%
Dropped/Delivered	27%	24%	38%	72%	219%	55%	153%	234%	497%	322%
Packet Delay (MS)										
Average	9232.176	11001.672	10591.897	9532.202	5349.240	5148.137	10762.626	13766.590	23814.220	23116.161
Maximum	14334.006	18128.256	15970.756	15533.506	9659.006	8970.756	54042.506	53944.256	52970.756	48626.256
MontpelierP/ Montpelier-Milano										
Packets Created	985	1121	1360	1297	1810	2160	2298	3183	2533	2901
Packets Delivered	508	365	548	449	686	539	394	523	633	507
Packets Dropped	160	134	231	201	400	651	717	1126	879	1369
Dropped/Created	16%	12%	17%	15%	22%	30%	31%	35%	35%	47%
Dropped/Delivered	31%	37%	42%	45%	58%	121%	182%	215%	139%	270%
Packet Delay (MS)										
Average	9583.161	9772.918	10175.253	10060.550	5355.856	5316.473	969.843	8830.266	13415.519	22379.379
Maximum	14864.753	18341.753	15620.003	15116.503	9521.003	8991.003	11682.253	43953.753	51839.003	49229.503

	Message & Response Sources: Packets Delivered/Packet Delay (Cont'd)									
Audio (128kbps)	Compressed Data					Uncompressed Data				
MPEG-1 (50%)	Bandwidth Used For Simulation									
Origin/Destination	50%	60%	70%	80%	90%	50%	60%	70%	80%	90%
CUNY-Bern/ Bern-Bonn										
Packets Created	1000	1108	1190	1992	2176	1844	2558	2832	2457	3246
Packets Delivered	453	499	509	553	453	404	479	431	530	499
Packets Dropped	126	158	194	560	597	455	895	927	1324	1402
Dropped/Created	13%	14%	16%	28%	27%	25%	35%	33%	54%	43%
Dropped/Delivered	28%	32%	38%	101%	132%	113%	187%	215%	250%	281%
Packet Delay (MS)										
Average	9915.819	12008.643	10335.108	9104.954	5851.351	5241.803	11443.566	15826.163	17256.881	20231.407
Maximum	14917.753	18984.753	15998.003	14998.003	9307.503	8656.003	52731.503	54958.253	54971.503	47931.753
CUNY-Bonn/ Milano-Bonn										
Packets Created	767	1039	1743	1523	1547	1674	2281	2423	3094	3586
Packets Delivered	529	481	455	568	372	503	424	459	428	326
Packets Dropped	128	167	330	261	292	376	743	783	1428	1418
Dropped/Created	17%	16%	19%	17%	19%	22%	33%	32%	46%	40%
Dropped/Delivered	24%	35%	73%	46%	78%	75%	175%	171%	334%	435%
Packet Delay (MS)										
Average	9939.267	10535.662	10448.654	9989.516	5889.999	5454.473	8338.471	10072.075	17194.749	21953.951
Maximum	14452.503	18646.503	15945.003	15468.003	9320.753	8960.753	48716.753	52945.003	52918.503	43830.503

Audio (128kbps)	Message & Response Sources: Packets Delivered/Packet Delay (Cont'd)									
MPEG-1 (50%)	Compressed Data					Uncompressed Data				
	Bandwidth Used For Simulation									
Origin/Destination	50%	60%	70%	80%	90%	50%	60%	70%	80%	90%
CUNY-Montpelier/ Montpelier-Milano										
Packets Created	1082	969	1537	1546	1758	1978	2155	3076	2677	2796
Packets Delivered	489	490	451	556	543	441	482	267	525	624
Packets Dropped	155	139	274	317	409	443	665	1349	1064	1181
Dropped/Created	14%	14%	18%	21%	23%	22%	31%	44%	40%	42%
Dropped/Delivered	32%	28%	61%	57%	75%	100%	138%	505%	203%	189%
Packet Delay (MS)										
Average	8969.393	11204.736	9946.361	8502.227	4775.361	5601.863	4105.161	7729.983	13661.532	16979.976
Maximum	14492.253	18640.253	15984.753	14971.503	9394.753	8947.503	51049.503	40734.003	50998.003	48640.253
CUNYPrinceton/ Princeton-Mont										
Packets Created	1040	1185	1455	1509	1721	1977	2517	2277	3197	3209
Packets Delivered	536	560	441	425	487	532	478	629	521	474
Packets Dropped	191	214	305	320	299	556	801	702	994	1546
Dropped/Created	18%	18%	21%	21%	17%	28%	32%	31%	31%	48%
Dropped/Delivered	36%	38%	69%	75%	61%	105%	168%	112%	191%	326%
Packet Delay (MS)										
Average	9852.637	11380.297	9502.408	9449.144	5103.472	6055.824	9911.519	10480.762	13748.807	23421.305
Maximum	14698.753	18712.753	15613.753	15196.003	9256.003	8971.503	49751.003	53905.253	49984.753	50030.753

Table 9. 31: Audio SimRun1 - Message & Response Sources: Packets Delivered/Delay

SimRun2	Message & Response Sources: Packets Delivered/Packet Delay									
Audio (128kbps)	Compressed Data					Uncompressed Data				
MPEG-1 (50%)	Bandwidth Used For Simulation									
Origin/Destination	50%	60%	70%	80%	90%	50%	60%	70%	80%	90%
CUNY-Milano/ CUNY-Princeton										
Packets Created	562	637	721	643	754	965	1057	1390	1551	2195
Packets Delivered	436	449	533	488	460	450	545	513	422	305
Packets Dropped	132	188	188	155	294	121	133	195	305	667
Dropped/Created	23%	30%	26%	24%	39%	13%	13%	14%	20%	30%
Dropped/Delivered	30%	42%	35%	32%	64%	27%	24%	38%	72%	219%
Packet Delay (MS)										
Average	594.112	1248.713	1126.559	974.53	694.904	9232.176	11001.672	10591.897	9532.202	5349.240
Maximum	2819.255	11717.755	6998.756	9846.256	5043.506	14334.006	18128.256	15970.756	15533.506	9659.006
MontpelierP/ Montpelier-Milano										
Packets Created	516	589	627	993	854	985	1121	1360	1297	1810
Packets Delivered	414	468	502	680	607	508	365	548	449	686
Packets Dropped	107	121	150	299	290	160	134	231	201	400
Dropped/Created	21%	21%	24%	30%	34%	16%	12%	17%	15%	22%
Dropped/Delivered	26%	26%	30%	44%	48%	31%	37%	42%	45%	58%
Packet Delay (MS)										
Average	914.012	536.504	1257.827	1237.201	1016.852	9583.161	9772.918	10175.253	10060.550	5355.856
Maximum	5709.361	2712.753	10360.253	7957.753	6775.503	14864.753	18341.753	15620.003	15116.503	9521.003

SimRun2	Message & Response Sources: Packets Delivered/Packet Delay (Cont'd.)									
Audio (128kbps)	Compressed Data					Uncompressed Data				
MPEG-1 (50%)	Bandwidth Used For Simulation									
Origin/Destination	50%	60%	70%	80%	90%	50%	60%	70%	80%	90%
CUNY-Bern/ Bern-Bonn										
Packets Created	394	596	729	772	898	1000	1108	1190	1992	2176
Packets Delivered	319	435	417	477	507	453	499	509	553	453
Packets Dropped	75	161	312	330	404	126	158	194	560	597
Dropped/Created	19%	27%	43%	43%	45%	13%	14%	16%	28%	27%
Dropped/Delivered	24%	37%	75%	69%	80%	28%	32%	38%	101%	132%
Packet Delay (MS)										
Average	492.768	436.099	754.977	1623.813	1835.493	9915.819	12008.643	10335.108	9104.954	5851.351
Maximum	4828.611	1002.503	6918.503	7983.503	12004.503	14917.753	18984.753	15998.003	14998.003	9307.503
CUNY-Bonn/ Milano-Bonn										
Packets Created	494	612	654	804	642	767	1039	1743	1523	1547
Packets Delivered	395	450	410	425	347	529	481	455	568	372
Packets Dropped	99	168	244	375	295	128	167	330	261	292
Dropped/Created	20%	27%	37%	47%	46%	17%	16%	19%	17%	19%
Dropped/Delivered	25%	37%	60%	88%	85%	24%	35%	73%	46%	78%
Packet Delay (MS)										
Average	467.434	582.266	773.287	913.98	619.024	9939.267	10535.662	10448.654	9989.516	5889.999
Maximum	1181.253	5628.503	3928.253	5759.253	7004.753	14452.503	18646.503	15945.003	15468.003	9320.753

SimRun2	Message & Response Sources: Packets Delivered/Packet Delay (Cont'd.)									
Audio (128kbps)	Compressed Data					Uncompressed Data				
MPEG-1 (50%)	Bandwidth Used For Simulation									
Origin/Destination	50%	60%	70%	80%	90%	50%	60%	70%	80%	90%
CUNY-Montpelier/ Montpelier-Milano										
Packets Created	568	598	677	584	735	1082	969	1537	1546	1758
Packets Delivered	440	446	481	354	352	489	490	451	556	543
Packets Dropped	128	152	198	227	383	155	139	274	317	409
Dropped/Created	23%	25%	29%	39%	52%	14%	14%	18%	21%	23%
Dropped/Delivered	29%	34%	41%	64%	109%	32%	28%	61%	57%	75%
Packet Delay (MS)										
Average	483.6	822.103	831.239	608.39	598.071	8969.393	11204.736	9946.361	8502.227	4775.361
Maximum	2606.003	5851.253	6051.753	3852.003	9991.503	14492.253	18640.253	15984.753	14971.503	9394.753
CUNYPrinceton/ Princeton-Mont										
Packets Created	538	584	694	624	903	1040	1185	1455	1509	1721
Packets Delivered	431	441	530	452	638	536	560	441	425	487
Packets Dropped	111	145	194	172	243	191	214	305	320	299
Dropped/Created	21%	25%	28%	28%	27%	18%	18%	21%	21%	17%
Dropped/Delivered	26%	33%	37%	38%	38%	36%	38%	69%	75%	61%
Packet Delay (MS)										
Average	458.356	618.309	1145.205	797.83	1200.042	9852.637	11380.297	9502.408	9449.144	5103.472
Maximum	2949.756	6811.899	4980.003	4775.753	12858.753	14698.753	18712.753	15613.753	15196.003	9256.003

Table 9. 32: Audio SimRun2 - Message & Response Sources: Packets Delivered/Delay

9.3. Image

Image is another class of data that is very suitable for compression. One frame of a television signal consists of 480 lines with 640 pixels/line. This frame will produce 4800 8 by 8 pixel blocks for the luminance signal for one frame. Compression is definitely useful in this case. MPEG-1 has been chosen to simulate image application. MPEG-1 standard provides a lossy compression algorithm that can be adjusted to balance image quality versus file size. Clearly, in the case of images, we either make do with lower resolution and/or lower quality images or we procure higher speed communications.

The compression rate for MPEG-1 is 30%. Image frame size is 640 pixel x 480 line x 8 bits/sample. The message size is 307250 bytes or 2.458Mbits, warmup length (specifies the time in the simulation when the application will begin to collect data) is 10 seconds and replication length (the number of seconds of simulated time during which statistics are collected for each report) is 50 seconds. The bandwidth used for simulation is 466.560Mbps. Just as in e-mail and audio applications, the distribution of the number of observations in an interval is Poisson distributed. And the parameter entered for the Poisson distribution is the interarrival time (IAT, the time from the start of one message to

the start of the next). The total number of simulation runs is 10. Each simulation run time is approximately 2-3 hours. The simulation run statistics for compressed and uncompressed data are shown in **Tables 9.33** and **9.34**, respectively.

Examination of the data in **Tables 9.33** and **9.34** show that as the bandwidth increases:

1. IAT decreases
2. Packets/Second increase (see **Figures 9.9** and **9.10**)
3. Bits/Second increase
4. Each simulation run time is approximately 2-3 hours and the message size used for Image is large.
5. The rate of transmission for compressed data at each bandwidth is less than for uncompressed data.

as was observed in E-mail and Audio applications. As the tables show, when IAT decreases the number of packets/second increases linearly in both data types. This is depicted in **Figures 9.9** and **9.10**.

Bandwidth: OC-9, 466.560Mbps, Replication Length: 50s					
Message Size: 307250 bytes = 2.458 Mbits					
MPEG-1 (30%)	Bandwidth Used for Simulation				
	50%	60%	70%	80%	90%
InterArrival Time (IAT)	0.015	0.0125	0.011	0.009	0.008
Poisson Distribution (poi)					
Frames/Second(fps)	67	80	93	107	120
MBits/Second(Mbps)	164.69	196.64	228.59	263.01	294.96
Simulation Time (h:mm)	2:10	2:35	2:40	2:45	3:00

Table 9. 33: Image Simulation Run Statistics for Compressed Data

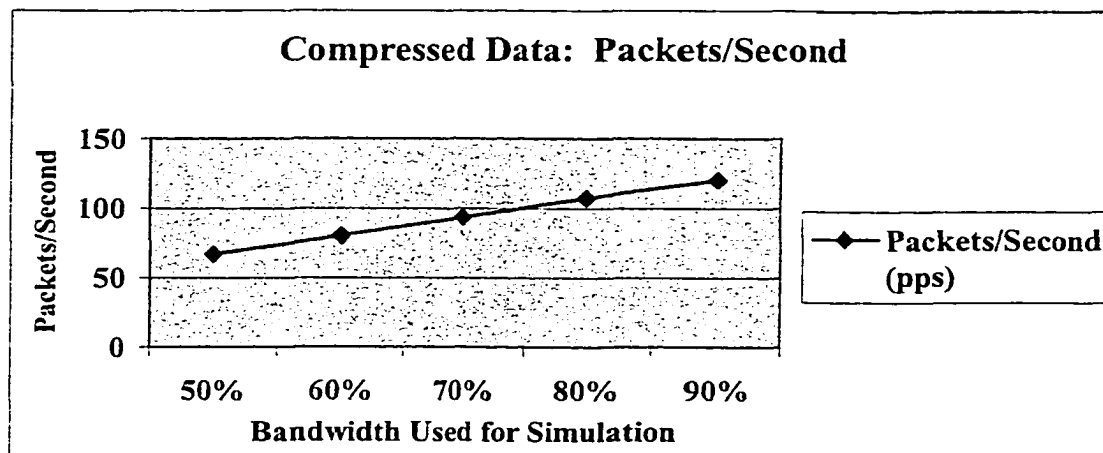


Figure 9. 15: Image - Compressed Data: Packets/Second

Bandwidth: OC-9, 466.560Mbps, Replication Length: 50s					
Message Size: 307250 bytes = 2.458 Mbits					
MPEG-1 (30%)	Bandwidth Used for Simulation				
	50%	60%	70%	80%	90%
InterArrival Time (IAT)	0.0105	0.009	0.0075	0.0066	0.006
Poisson Distribution (poi)					
Packets/Second(pps)	95	114	133	152	171
Bits/Second(bps)	233.51	280.21	326.91	373.62	420.32
Simulation Time (min)	1:40	1:45	3:00	3:00	3:10

Table 9. 34: Image Simulation Run Statistics for Uncompressed Data

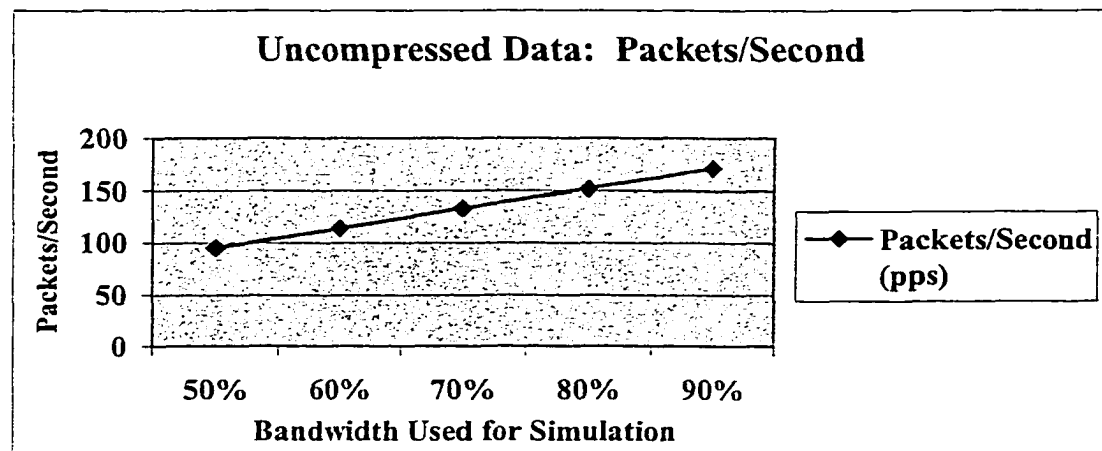


Figure 9. 16: Image - Uncompressed Data: Packets/Second

MPEG-1

MPEG-1 (approved by ITU-T, November 1992) is chosen for image compression because it is the standard for image on storage media. It is intended for low-resolution image sequences like those composed of 320 pixel x 240 line images. It performs well using 1.5 Mbits/second transport links. MPEG2 (transport) is intended for higher resolution applications, such as full-resolution, 640 pixel x 480 line video image sequences [56]. The MPEG2 standard requires transport links between 4 and 10 Mbits/second, depending on the required decompressed image quality. Reports are produced automatically at the end of each simulation run. COMNET III generates a statistical report of Link Utilization for various processing nodes as shown in **Tables 9.35** and **9.36** for compressed and uncompressed data, respectively.

Analysis

- **Compressed Data**

The number of frames delivered from CUNY to Milano is 157636, 165397, 215800, 188802, and 241496 and at 50%, 60%, 70%, 80%, and 90% bandwidths, respectively. The number of packets delivered by the same link is 17870, 18750, 24465, 21403, and 27377 at 50%, 60%, 70%, 80%, and 90% bandwidths, respectively. From 50% to 70% we see a linear increase in the

number of frames/packets delivered. In both cases, the increase is 5% (from 50% to 60%) and 30% (from 60% to 70%). But, from 70% to 80%, the number of frames/packets delivered decreases by 12% and increases again from 80% to 90% by 28%. The slope of the curve is positive for 50% to 70% bandwidth simulation runs, negative for 80% and again positive for 90%, in each category, packets/second, bytes delivered, kilobits per second (kbps), and link utilization percentage. The second link from CUNY to Princeton oscillates for all types. But, for the remaining two links, from Montpelier to Princeton and from Milano to Montpelier the number of frames/packets delivered gradually increases linearly. The same holds for the other categories for these two links.

MPEG-1 (30%)	Compressed Data of Image				
MSGSIZE: 307250B	Bandwidth Used For Simulation				
Link	50%	60%	70%	80%	90%
From CUNY-Milano					
Frames Delivered	157636	165397	215800	188802	241496
Packets Delivered	17870	18750	24465	21403	27377
Packets/second	357.400	375.000	489.300	428.060	547.540
Bytes Delivered	1097932800	1152000000	1503142400	1315008000	1682086400
KBPS Delivered	175669.248	184320.000	240502.784	210401.280	269133.824
Utilization (%)	39.41	41.35	53.95	47.20	60.37
From CUNYPrincet					
Frames Delivered	127537	230371	201179	271557	217954
Packets Delivered	14436	26117	22807	30787	24709
Packets/second	288.720	522.340	456.140	615.740	494.180
Bytes Delivered	886950400	1604646400	1401267200	1891571200	1518144000
KBPS Delivered	141912.064	256743.424	224202.752	302651.392	242903.04
Utilization (%)	31.84	57.59	50.29	67.89	54.49
From MontpelierPri					
Frames Delivered	129897	162256	181816	194007	252357
Packets Delivered	14725	18393	20612	21994	28610
Packets/second	294.500	367.860	412.240	439.880	572.200
Bytes Delivered	904704000	1130073600	1266406400	1351321600	1757811200
KBPS Delivered	144752.64	180811.776	202625.024	216211.456	281249.792
Utilization (%)	32.47	40.56	45.45	48.50	63.09
From Milano-Mont					
Frames Delivered	165130	169702	217326	240460	293453
Packets Delivered	18720	19238	24640	27263	33277
Packets/second	374.400	384.760	492.800	545.260	665.540
Bytes Delivered	1150156800	1182003200	1513779200	1674880000	2044121600
KBPS Delivered	184025.088	189120.512	242204.672	267980.800	327059.456
Utilization (%)	41.28	42.43	54.33	60.12	73.36

Table 9. 35: Image - Links: Channel Utilization/Utilization by Application for Compressed Data, OC-9, 466.560Mbps

- **Uncompressed Data**

In case of uncompressed data, the number of frames delivered from CUNY to Milano is a steady upward increase, 188149, 18802, 241496, and 292237 at 50%, 60%, 70%, 80% bandwidths, respectively, and at 90% bandwidth the number decreases to 275676 from 292237. Similarly, the number of packets delivered by the same link increases linearly from 50% to 80% and decreases from 80% to 90%. From 50% to 60% we see a slight increase of 0.3% in frames/packets delivered. From 60% to 70%, the increase is 28%; from 70% to 80%, it is 21% and from 80% to 90% the decrease is 6%. The slope of number of packets delivered is upward in each category, packets/second, bytes delivered, kilobits per second (kbps), and link utilization percentage for 50%-80% bandwidths and for 90% bandwidth it is downward. In case of from CUNY to Princeton link, the number of frames/packets delivered goes down by 20% at 70% bandwidth. From Montpelier to Princeton link, the number of frames/packets delivered decreases by 1% at 60% bandwidth and from Milano to Montpelier link the decrease is by 2% at 80% bandwidth. These changes also reflect in other categories, packets/second, bytes delivered, kbps delivered, and utilization percentage as shown in **Table 9.36**.

MPEG-1 (30%)	Uncompressed Data of Image				
MsgSize: 307250B	Bandwidth Used For Simulation				
Link	50%	60%	70%	80%	90%
From CUNY-Milano					
Frames Delivered	188149	188802	241496	292237	275676
Packets Delivered	21330	21403	27377	33132	31254
Packets/second	426.600	428.060	547.540	662.640	625.080
Bytes Delivered	1310515200	1315008000	1682086400	2035686400	1920307200
KBPS Delivered	209682.432	210401.28	269133.824	325709.824	307249.152
Utilization (%)	47.04	47.20	60.37	73.06	68.92
From CUNYPrincet					
Frames Delivered	172143	271557	217954	240052	279687
Packets Delivered	19515	30787	24709	27215	31707
Packets/second	390.300	615.740	494.180	544.300	634.140
Bytes Delivered	1199001600	1891571200	1518144000	1672102400	1948134400
KBPS Delivered	191840.256	302651.392	242903.040	267536.384	311701.504
Utilization (%)	43.04	67.89	54.49	60.01	69.92
From MontpelierPri					
Frames Delivered	195639	194007	252357	256304	334387
Packets Delivered	22179	21994	28610	29056	37910
Packets/second	443.580	439.880	572.200	581.120	758.200
Bytes Delivered	1362688000	1351321600	1757811200	178525440	2329280000
KBPS Delivered	21830.08	216211.456	281249.792	285640.704	372684.8
Utilization (%)	48.91	48.5	63.09	64.08	83.60
From Milano-Mont					
Frames Delivered	179535	240460	293453	288568	291302
Packets Delivered	20353	27263	33277	32716	33026
Packets/second	407.680	545.260	665.540	654.320	660.520
Bytes Delivered	1250496000	1674880000	2044121600	2010086400	2029056000
KBPS Delivered	200079.36	267980.8	327059.456	321613.824	324648.96
Utilization (%)	44.88	60.12	73.36	72.14	72.83

Table 9. 36: Image - Links: Channel Utilization/Utilization by Application for Uncompressed Data, OC-9, 466.560Mbps

First row in **Tables 9.35** and **9.36** represents packets delivered during the simulation. They fluctuate for some links and increase as the bandwidth increases for others. This fluctuation could be a function of network delay and/or link/node failure.

Fluctuations in:

- Packets/Second (Packets Delivered/Replication Length, 50s);
- Bytes Delivered (Packets Delivered * Message Size);
- KBPS (kilobits/second, Packets/Second * Message Size * 8bits)
- Link Utilization

can be seen as reflections of the fluctuations in packets delivered.

Comparison of the results shown in **Table 9.37** demonstrates that the utilization rate for compressed data is consistently lower than for uncompressed data in most cases. This implies that compression permits transmission of larger volume of data in shorter intervals.

Image	Link Utilization (%)	
Link	Compressed Data	Uncompressed Data
From CUNY-Milano		
50%	39.41	47.04
60%	41.35	47.20
70%	53.95	60.37
80%	47.20	73.06
90%	60.37	68.92
From CUNYPrinceton		
50%	31.84	43.04
60%	57.59	67.89
70%	50.29	54.49
80%	67.89	60.01
90%	54.49	69.92
From MontpelierPri		
50%	32.47	48.91
60%	40.56	48.50
70%	45.45	63.09
80%	48.50	64.08
90%	63.09	83.60
From Milano-Mont		
50%	41.28	44.88
60%	42.43	60.12
70%	54.33	73.36
80%	60.12	72.14
90%	73.36	72.83

Table 9. 37: Image - Comparison of Utilization (%) for Compressed & Uncompressed Data

WAN Clouds: Access Link Statistics Analysis

Tables 9.38, 9.39, 9.40, and 9.41 present statistics for 7 access links: (1) CUNY-Montpelier, (2) CUNY-Milano, (3) CUNY-Bern, (4) CUNY-Bonn, (5) Princeton-Montpelier, (6) CUNY-Princeton, and (7) Montpelier-Milano. Only three are chosen for analysis to demonstrate the pattern of variation over access links.

Compressed Data	Frames Accepted				
Access Link	50%	60%	70%	80%	90%
CUNY-Montpelier	14292	14180	13569	14266	14180
CUNY-Milano	14119	13734	14069	14247	14284
CUNY-Bern	13488	13789	14244	14179	14224
CUNY-Bonn	13611	13358	14100	14245	14208
Princeton-Montpelier	11883	14187	13213	13954	14233
CUNY-Princeton	12750	14277	14293	14089	14240
Montpelier-Milano	13668	14174	13503	14031	14162

Table 9. 38: Image - WAN Cloud Access Link Statistics: Frames Accepted (Compressed Data)

Compressed Data	Frames Dropped				
Access Link	50%	60%	70%	80%	90%
CUNY-Montpelier	1525	1637	1531	1549	1635
CUNY-Milano	1553	1573	1548	1568	1532
CUNY-Bern	1460	1579	1573	1636	1591
CUNY-Bonn	1545	1558	1551	1571	1606
Princeton-Montpelier	1280	1542	1497	1560	1584
CUNY-Princeton	1464	1542	1530	1580	1581
Montpelier-Milano	1542	1641	1520	1624	1656

Table 9. 39: Image - WAN Cloud Access Link Statistics: Frames Dropped (Compressed Data)

Uncompressed Data	Frames Accepted				
Access Link	50%	60%	70%	80%	90%
CUNY-Montpelier	14195	14266	14180	14256	14200
CUNY-Milano	14262	14247	14284	14289	14164
CUNY-Bern	14270	14179	14224	14225	14291
CUNY-Bonn	14240	14245	14208	14233	14193
Princeton-Montpelier	13994	13954	14233	14227	14333
CUNY-Princeton	14170	14089	14240	14192	14287
Montpelier-Milano	14256	14031	14162	14263	14210

Table 9. 40: Image - WAN Cloud Access Link Statistics: Frames Accepted (Uncompressed Data)

Uncompressed Data	Frames Dropped				
Access Link	50%	60%	70%	80%	90%
CUNY-Montpelier	1621	1549	1635	1559	1615
CUNY-Milano	1554	1568	1532	1532	1659
CUNY-Bern	1545	1636	1591	1591	1524
CUNY-Bonn	1575	1571	1606	1581	1622
Princeton-Montpelier	1575	1560	1584	1592	1499
CUNY-Princeton	1613	1580	1581	1626	1546
Montpelier-Milano	1546	1624	1656	1552	1618

Table 9. 41: Image - WAN Cloud Access Link Statistics: Frames Dropped (Uncompressed Data)

- **Compressed Data**

CUNY-Montpelier access link accepted 14292 frames at 50% and 14180 frames at 60%, a decrease of approximately 1%. At 70% the number of frames accepted by the link went down to 13569 frames, a decrease of 4%. The number of frames accepted at 80% increased by 5% from 70% and at 90% the frames accepted are 14180, a decrease of 1%. CUNY-Montpelier access link dropped frames at each simulation run. The number of dropped frames at each bandwidth percentage is 1525 at 50%, 1637 at 60%, 1531 at 70%, 1549 at 80% and 1635 at 90%. The drop rate is 11%, 12%, 11%, 11%, and 12% for all five different bandwidth simulation runs, 50%, 60%, 70%, 80%, and 90%, respectively.

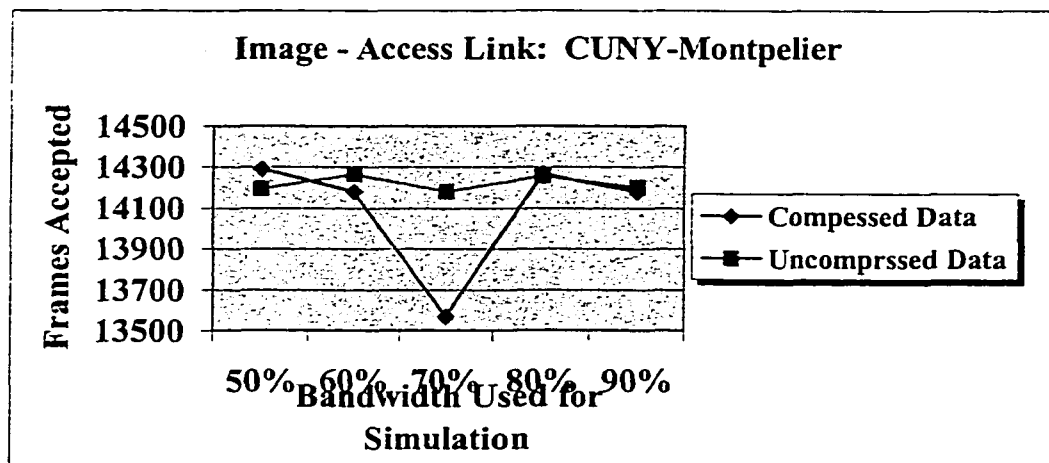


Figure 9. 17: Image– Selected Access Link: CUNY-Montpelier (Frames Accepted)

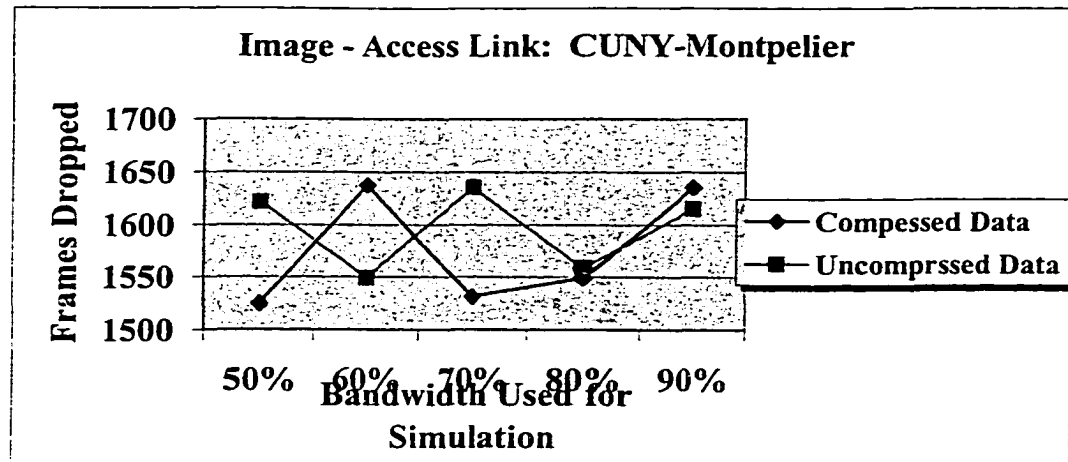


Figure 9. 18: Image— Selected Access Link: CUNY-Montpelier (Frames Dropped)

CUNY-Bonn access link accepted 13611 frames at 50% and 13358 frames at 60%, a decrease of 2 percent. At 70% the number of frames accepted by the link went up to 14100, an increase of 5%. The number of frames accepted at 80% went up to 14245, 1% increase, and at 90% the frames accepted are 14208, a difference of 37 frames. CUNY-Bonn access link dropped frames at each simulation run. The number of dropped frames at each bandwidth percentage is 1545 at 50%, 1558 at 60%, 1551 at 70%, 1571 at 80% and 1606 at 90%. The drop rate is 11%, 12%, 11%, 11%, and 11% for all five different bandwidth simulation runs, 50%, 60%, 70%, 80%, and 90%, respectively.

The number of frames accepted by Montpelier-Milano access link for compressed data at different bandwidth simulation runs are 13668, 14174, 13503, 14031, and 14162. The number of accepted frames decreased at 70% by nearly 5% and again increased at 80% (by 4%) and 90% (by 1%) bandwidths. Montpelier-Milano access link dropped frames at each simulation run. The number of dropped frames at each bandwidth percentage is 1542 at 50%, 1641 at 60%, 1520 at 70%, 1624 at 80% and 1656 at 90%. The drop rate ranges from 11% to 12%.

- **Uncompressed Data**

The number of accepted frames by CUNY-Montpelier access link at 50% bandwidth is 14195 and at 60% an increase of 71 frames or 0.5%. At 70% the number of frames accepted by the link went down to 14180 frames, a decrease of 0.6%. The number of frames accepted at 80% differ from 70% by 76 frames or 0.5% and at 90% the frames accepted are 14200, a decrease of 0.4%. CUNY-Montpelier access link dropped frames at each simulation run. The number of dropped frames at each bandwidth percentage is 1621 at 50%, 1549 at 60%, 1635 at 70%, 1559 at 80% and 1615 at 90%. The drop rate is 11%, 11%, 12%, 11%, and 11% for all five different bandwidth simulation runs, respectively.

The number of frames accepted by CUNY-Bonn access link are 14240, 14245, 14208, 14233, and 14193 at 50%, 60%, 70%, 80%, and 90% bandwidth simulation runs, respectively. These numbers indicate that the link oscillates between 70% and 90% simulation runs. CUNY-Bonn access link dropped frames at each simulation run. The number of dropped frames at each bandwidth percentage is 1575 at 50%, 1571 at 60%, 1606 at 70%, 1581 at 80% and 1622 at 90%. The drop rate is 11% for all five different bandwidth simulation runs.

The number of frames accepted by Montpelier-Milano access link for uncompressed data at different bandwidth simulation runs are 14256, 14031, 14162, 14263, and 14210. The numbers indicate that at 60% the number of frames accepted decreased by approximately 2%, increased linearly from 70% to 80%, and decreased by 0.4% at 90% bandwidth. Montpelier-Milano access link dropped frames at each simulation run. The number of dropped frames at each bandwidth percentage is 1546 at 50%, 1624 at 60%, 1656 at 70%, 1552 at 80% and 1618 at 90%. The drop rate ranges from 11% to 12%.

Comparison of Frames-Dropped

The drop rate of compressed and uncompressed data and their differences are shown in **Table 9.42**. Also shown in the table is the comparison of the link utilization rate for both data types.

The drop rate of CUNY-Montpelier access link at 50% bandwidth for compressed is equal to that of uncompressed data, 11%. Similarly, the difference for the remaining bandwidth percentages, 60%-90%, is very small, $\pm 1\%$ or 0. These differences are well within the margin of error, $\pm 5\%$. The link utilization rate is 100% for compressed and uncompressed data as shown in the table.

The drop rate of CUNY-Bonn access link at 50%, 70%-90% bandwidths for compressed and uncompressed data is 11%. The difference between the two is 0. At 60% bandwidth the difference is 1%. These differences are well within the margin of error, $\pm 5\%$. The link utilization rate for compressed data at 50% is 95.82%, at 60% 94.31%, at 70% it is 98.96%, and at 80%-90% it is 100%. For uncompressed data the link utilization rate is 100% for all bandwidths.

The drop rate of Montpelier-Milano access link for compressed data is 11% which is equal to that of uncompressed data for all bandwidth simulation runs except the 80% run. The difference at 80% is 1%. The link utilization of compressed data at 50% bandwidth is 96.18% compared to 99.91% for uncompressed data. We observe that the link utilization rate for compressed data at 60% is more than for uncompressed data. The utilization rate for compressed data is 100% whereas it is 98.96% for uncompressed data. The link utilization rate for compressed data at 70% and 80% bandwidths is 94.99% and 98.96,

respectively. And the link utilization rate for uncompressed data for the same bandwidths is 100%. At 90% bandwidth the link utilization rate is the same for both types of data.

This demonstrates that while the drop rate is more or less the same for both compressed and uncompressed data, compressing data has a clear and substantive advantage in terms of the link utilization rate. Thus, data compression allows for transmission of substantially larger volume of data in shorter intervals.

Image (OC-9)	WAN CLOUDS: ACCESS LINK STATS (ENTRY)									
MPEG-1 (30%)	Comparison of Compressed & Uncompressed Image Data									
	Frames Accepted/Dropped/Utilization									
Cloud: Access Link	Cmp 50%	Uncmp 50%	Cmp 60%	Uncmp 60%	Cmp 70%	Uncmp 70%	Cmp 80%	Uncmp 80%	Cmp 90%	Uncmp 90%
CUNY-Montpelier										
Frames Accepted	14292	14195	14180	14266	13569	14180	14266	14256	14180	14200
Frames Dropped	1525	1621	1637	1549	1531	1635	1549	1559	1635	1615
Drop Rate	0.1067	0.1141	0.11544	0.10857	0.1128	0.1153	0.1085	0.1093	0.11530	0.11373
Drop Rate (%)	11%	11%	12%	11%	11%	12%	11%	11%	12%	11%
Difference	-1%		1%		-0%		-0%		0%	
Link Utilization (%)	100.00	100.00	100.00	100.00	95.47	100.00	100.00	100.00	100.00	100.00
CUNY-Milano										
Frames Accepted	14119	14262	13734	14247	14069	14284	14247	14289	14284	14164
Frames Dropped	1553	1554	1573	1568	1548	1532	1568	1532	1532	1659
Drop Rate	0.1099	0.1089	0.11453	0.11005	0.1100	0.1072	0.1100	0.1072	0.10725	0.11712
Drop Rate (%)	11%	11%	11%	11%	11%	11%	11%	11%	11%	12%
Difference	0%		0%		0%		0%		-1%	
Link Utilization (%)	99.08	100.00	96.78	100.00	98.75	100.00	100.00	100.00	100.00	100.00

Image (OC-9)	Frames Accepted/Dropped/Utilization (Cont'd.)									
Cloud:	Cmp	Uncmp	Cmp	Uncmp	Cmp	Uncmp	Cmp	Uncmp	Cmp	Uncmp
Access Link	50%	50%	60%	60%	70%	70%	80%	80%	90%	90%
CUNY-Bern										
Frames Accepted	13488	14270	13789	14179	14244	14224	14179	14225	14224	14291
Frames Dropped	1460	1545	1579	1636	1573	1591	1636	1591	1591	1524
Drop Rate	0.1082	0.1082	0.11451	0.11538	0.1104	0.1118	0.1153	0.1118	0.11185	0.10664
Drop Rate (%)	11%	11%	11%	12%	11%	11%	12%	11%	11%	11%
Difference	-0%		-0%		-0%		0%		1%	
Link Utilization (%)	94.51	100.00	97.17	100.00	100.00	100.00	100.00	100.00	100.00	100.00
CUNY-Bonn										
Frames Accepted	13611	14240	13358	14245	14100	14208	14245	14233	14208	14193
Frames Dropped	1545	1575	1558	1571	1551	1606	1571	1581	1606	1622
Drop Rate	0.1135	0.1106	0.11663	0.11028	0.11	0.1130	0.1102	0.1110	0.11303	0.11428
Drop Rate (%)	11%	11%	12%	11%	11%	11%	11%	11%	11%	11%
Difference	0%		1%		-0%		-0%		-0%	
Link Utilization (%)	95.82	100.00	94.31	100.00	98.96	100.00	100.00	100.00	100.00	100.00
Princeton-Montpelier										
Frames Accepted	11883	13994	14187	13954	13213	14233	13954	14227	14233	14333
Frames Dropped	1280	1575	1542	1560	1497	1584	1560	1592	1584	1499
Drop Rate	0.1077	0.1125	0.10869	0.11179	0.1132	0.1112	0.1117	0.1118	0.11129	0.10458
Drop Rate (%)	11%	11%	11%	11%	11%	11%	11%	11%	11%	10%
Difference	-0%		-0%		0%		-0%		1%	
Link Utilization (%)	83.22	98.44	99.44	98.10	93.00	100.00	98.10	100.00	100.00	100.00

Image (OC-9)	Frames Accepted/Dropped/Utilization (Cont'd.)									
Cloud:	Cmp	Uncmp	Cmp	Uncmp	Cmp	Uncmp	Cmp	Uncmp	Cmp	Uncmp
Access Link	50%	50%	60%	60%	70%	70%	80%	80%	90%	90%
CUNY-Princeton										
Frames Accepted	12750	14170	14277	14089	14293	14240	14089	14192	14240	14287
Frames Dropped	1464	1613	1542	1580	1530	1581	1580	1626	1581	1546
Drop Rate	0.1148	0.1138	0.10800	0.11214	0.1070	0.1110	0.1121	0.1145	0.11102	0.10821
Drop Rate (%)	11%	11%	11%	11%	11%	11%	11%	11%	11%	11%
Difference	0%		-0%		-0%		-0%		0%	
Link Utilization (%)	89.87	99.77	100.00	99.02	100.00	100.00	99.02	100.00	100.00	100.00
Montpelier-Milano										
Frames Accepted	13668	14256	14174	14031	13503	14162	14031	14263	14162	14210
Frames Dropped	1542	1546	1641	1624	1520	1656	1624	1552	1656	1618
Drop Rate	0.1128	0.1084	0.11577	0.11574	0.1125	0.1169	0.1157	0.1088	0.11693	0.11386
Drop Rate (%)	11%	11%	12%	12%	11%	12%	12%	11%	12%	11%
Difference	0%		0%		-0%		1%		0%	
Link Utilization (%)	96.18	99.91	100.00	98.96	94.99	100.00	98.96	100.00	100.00	100.00

Table 9. 42: Image - WAN Cloud: Access Link Statistics- Frames Accepted, Frames Dropped & Link Utilization

Message & Response Sources

Table 9.43 shows data for 6 distinct origin nodes that send messages to 6 different destination nodes. Of these, only two nodes are chosen for analysis that follows:

CUNY-Milano/CUNY-Princeton (Origin/Destination)

Compressed Data

CUNY-Milano, origin node, created 17870, 18750, 25735, 22345, and 31670 to send to CUNY-Princeton, destination node, at 50%, 60%, 70%, 80%, and 90% bandwidth, respectively. The number of packets created increases from one bandwidth to another linearly from 50% to 70% bandwidths and 80% bandwidth decreased by 13%. The number of packets delivered at the destination is 2244, 1789, 1792, 1411, and 1709 at the five simulation runs. The number of packets dropped similarly is 3090, 5167, 11843, 8345, and 17323. The average packet delay (in milliseconds) at 60% and 70% is less than the average delay at 50% bandwidth. At 80% and 90% bandwidths the delay is more than the delay at the other bandwidths.

Uncompressed Data

CUNY-Milano, origin node, created 21330, 22345, 31670, 40150, and 40290 to send to CUNY-Princeton, destination node, at 50%, 60%, 70%, 80%, and 90% bandwidth, respectively. We observe that the statistics of uncompressed data at 60% and 70% is equal to the statistics of compressed data at 80% and 90%, respectively. The number of packets created increases from one bandwidth to another linearly. The number of packets delivered at the destination is 2010, 1411, 1709, 1514, and 1313 at the five simulation runs. These numbers show a decrease of 30% from 2010 packets delivered at 50% to 1411 packets delivered at 60%. The number of packets delivered increases at 70% and decreases at 80% and 90% bandwidths. And the number of packets dropped is 6750, 8345, 17323, 26992, and 27823 at the five simulation runs. Again, these numbers indicate that drop is linear like that of packets created. The average packet delay (in milliseconds) oscillates between 50% and 90% bandwidths.

Comparison of Compressed and Uncompressed Data

The number of packets created of compressed data is lower than that of uncompressed data for all 5 bandwidth simulation runs. These statistics are given in **Table 9.43**. The number of packets delivered for compressed data is less at 80% (1411/1514) and more at the rest of the bandwidths than for uncompressed

data. The number of packets dropped for compressed data is less than for uncompressed data for all five bandwidths. The ratio of packets dropped over packets delivered (packets dropped/packets delivered) for compressed data also is less than for uncompressed data. The average packet delay (in milliseconds) for compressed data is less than for uncompressed data except at 90% bandwidth.

Image (OC-9)	Message & Response Sources: Packets Delivered/Packet Delay									
MPEG-1 (30%)	Compressed Data						Uncompressed Data			
	Bandwidth Used For Simulation									
Origin/Destination	50%	60%	70%	80%	90%	50%	60%	70%	80%	90%
CUNY-Milano/ CUNY-Princeton										
Packets Created	17870	18750	25735	22345	31670	21330	22345	31670	40150	40290
Packets Delivered	2244	1789	1792	1411	1709	2010	1411	1709	1514	1313
Packets Dropped	3090	5167	11843	8345	17323	6750	8345	17323	26992	27823
Dropped/Created	17%	28%	46%	37%	55%	32%	37%	55%	67%	69%
Dropped/Delivered	138%	289%	661%	591%	1014%	336%	591%	1014%	1783%	2119%
Packet Delay (MS)										
Average	30097.773	29947.686	28521.704	31577.359	33299.560	33900.009	31577.359	33299.560	33145.263	32521.741
Maximum	51939.093	50997.918	49856.169	55982.977	55597.548	51993.309	55982.977	55597.548	54988.396	54708.867
MontpelierP/ Montpelier-Milano										
Packets Created	14725	18710	20760	22290	29810	22645	22290	29810	40220	46030
Packets Delivered	2244	2205	1664	2049	2109	1252	2049	2109	1997	2048
Packets Dropped	2165	4007	6476	8150	15656	7483	8158	15656	24885	32085
Dropped/Created	15%	21%	31%	37%	53%	33%	37%	53%	62%	70%
Dropped/Delivered	96%	182%	389%	398%	742%	598%	398%	742%	1246%	1567%
Packet Delay (MS)										
Average	30119.095	31858.860	32591.869	33823.905	33979.349	23136.658	33823.905	33979.349	34322.203	33953.155
Maximum	52615.652	53665.703	51989.716	55504.357	55672.947	47548.418	55504.357	55672.947	56998.395	54992.349

Image (OC-9)	Message & Response Sources: Packets Delivered/Packet Delay (Cont'd.)									
MPEG-1 (30%)	Compressed Data						Uncompressed Data			
	Bandwidth Used For Simulation									
Origin/Destination	50%	60%	70%	80%	90%	50%	60%	70%	80%	90%
CUNY-Bern/ Bern-Bonn										
Packets Created	16465	19415	21430	22635	30705	26460	22635	30705	25655	40630
Packets Delivered	2243	2202	2168	2034	2097	2239	2034	2097	2024	2025
Packets Dropped	4555	5203	6963	9318	16278	12423	9318	16278	12009	26569
Dropped/Created	28%	27%	32%	41%	53%	47%	41%	53%	47%	65%
Dropped/Delivered	203%	236%	321%	458%	776%	555%	458%	776%	593%	1312%
Packet Delay (MS)										
Average	31373.488	31774.331	32467.884	31190.082	31504.257	32028.270	31190.082	31504.257	31464.767	31577.202
Maximum	52983.131	50754.908	53986.423	54979.182	53045.993	52995.645	54979.182	53045.993	54995.639	53974.567
CUNY-Bonn/ Milano-Bonn										
Packets Created	16605	18570	19465	21730	36900	26175	21730	36900	38025	43660
Packets Delivered	1725	1005	2123	1920	2085	2171	1920	2085	2034	2001
Packets Dropped	2496	4028	6536	7784	22888	12190	7784	22888	24143	29342
Dropped/Created	15%	22%	34%	36%	62%	47%	36%	62%	63%	67%
Dropped/Delivered	145%	401%	308%	405%	1098%	561%	405%	1098%	1187%	1466%
Packet Delay (MS)										
Average	29804.011	22952.595	31685.880	32424.733	30740.664	32486.062	32424.733	30740.664	30892.738	32212.681
Maximum	51970.619	47838.186	53998.936	54969.304	52987.741	53993.011	54969.304	52987.741	54992.346	52998.934

Image (OC-9)	Message & Response Sources: Packets Delivered/Packet Delay (Cont'd.)									
MPEG-1 (30%)	Compressed Data						Uncompressed Data			
	Bandwidth Used For Simulation									
Origin/Destination	50%	60%	70%	80%	90%	50%	60%	70%	80%	90%
CUNY-Montpelier/ Montpelier-Milano										
Packets Created	16160	19455	20205	27600	37420	23245	27600	37420	36080	47660
Packets Delivered	1841	2231	2173	2037	1690	2248	2037	1690	2012	2045
Packets Dropped	2317	5219	6527	12443	22111	9221	12443	22111	21915	33671
Dropped/Created	14%	27%	32%	45%	59%	40%	45%	59%	61%	71%
Dropped/Delivered	126%	234%	300%	611%	1308%	410%	611%	1308%	1089%	1647%
Packet Delay (MS)										
Average	30421.181	32212.485	32087.260	31032.070	28648.767	31983.596	31032.070	28648.767	31393.718	30819.525
Maximum	50989.717	53982.473	51994.984	50994.329	4994.327	53987.742	50994.329	49994.327	54997.615	54983.787
CUNYPrinceton/ Princeton-Mont										
Packets Created	14675	27135	23600	36080	25540	19515	36080	25540	30995	48020
Packets Delivered	2247	2221	2202	2012	2072	1907	2012	2072	2012	2062
Packets Dropped	1726	11579	9016	20974	10803	5537	20974	10803	16061	33449
Dropped/Created	12%	43%	38%	58%	42%	28%	58%	42%	52%	70%
Dropped/Delivered	77%	521%	409%	1042%	521%	290%	1042%	521%	798%	1622%
Packet Delay (MS)										
Average	30097.414	30688.837	32665.002	30353.210	32593.133	32493.930	30353.210	32593.133	31960.660	33223.669
Maximum	50996.303	49992.351	53018.991	52987.744	55426.644	52999.597	52987.744	55426.644	53976.541	55995.642

Table 9. 43: Image - Message & Response Sources: Packets Delivered/Dropped/Delay

MontpelierP/Montpellier-Milano (Origin/Destination)

Compressed Data

MontpelierP, origin node, created packets 14725, 18710, 20760, 22290, and 29810 to send to Montpellier-Milano, destination node, at 50%, 60%, 70%, 80%, and 90% bandwidths, respectively. The number of packets delivered at the destination similarly is 2244, 2205, 1664, 2049, and 2109. And the number of packets dropped is 2165, 4007, 6476, 8150, and 15656. The number of packets created increases linearly. But, the number of packets delivered decreases from 50% to 70% bandwidths and increases at 80% and 90% bandwidths. The number of packets dropped by the network also increases linearly. The average packet delay (in milliseconds) increases linearly from 50% to 90% bandwidths.

Uncompressed Data

MontpelierP, origin node, created 22645, 22290, 29810, 40220, and 46030 to send to Montpellier-Milano, destination node, at 50%, 60%, 70%, 80%, and 90% bandwidths, respectively. The number of packets delivered at the destination similarly is 1252, 2049, 2109, 1997, and 2048. And the number of packets dropped is 7483, 8158, 15656, 24885, and 32085. The number of packets created decreases at 60% bandwidth by approximately 2% and increases linearly from 60% to 90% bandwidths. The number of packets delivered at the destination node increases linearly from 50% to 70% bandwidths, decreases by 5% at 80%

bandwidth and again increases at 90% bandwidth. The number of packets dropped increases linearly. The ratio of packets dropped over packets delivered (packets dropped/packets delivered) is 598% (7483/1252), 398% (8158/2049), 742% (15656/2109), 1246% (24885/1997), and 1567% (32085/2048) at 50%-90% bandwidths, respectively. The average packet delay (in milliseconds) increases linearly from 50% to 80% and decreases at 90% bandwidth.

Comparison of Compressed and Uncompressed Data

The number of packets created for compressed data is less than for uncompressed data for all 5 bandwidth simulation runs, a function of compression. However, for compressed data the number of packets delivered is more at all bandwidths than for uncompressed data except at 70% bandwidth (1664/2109). The number of packets dropped for compressed data is less than for uncompressed data at all simulation runs. The ratio of packets dropped over packets delivered (packets dropped/packets delivered) is 96%, 182%, 389%, 398%, and 742% for compressed data and for uncompressed data it is 598%, 398%, 742%, 1246%, and 1567%. We observe that the ratio of packets dropped over packets delivered of uncompressed data is very high. This can be attributed to network saturation.

9.4. Video

Digital images are another area for compression algorithms. Moving pictures are a popular medium for expression. The amount of information is staggering even after the relentless success of the magnetic storage industry. Good video signals change the image 20 to 30 times per second, which means that even a minute of video is made up of 1200 still images. There is redundant information in the images that can be excluded to compress the file. More importantly, the eye is often quite forgiving, so the reconstruction does not need to be perfect. The video algorithms are all lossy, and this allows them to save substantially more space.

There are a number of different video compression standards. Most of them are variations of the work of two different committees: The International Telephony Union's Specialist Group on Coding for Visual Telephony and the International Standards Organization's motion picture expert group, known as MPEG. The most popular standards are the ISO's MPEG-2 and the ITU's H.261. MPEG-2 has been chosen to simulate video application.

The compression rate for MPEG-2 is 30%. Video sampling dimension is 352 pixel x 240 line x 12 bits x 30 frames/second. The message size is 3800000 bytes or 30.4Mb, warmup length (specifies the time in the simulation when the application will begin to collect data) is 10 seconds and replication length (the number of seconds of simulated time during which statistics are collected for each report) is 50 seconds. The bandwidth used for video application simulation is OC-9, 466.560Mbps. Just as in the previous three applications, the distribution of the number of observations in an interval is Poisson distributed. And the parameter entered for the Poisson distribution is the interarrival time (IAT, the time from the start of one message to the start of the next). The total number of simulation runs are 10. Each run time is approximately 2-11 hours. The simulation run statistics for compressed and uncompressed data are shown in **Tables 9.44** and **9.45**, respectively.

Examination of the data in **Tables 9.44** and **9.45** show that as the bandwidth increases:

1. IAT decreases
2. Packets/Second increase (see **Figures 9.19** and **9.20**)
3. Bits/Second increase
4. Each simulation run time is approximately 2-11 hours and the message size used for Video is very large, 3.8MB.

5. The rate of transmission for compressed data at each bandwidth is less than for uncompressed data.

as was observed in E-mail, Audio, and Image applications. As may be observed from the tables, when LAT decreases the number of packets/second increases linearly in both data types. This is depicted in **Figures 9.19** and **9.20**.

Video (OC-9)	Bandwidth Used for Simulation				
MPEG-2 (30%)	50%	60%	70%	80%	90%
InterArrival Time (IAT)	0.167	0.142	0.125	0.111	0.1
Poisson Distribution (poi)					
Frames/Second(fps)	6	7	8	9	11
MBits/Second(Mbps)	182.4	212.80	243.20	273.60	334.40
Simulation Time (hh:mm)	2:10	3:35	4:30	5:00	7:00

Table 9. 44: Video Simulation Run Statistics for Compressed Data
Message Size: 3.8MB = 30.4 Mbits, Bandwidth: OC-9, 466.560Mbps,
Replication Length: 50s

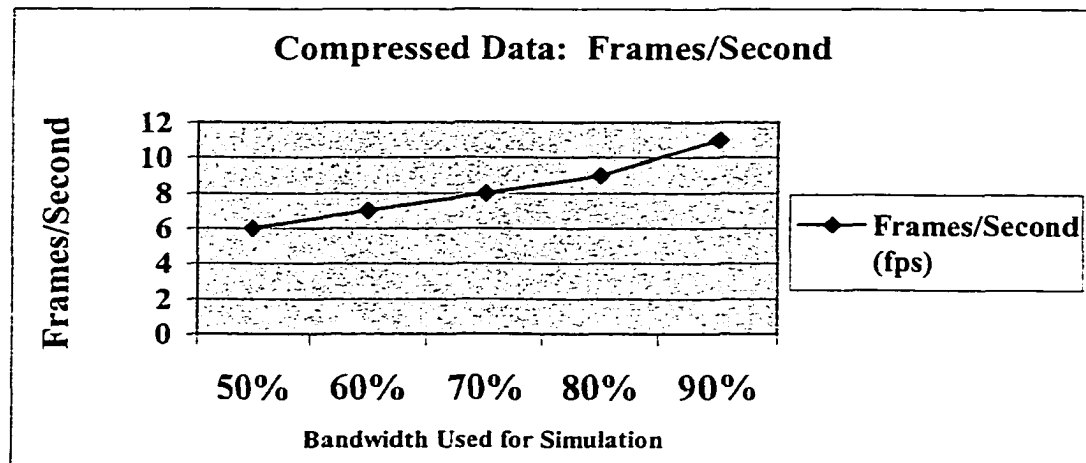


Figure 9. 19: Video - Compressed Data of Video: Frames/Second

Video (OC-9)	Bandwidth Used for Simulation				
MPEG-2 (30%)	50%	60%	70%	80%	90%
InterArrival Time (IAT)	0.125	0.111	0.091	0.083	0.071
Poisson Distribution (poi)					
Packets/Second(pps)	8	9	11	12	14
Bits/Second(bps)	243.20	273.60	334.40	364.80	425.60
Simulation Time (min)	4:30	5:00	7:00	9:20	10:30

Table 9. 45: Video Simulation Run Statistics for Uncompressed Data
Message Size: 3.8MB = 30.4 Mbits, Bandwidth: OC-9, 466.560Mbps,
Replication Length: 50s

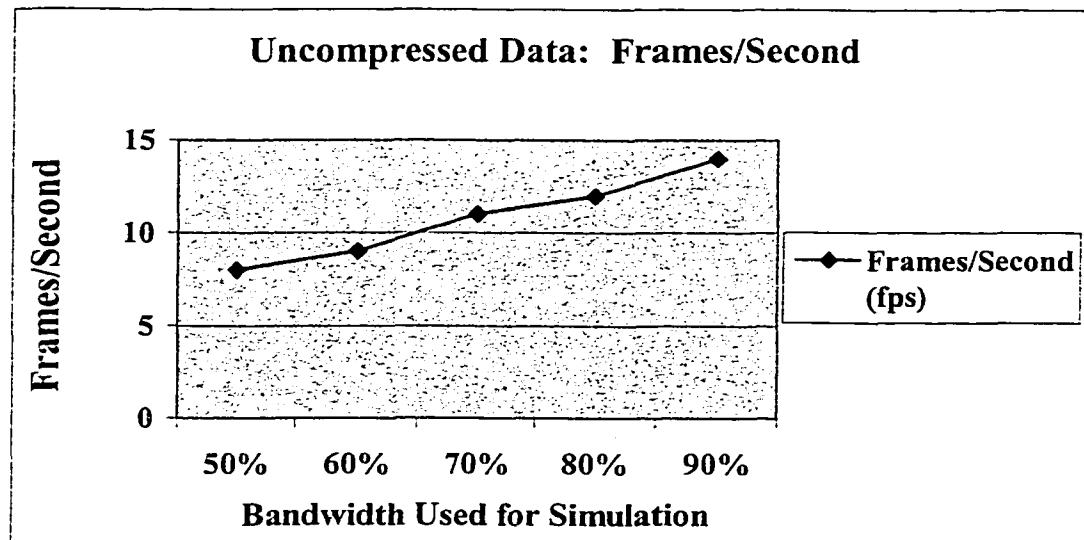


Figure 9. 20: Video - Uncompressed Data of Video: Frames/Second

MPEG-2

MPEG-2 (approved by ITU-T, November 1994) is chosen for video compression because it is the standard of video for CD-ROMs and certain TV signals. Reports are produced automatically at the end of each simulation run. COMNET III generates a statistical report of Link Utilization for various processing nodes as shown in Tables 9.46 and 9.47 for compressed and uncompressed data, respectively.

Analysis

- **Compressed Data**

The number of frames delivered from CUNY to Milano is 183553, 164897, 176283, 248857, and 220057 and the number of packets delivered by the same link is 20186, 18135, 19386, 27371, and 24203 at 50%, 60%, 70%, 80%, and 90% bandwidths, respectively. From 50% to 60% we see a decrease of 10% in the number of frames/packets delivered. From 60% to 70%, the number of frames/packets delivered increased by 7% and again from 70% to 80% the number increased by 41%. But, from 80% to 90%, we see a decrease of 11%. The slope of the curve is negative for 50% to 60% bandwidth simulation runs, positive for 60% to 80% and again negative for 80% to 90%, in each category, packets/second, bytes delivered, kilobits per second (kbps), and link utilization

percentage. The number of frames/packets delivered increases from 50% to 70%, decreases at 80% and again increases at 90% for the remaining three links: from CUNY to Princeton; from Montpelier to Princeton and from Milano to Montpelier. The same holds for the other categories. These statistics are shown in **Table 9.46**.

Video (OC-9)	Compressed Data of Video				
MPEG-2 (30%)	Bandwidth Used For Simulation				
Link	50%	60%	70%	80%	90%
From CUNY-Milano					
Frames Delivered	183553	164897	176283	248857	220057
Packets Delivered	20186	18135	19386	27371	24203
Packets/second	403.720	362.700	387.720	547.420	484.060
Bytes Delivered	127842400	1148520000	1227784000	1733384000	1532752000
KBPS Delivered	204547.8400	183763.200	196445.440	277341.440	245240.320
Utilization (%)	45.89	41.22	44.07	62.21	55.01
From CUNYPrincet					
Frames Delivered	138493	217273	244738	244102	258457
Packets Delivered	15230	23899	26918	26848	28430
Packets/second	304.600	477.980	538.360	536.960	568.600
Bytes Delivered	964560000	1513376000	1704672000	1700232000	1800240000
KBPS Delivered	154329.600	242140.160	272747.520	272037.120	288038.400
Utilization (%)	34.62	54.32	61.18	61.03	64.61
From MontpelierPri					
Frames Delivered	137684	201653	216390	192588	268179
Packets Delivered	15141	22178	23799	21181	29495
Packets/second	302.820	443.560	475.980	423.620	589.900
Bytes Delivered	958904000	1404512000	1507176000	1341384000	1868000000
KBPS Delivered	153424.64	224721.920	241148.160	214621.440	298880.000
Utilization (%)	34.42	50.41	54.10	48.15	67.04
From Milano-Mont					
Frames Delivered	179203	213813	261221	228018	236682
Packets Delivered	19711	23514	28748	25084	26030
Packets/second	394.220	470.280	574.960	501.680	520.600
Bytes Delivered	1248144000	1489216000	1819552000	1588216000	1648560000
KBPS Delivered	199703.040	238274.560	291128.320	254114.560	263769.600
Utilization (%)	44.80	53.45	65.31	57.00	59.17

Table 9. 46: Video - Links: Channel Utilization/Utilization by Application for Compressed Data

- **Uncompressed Data**

In case of uncompressed data, the number of frames delivered from CUNY to Milano is 176283, 248857, 279433, 269437, and 321793 and the number of packets delivered is 19386, 27371, 30740, 29638, and 35396 at 50%, 60%, 70%, 80%, and 90% bandwidths, respectively. We see an increase of 41% and 12% at 60% and 70%, respectively, in frames/packets delivered. At 80% the number of frames/packets delivered decreases by approximately 4%. And 90% the number of frames/packets delivered increases by 19%. Similarly, the link utilization percentage is upward from 50% to 70%, downward at 80%, and again upward at 90%. In case of from CUNY to Princeton link, the number of frames/packets delivered goes down by 0.25% at 60% bandwidth and increases at 70% to 90%. From Montpelier to Princeton link, the number of frames/packets delivered decreases by 11% at 60% bandwidth and increases at 70% to 90% bandwidths. From Milano to Montpelier link the decrease is 12% and 10% at 60% and 70% bandwidths, respectively. At 80% the number of frames/packets delivered increases by 61% and at 90% decreases by nearly 8%. These changes also reflect in other categories, packets/second, bytes delivered, kbps delivered, and utilization percentage as shown in **Table 9.47**.

Video (OC-9)	Uncompressed Data of Video				
MPEG-2 (30%)	Bandwidth Used For Simulation				
Link	50%	60%	70%	80%	90%
From CUNY-Milano					
Frames Delivered	176283	248857	279433	269437	321793
Packets Delivered	19386	27371	30740	29638	35396
Packets/second	387.720	547.420	614.800	592.760	707.920
Bytes Delivered	1227784000	1733384000	1946400000	1876792000	2241544000
KBPS Delivered	196445.440	2777341.440	311424.000	300286.720	358647.040
Utilization (%)	44.07	62.21	69.86	67.36	80.45
From CUNYPrincet					
Frames Delivered	244738	244102	295607	307965	346486
Packets Delivered	26918	26848	32514	33875	38118
Packets/second	538.360	536.960	650.280	677.500	762.360
Bytes Delivered	1704672000	1700232000	2059056000	2145160000	2413512000
KBPS Delivered	272747.520	272037.120	329448.960	343225.600	386161.920
Utilization (%)	61.18	61.030	73.900	76.990	86.620
From MontpelierPri					
Frames Delivered	216390	192588	222386	291443	320897
Packets Delivered	23799	21181	24460	32056	35299
Packets/second	475.980	423.620	489.200	641.120	705.980
Bytes Delivered	1507176000	1341384000	1549000000	2030024000	2235216000
KBPS Delivered	241148.160	214621.440	247840.000	324803.840	357634.560
Utilization (%)	54.10	48.15	55.60	72.86	80.22
From Milano-Mont					
Frames Delivered	261221	228018	205988	332239	307268
Packets Delivered	28748	25084	22657	36573	33817
Packets/second	574.960	501.680	453.140	731.460	676.340
Bytes Delivered	1819552000	1588216000	1434688000	2314312000	2140288000
KBPS Delivered	291128.320	254114.560	229550.080	370289.920	342446.080
Utilization (%)	65.31	57.00	51.5	83.06	76.82

Table 9. 47: Video - Links: Channel Utilization/Utilization by Application for Uncompressed Data

First row in **Tables 9.46** and **9.47** represents packets delivered during the simulation. They fluctuate for some links and increase as the bandwidth increases for others. This fluctuation could be a function of network delay and/or link/node failure.

Fluctuations in:

- Packets/Second (Packets Delivered/Replication Length, 50s);
- Bytes Delivered (Packets Delivered * Message Size);
- KBPS (kilobits/second, Packets/Second * Message Size * 8bits)
- Link Utilization

can be seen as reflections of the fluctuations in packets delivered.

Comparison of the results shown in **Table 9.48** demonstrates that the utilization rate for compressed data is lower than for uncompressed data in most cases. This implies that compression permits transmission of larger volume of data in shorter intervals.

Video	Link Utilization (%)	
Link	Compressed Data	Uncompressed Data
From CUNY-Milano		
50%	45.89	44.07
60%	41.22	62.21
70%	44.07	69.86
80%	62.21	67.36
90%	55.01	80.45
From CUNYPrinceton		
50%	34.62	61.18
60%	54.32	61.03
70%	61.18	73.90
80%	61.03	76.99
90%	64.41	86.62
From MontpelierPri		
50%	34.42	54.10
60%	50.41	48.15
70%	54.10	55.60
80%	48.15	72.86
90%	67.04	80.22
From Milano-Mont		
50%	44.80	65.31
60%	53.45	57.00
70%	65.31	51.50
80%	57.00	83.06
90%	59.17	76.82

Table 9. 48: Video - Comparison of Utilization (%) for Compressed & Uncompressed Data

WAN Clouds: Access Link Statistics Analysis

Tables 9.49, 9.50, 9.51, and 9.52 present statistics for compressed and uncompressed data for 7 access links: (1) CUNY-Montpelier, (2) CUNY-Milano, (3) CUNY-Bern, (4) CUNY-Bonn, (5) Princeton-Montpelier, (6) CUNY-Princeton, and (7) Montpelier-Milano. Only three links are chosen for analysis to demonstrate the pattern of variation over access links.

Compressed Data	Frames Accepted				
Access Link	50%	60%	70%	80%	90%
CUNY-Montpelier	15991	19597	20839	26006	26068
CUNY-Milano	18087	16232	17445	24557	21890
CUNY-Bern	15997	18008	21389	22348	24771
CUNY-Bonn	13451	19319	26330	20722	27893
Princeton-Montpelier	13645	19896	21419	19019	26403
CUNY-Princeton	13742	21568	24325	24139	25574
Montpelier-Milano	17664	21076	25852	22544	23406

Table 9. 49: Video - WAN Cloud Access Link Statistics: Frames Accepted (Compressed Data)

Compressed Data	Frames Dropped				
Access Link	50%	60%	70%	80%	90%
CUNY-Montpelier	1868	2082	2421	2818	2835
CUNY-Milano	2097	1905	1939	2816	2311
CUNY-Bern	1763	2052	2430	2530	2733
CUNY-Bonn	1509	2261	2952	2300	3146
Princeton-Montpelier	1498	2282	2380	2164	3092
CUNY-Princeton	1490	2334	2673	2693	2854
Montpelier-Milano	2047	2436	2896	2540	2624

Table 9. 50: Video - WAN Cloud Access Link Statistics: Frames Dropped (Compressed Data)

Uncompressed Data	Frames Accepted				
Access Link	50%	60%	70%	80%	90%
CUNY-Montpelier	20839	26006	29989	31754	30578
CUNY-Milano	17445	24557	27570	26608	31898
CUNY-Bern	21389	22348	26344	28334	28380
CUNY-Bonn	26330	20722	27937	30265	32306
Princeton-Montpelier	21419	19019	21968	28808	31785
CUNY-Princeton	24325	24139	29207	30481	34164
Montpelier-Milano	25852	22544	20389	32888	30344

Table 9. 51: Video - WAN Cloud Access Link Statistics: Frames Accepted (Uncompressed Data)

Uncompressed Data	Frames Dropped				
Access Link	50%	60%	70%	80%	90%
CUNY-Montpelier	2421	2818	3350	3408	3385
CUNY-Milano	1939	2816	3170	3030	3500
CUNY-Bern	2430	2530	2880	3190	3052
CUNY-Bonn	2952	2300	3105	3397	3540
Princeton-Montpelier	2380	2164	2492	3246	3514
CUNY-Princeton	2673	2693	3261	3400	3896
Montpelier-Milano	2896	2540	2266	3687	3471

Table 9. 52: Video - WAN Cloud Access Link Statistics: Frames Dropped (Uncompressed Data)

- **Compressed Data**

The number of frames accepted by CUNY-Montpelier access link are 15991, 19597, 20839, 26006, and 26068 at 50%, 60%, 70%, 80%, and 90% bandwidths, as shown in **Figure 9.21**. The number of dropped frames at each bandwidth percentage is 1868 at 50%, 2082 at 60%, 2421 at 70%, 2818 at 80% and 2835 at 90%, as shown in **Figure 9.22**. The numbers of frames accepted and dropped by the link show that the increase from one bandwidth to another is upward and the slope is positive. The drop rate is 12%, 11%, 12%, 11%, and 11% for all five different bandwidth simulation runs, respectively.

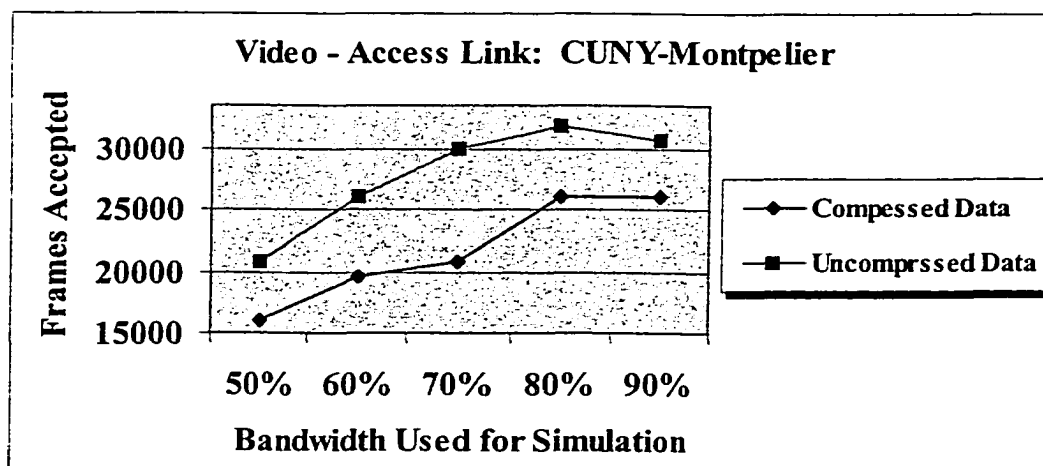


Figure 9. 21: Video– Selected Access Link: CUNY-Montpelier (Frames Accepted)

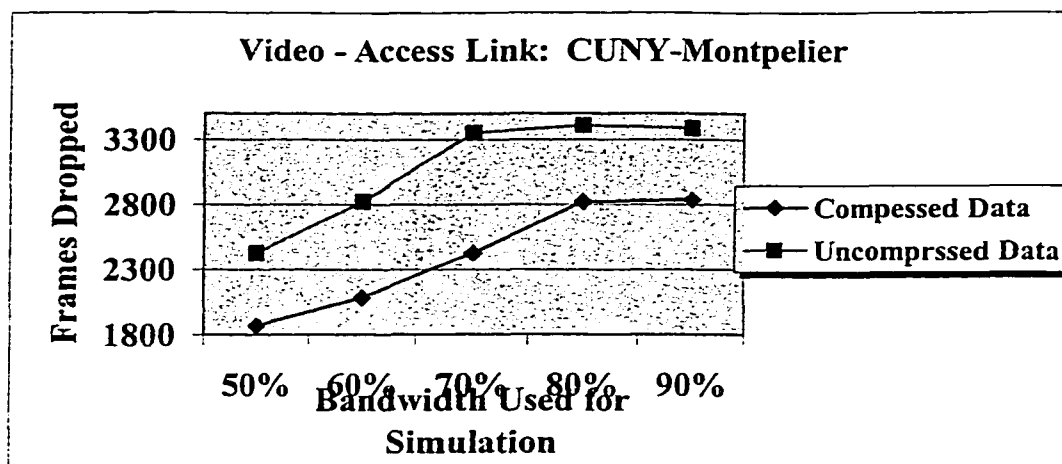


Figure 9. 22: Video– Selected Access Link: CUNY-Montpelier (Frames Dropped)

CUNY-Bonn access link accepted 13451 frames at 50% and 19391 frames at 60%, and 26330 frames at 70% bandwidths. At 80% the number of frames accepted by the link went down to 20722, a decrease of 21%. The number of frames accepted at 90% went up to 27893, an increase of 35%. The number of dropped frames at each bandwidth percentage is 1509 at 50%, 2261 at 60%, 2952 at 70%, 2300 at 80% and 3146 at 90%. The drop rate is 11%, 12%, 11%, 11%, and 11% for all five different bandwidth simulation runs, respectively.

The number of frames accepted by Montpelier-Milano access link for compressed data at different bandwidth simulation runs are 17664, 21076, 25852, 22544, and 23406. The number of accepted frames decreases at 80% by nearly

13% and again increases at 90% by 4%. The number of dropped frames at each bandwidth percentage is 2047 at 50%, 2436 at 60%, 2896 at 70%, 2540 at 80% and 2624 at 90%. The drop rate ranges from 11% to 12%.

- **Uncompressed Data**

The number of frames accepted by CUNY-Montpelier link is 20839, 26006, 29989, 31754, and 30578, as presented in **Table 9.51**. From 50% to 80% bandwidths, the numbers indicate that the increase is linear. However, at 90% the number of frames accepted by the link goes down to 30578, a decrease of 4%. The number of dropped frames at each bandwidth percentage is 2421 at 50%, 2818 at 60%, 3350 at 70%, 3408 at 80% and 3385 at 90%. At 90% the number of frames dropped decreases by 0.7%. The drop rate is 12%, 11%, 11%, 11%, and 11% for all five different bandwidth simulation runs, respectively. **Figures 9.21** and **9.22** depict the number of frames accepted and dropped, respectively.

The number of frames accepted by CUNY-Bonn access link are 26330, 20722, 27937, 30265, and 32306 at 50%, 60%, 70%, 80%, and 90% bandwidth simulation runs, respectively. These numbers indicate that at 60% the number of frames accepted by the link decreases by 21%. From 70% to 90% the increase is upward. The number of dropped frames at each bandwidth percentage is 2952 at 50%, 2300 at 60%, 3105 at 70%, 3397 at 80% and 3540 at 90%. At 60% the

number of frames dropped by the link decreases by 22%. The drop rate is 11% for all five different bandwidth simulation runs.

The number of frames accepted by Montpelier-Milano access link for uncompressed data at different bandwidth simulation runs are 25852, 22544, 20389, 32888, and 30344. We observe that at 60% and 70% the number of frames accepted by the link decreases by approximately 13% and 10%, respectively. At 80% there is an increase of 61% and a decrease of 8% at 90%. The number of dropped frames at each bandwidth percentage is 2896 at 50%, 2540 at 60%, 2266 at 70%, 3687 at 80% and 3471 at 90%. The decrease is 12% and 11% at 60% and 70% bandwidths, respectively. The drop rate is 11% for all five different bandwidth simulation runs.

Comparison of Frames-Dropped

The drop rate of compressed and uncompressed data and their differences are shown in **Table 9.53**. Also shown in the table is the comparison of the link utilization rate for both data types.

The drop rate of CUNY-Montpelier access link at 50% bandwidth for compressed data is equal to that of uncompressed data, 12%. Similarly, the drop

rates are equal for the remaining bandwidth percentages, 60%-90%. The link utilization rate for compressed data for all bandwidths is less than for uncompressed data.

The drop rate of CUNY-Bonn access link at 50%, 70%-90% bandwidths for compressed and uncompressed data is 11%. The difference between the two is 0. At 60% bandwidth the difference is 1%. These differences are well within the margin of error, $\pm 5\%$. The link utilization rate for compressed data is 32.50%, 46.87%, 63.60%, 50.01% and 67.42% at 50%-90% bandwidths. The link utilization for uncompressed data is 63.60%, 50.01%, 67.43%, 73.11% and 77.85% at 50%-90% bandwidths. We note that the link utilization rate of uncompressed data at 50% and 60% bandwidths is equal to that of compressed data at 70% and 80% bandwidths, respectively. Clearly, the link utilization rate for compressed data is less than for uncompressed data.

The drop rate of Montpelier-Milano access link of compressed and uncompressed data is approximately 11%. The link utilization rate overall for compressed data is less than that of uncompressed data with one exception- at 70% bandwidth.

This demonstrates that while the drop rate is more or less the same for both compressed and uncompressed data, compressing data has a clear and substantive advantage in terms of the link utilization rate. Thus, data compression allows for transmission of substantially larger volume of data in shorter intervals.

Video (OC-9)	WAN CLOUDS: ACCESS LINK STATS (ENTRY)									
MPEG-2(30%)	Comparison of Compressed & Uncompressed Video Data									
Message Size: 3.8MB	Frames Accepted/Dropped/Utilization									
Cloud: Access Link	Cmp 50%	Uncmp 50%	Cmp 60%	Uncmp 60%	Cmp 70%	Uncmp 70%	Cmp 80%	Uncmp 80%	Cmp 90%	Uncmp 90%
CUNY-Montpelier										
Frames Accepted	15991	20839	19597	26006	20839	29989	26006	31754	26068	30578
Frames Dropped	1868	2421	2082	2818	2421	3350	2818	3408	2835	3385
Drop Rate	0.1168	0.11617	0.1062	0.10835	0.1161	0.11170	0.1083	0.10732	0.1087	0.11070
Drop Rate (%)	12%	12%	11%	11%	12%	11%	11%	11%	11%	11%
Difference	0%		-0%		0%		0%		-0%	
Link Utilization (%)	38.80	50.52	47.10	62.61	50.52	72.42	62.61	76.36	62.78	73.76
CUNY-Milano										
Frames Accepted	18087	17445	16232	24557	17445	27570	24557	26608	21890	31898
Frames Dropped	2097	1939	1905	2816	1939	3170	2816	3030	2311	3500
Drop Rate	0.1159	0.11114	0.1173	0.11467	0.1111	0.11498	0.1146	0.11387	0.1055	0.10972
Drop Rate (%)	12%	11%	12%	11%	11%	11%	11%	11%	11%	11%
Difference	0%		0%		-0%		0%		-0%	
Link Utilization (%)	43.85	42.11	39.40	59.46	42.11	66.76	59.46	64.37	52.57	76.89

Video (OC-9)	Frames Accepted/Dropped/Utilization (Cont'd.)									
Cloud: Access Link	Cmp 50%	Uncmp 50%	Cmp 60%	Uncmp 60%	Cmp 70%	Uncmp 70%	Cmp 80%	Uncmp 80%	Cmp 90%	Uncmp 90%
CUNY-Bern										
Frames Accepted	15997	21389	18008	22348	21389	26344	22348	28334	24771	28380
Frames Dropped	1763	2430	2052	2530	2430	2880	2530	3190	2733	3052
Drop Rate	0.1102	0.11360	0.1139	0.11320	0.1136	0.10932	0.1132	0.11258	0.1103	0.10754
Drop Rate (%)	11%	11%	11%	11%	11%	11%	11%	11%	11%	11%
Difference	-0%		0%		0%		0%		0%	
Link Utilization (%)	38.58	51.74	43.57	54.04	51.74	63.47	54.04	68.47	59.74	68.27
CUNY-Bonn										
Frames Accepted	13451	26330	19319	20722	26330	27937	20722	30265	27893	32306
Frames Dropped	1509	2952	2261	2300	2952	3105	2300	3397	3146	3540
Drop Rate	0.1121	0.11211	0.1170	0.11099	0.1121	0.11114	0.1109	0.11224	0.1127	0.10957
Drop Rate (%)	11%	11%	12%	11%	11%	11%	11%	11%	11%	11%
Difference	0%		1%		0%		-0%		0%	
Link Utilization (%)	32.50	63.60	46.87	50.01	63.60	67.43	50.01	73.11	67.42	77.85
Princeton-Montpelier										
Frames Accepted	13645	21419	19896	19019	21419	21968	19019	28808	26403	31785
Frames Dropped	1498	2380	2282	2164	2380	2492	2164	3246	3092	3514
Drop Rate	0.1097	0.11111	0.1146	0.11378	0.1111	0.11343	0.1137	0.11267	0.1171	0.11055
Drop Rate (%)	11%	11%	11%	11%	11%	11%	11%	11%	12%	11%
Difference	-0%		0%		-0%		0%		1%	
Link Utilization (%)	32.89	51.70	48.17	46.01	51.70	53.13	46.01	69.63	64.07	76.67

Video (OC-9)	Frames Accepted/Dropped/Utilization (Cont'd.)									
Cloud: Access Link	Cmp 50%	Uncmp 50%	Cmp 60%	Uncmp 60%	Cmp 70%	Uncmp 70%	Cmp 80%	Uncmp 80%	Cmp 90%	Uncmp 90%
CUNY-Princeton										
Frames Accepted	13742	24325	21568	24139	24385	29207	24139	30481	25574	34164
Frames Dropped	1490	2673	2334	2693	2673	3261	2693	3400	2854	3896
Drop Rate	0.1084	0.10988	0.1082	0.11156	0.1098	0.11165	0.1115	0.11154	0.1115	0.11403
Drop Rate (%)	11%	11%	11%	11%	11%	11%	11%	11%	11%	11%
Difference	-0%		-0%		-0%		0%		-0%	
Link Utilization (%)	33.09	58.64	51.92	58.28	58.64	70.52	58.28	73.59	61.75	82.66
Montpelier-Milano										
Frames Accepted	17664	25852	21076	22544	25852	20389	22544	32888	23406	30344
Frames Dropped	2047	2896	2436	2540	2896	2266	2540	3687	2624	3471
Drop Rate	0.1158	0.11202	0.1155	0.11266	0.1120	0.11113	0.1126	0.11210	0.1121	0.11438
Drop Rate (%)	12%	11%	12%	11%	11%	11%	11%	11%	11%	11%
Difference	0%		0%		0%		0%		-0%	
Link Utilization (%)	42.81	62.41	51.08	54.48	62.41	49.21	54.48	79.38	56.55	73.41

Table 9. 53: Video - WAN Cloud: Access Link Statistics- Frames Accepted, Frames Dropped & Link Utilization

Message & Response Sources

Table 9.54 shows data for 6 distinct origin nodes that send messages to 6 different destination nodes. Of these, only two nodes are chosen for analysis that follows:

CUNY-Milano/CUNY-Princeton (Origin/Destination)

Compressed Data

CUNY-Milano, origin node, created 20580, 18060, 19980, 29100, and 26040 to send to CUNY-Princeton, destination node, at 50%, 60%, 70%, 80%, and 90% bandwidth, respectively. The number of packets created decreases by 12% at 60% bandwidth, increases linearly at 70% and 80% bandwidths, and again decreases at 90% by nearly 11%. The number of packets delivered at the destination is 8382, 6162, 6572, 6095, and 5385 at the five simulation runs, respectively. And the number of packets dropped similarly is 3061, 2583, 2648, 5263, and 4690. The number of packets delivered and packets dropped show a pattern of oscillation, as may be observed from **Table 9.54**. The average packet delay (in milliseconds) also oscillates over the range of the bandwidths.

Uncompressed Data

CUNY-Milano, origin node, created 19980, 29100, 35100, 33600, and 41280 to send to CUNY-Princeton, destination node, at 50%, 60%, 70%, 80%, and 90% bandwidth, respectively. The number of packets created increases from 50% to 70% bandwidths linearly, decreases at 80% by 4%, and at 90% bandwidth increases by 23%. The number of packets delivered at the destination is 6572, 6095, 8910, 6306, and 7488 at the five simulation runs, respectively. These numbers show a decrease from 6572 packets delivered at 50% to 6095 packets delivered at 60%, a decrease of 7%. This shows that the number of packets delivered oscillates over the range of the bandwidths. And the number of packets dropped is 2648, 5263, 8200, 7680, and 10575 at 50%, 60%, 70%, 80%, and 90% bandwidths, respectively. The average packet delay (in milliseconds) increases from 50% bandwidth to 70% bandwidth, decreases at 80%, and again increases at 90%.

Comparison of Compressed and Uncompressed Data

The number of packets created for compressed data is nearly 3% more at 50% bandwidth (20580/19980) than that of uncompressed data and less for the rest of the simulation runs, as shown in **Table 9.54**. The number of packets delivered also is about 2% more at 50% (8382/6572) and 1% more at 60%

(6162/6095) bandwidths. And at 70%-90% bandwidths the number of packets delivered is less (6572/8910, 6095/6306, 5385/7488). The number of packets dropped is more at 50% and less for 60%-90% simulation runs. The ratio of packets dropped over packets delivered (packets dropped/packets delivered) is 37% (3061/8382), 42% (2583/6162), 40% (2648/6572), 86% (5263/6095), and 87% (4690/5385) for compressed data and it is 40% (2648/6572), 86% (5263/6095), 92% (8200/8910), 122% (7680/6306), and 141% (7488/10575) for uncompressed data. We see high percentages at 60%-90% bandwidths for uncompressed data. This can be attributed to network saturation. The average packet delay (in milliseconds) for compressed data is less than for uncompressed data for all 5 bandwidth simulation runs.

	Message & Response Sources: Packets Delivered/Packet Delay									
Video (OC-9)	Compressed Data					Uncompressed Data				
MPEG-2 (30%)	Bandwidth Used For Simulation									
Origin/Destination	50%	60%	70%	80%	90%	50%	60%	70%	80%	90%
CUNY-Milano/ CUNY-Princeton										
Packets Created	20580	18060	19980	29100	26040	19980	29100	35100	33600	41280
Packets Delivered	8382	6162	6572	6095	5385	6572	6095	8910	6306	7488
Packets Dropped	3061	2583	2648	5263	4690	2648	5263	8200	7680	10575
Dropped/Created	15%	14%	13%	18%	18%	13%	18%	23%	23%	26%
Dropped/Delivered	37%	42%	40%	86%	87%	40%	86%	92%	122%	141%
Packet Delay (MS)										
Average	21545.220	23800.986	23152.951	24430.271	24285.017	23152.951	24430.271	26853.650	26317.635	27174.642
Maximum	35761.646	39753.589	38442.972	41754.478	15937.652	38442.972	41754.478	43751.467	46754.748	45997.231
MontpelierP/ Montpelier-Milano										
Packets Created	15000	24180	24120	20820	36540	24120	20820	25980	34440	39060
Packets Delivered	5290	7339	5859	5591	5199	5859	5591	4981	4806	6078
Packets Dropped	2096	5070	3391	3019	10755	3391	3019	4544	5449	8019
Dropped/Created	14%	21%	14%	15%	29%	14%	15%	17%	16%	21%
Dropped/Delivered	40%	69%	58%	54%	207%	58%	54%	91%	113%	132%
Packet Delay (MS)										
Average	18937.367	24205.829	24237.068	24170.535	25957.609	24237.068	24170.535	24086.892	23164.516	29016.788
Maximum	34686.021	39251.421	37752.701	41923.018	45998.479	37752.701	41923.018	42185.979	45995.462	46195.858

	Message & Response Sources: Packets Delivered/Packet Delay (Cont'd.)									
Video (OC-9)	Compressed Data					Uncompressed Data				
MPEG-2 (30%)	Bandwidth Used For Simulation									
Origin/Destination	50%	60%	70%	80%	90%	50%	60%	70%	80%	90%
CUNY-Bern/ Bern-Bonn										
Packets Created	18660	19680	25260	24600	31680	25260	24600	33420	40020	35880
Packets Delivered	4710	5234	5443	5940	7801	5443	5940	3940	4608	5985
Packets Dropped	3190	2661	4471	3607	8108	4471	3607	7526	11571	8189
Droppped/Created	17%	14%	18%	15%	26%	18%	15%	23%	29%	23%
Dropped/Delivered	68%	51%	82%	61%	104%	82%	61%	191%	251%	137%
Packet Delay (MS)										
Average	18316.892	22700.776	21987.922	22305.040	27316.763	21987.922	22305.040	27282.640	27406.769	25983.689
Maximum	34815.675	38995.320	37999.714	41998.341	44995.186	37999.714	41998.341	42999.850	45999.167	45915.336
CUNY-Bonn/ Milano-Bonn										
Packets Created	15060	21900	30060	26460	33360	30060	26460	34320	39840	45780
Packets Delivered	4782	4719	6200	6663	5494	6200	6663	5142	5790	5873
Packets Dropped	2034	3125	4476	6490	6439	4476	6490	6549	10872	13719
Dropped/Created	14%	14%	15%	25%	19%	15%	25%	19%	27%	30%
Dropped/Delivered	43%	66%	72%	97%	117%	72%	97%	127%	188%	234%
Packet Delay (MS)										
Average	22132.069	23018.402	23614.416	24479.970	28458.794	23614.416	24479.970	25639.160	27681.452	26808.653
Maximum	33390.164	38994.223	37864.572	40783.982	43289.376	37864.572	40783.982	42996.558	45989.974	44777.672

	Message & Response Sources: Packets Delivered/Packet Delay (Cont'd.)									
Video (OC-9)	Compressed Data					Uncompressed Data				
MPEG-2 (30%)	Bandwidth Used For Simulation									
Origin/Destination	50%	60%	70%	80%	90%	50%	60%	70%	80%	90%
CUNY-Montpelier/ Montpelier-Milano										
Packets Created	17940	22200	26160	32820	31020	26160	32820	35580	42240	39300
Packets Delivered	6662	3941	5134	5138	6174	5134	5138	6252	6315	4684
Packets Dropped	2578	2551	5883	7380	5012	5883	7380	5673	11238	9232
Dropped/Created	14%	11%	22%	22%	16%	22%	22%	16%	27%	23%
Dropped/Delivered	39%	65%	115%	144%	81%	115%	144%	91%	178%	197%
Packet Delay (MS)										
Average	19406.890	20115.726	18922.891	22990.946	27608.174	18922.891	22990.946	27288.050	28376.281	27056.934
Maximum	35180.295	38503.759	35627.469	41999.439	44995.872	35627.469	41999.439	42992.167	46997.383	45999.851
CUNYPrinceton/ Princeton-Mont										
Packets Created	15180	26040	27600	28500	31140	27600	28500	39300	40320	50520
Packets Delivered	4643	7026	6277	5270	4508	6277	5270	6610	6924	7160
Packets Dropped	1994	5261	3902	4755	6001	3902	4755	10272	10711	16368
Dropped/Created	13%	20%	14%	17%	19%	14%	17%	26%	27%	32%
Dropped/Delivered	43%	75%	62%	90%	133%	62%	90%	155%	155%	229%
Packet Delay (MS)										
Average	20435.045	27705.592	21806.770	24620.266	23439.293	21806.770	24620.266	23334.466	26711.550	28540.180
Maximum	35148.382	39613.026	38147.012	41295.547	42565.315	38147.012	41295.547	43147.148	46133.979	46994.363

Table 9. 54: Video - Message & Response Sources: Packets Delivered/Dropped/Delay

MontpelierP/Montpelier-Milano (Origin/Destination)

Compressed Data

MontpelierP, origin node, created packets 15000, 24180, 24120, 20820, and 36540 to send to Montpelier-Milano, destination node, at 50%, 60%, 70%, 80%, and 90% bandwidths, respectively. The number of packets delivered at the destination is 52900, 7339, 5859, 5591, and 5199 at the five simulation runs, respectively. And the number of packets dropped similarly is 2096, 5070, 3391, 3019, and 10755. The number of packets created decreases at 70% and 80% bandwidths and increases at 90%. But, the number of packets delivered decreases linearly from 60% to 90% bandwidths. The number of packets dropped by the network decreases at 70% and 80% bandwidths and increases at 90%. The average packet delay (in milliseconds) increases linearly from 50% to 70%, decreases at 80%, and again increases at 90% bandwidths

Uncompressed Data

MontpelierP, origin node, created 24120, 20820, 25980, 34440, and 39060 to send to Montpelier-Milano, destination node, at 50%, 60%, 70%, 80%, and 90% bandwidths, respectively. The number of packets delivered at the destination is 5859, 5591, 4981, 4806, and 6078 at the five simulation runs, respectively. And the number of packets dropped similarly is 3391, 3019, 4544, 5449, and 8019. The number of packets created by the origin node decreases at

60% bandwidth and increases linearly from 60% to 90% bandwidths. The ratio of packets dropped over packets delivered is 58% (3391/5859), 54% (3019/5591), 91% (4544/4981), 113% (5449/4806), and 132% (8019/6078) at 50%-90% bandwidths. The average packet delay (in milliseconds) decreases linearly from 50% to 80% and increases at 90% bandwidth.

Comparison of Compressed and Uncompressed Data

The number of packets created for compressed data is more than for uncompressed data at 60% bandwidth (24180/20820) and less for the rest of the simulation runs than for uncompressed data. However, for compressed data the number packets delivered is more for 60%-80% bandwidths (7339/5591, 5859/4981 and 5591/4806). The number of packets dropped for compressed data is more than for uncompressed data at 60% and 90% bandwidths (5070/3019, 10755/8019) and less for the rest of the simulation runs. The ratio of packets dropped over packets delivered (packets dropped/packets delivered) is 40%, 69%, 58%, 54%, and 207% for compressed data and for uncompressed data it is 58%, 54%, 91%, 113%, and 132%. We observe that the ratio of packets dropped over packets delivered is more at 60% and 90% bandwidths for compressed data than for uncompressed data.

9.5. HDTV

HDTV (High-Definition TV) is not the same as DTV (digital TV). The high-definition was initially achieved by an analog signal, but now is standardized as a digital format. Initially HDTV was IDTV (I was for improved), but the 1000-line threshold of vertical resolution, as high as 1280 lines, produced twice the resolution. HDTV standards began evolving in the early 1980s and were finalized in the early 1990s. Similar in nature to the video standards HDTV standards increase the number of pixels per line and the number of lines in the image frame, 1920 and 1080, respectively. HDTV standards put the screen width at 16:9 aspect ratio to emulate the wider motion picture screens. HDTV is the maximum resolution and the highest quality of video and sound. The huge bit rate prevents HDTV transmission unless signals are squeezed into a smaller space. In the nineties, advances in digital compression allowed the huge bit stream necessary for HDTV to fit into the 19 Mbps range possible within the confines of a 6-MHz channel. Compressed digital HDTV requires the full bandwidth of a television channel (6 MHz) to deliver the maximum pixel count of 2,073,600 (1920 x 1080).

Although it will take years for the commercial broadcast television industry to fully exploit HDTV, industry is already making widespread use of HDTV for applications requiring higher spatial resolution. The equipment costs, however, are still high. As the commercial television broadcaster and consumer demands increase over time, HDTV costs will drop.

MPEG-2 was originally designed for compressing broadcast quality video into 4 to 6 Mbps, so it could fit in a NTSC (National Television Standard Committee) or PAL (Phase Alternating Line) broadcast channel. Later, MPEG-2 was expanded to support higher resolution, including HDTV. Thus, MPEG-2 has been chosen to simulate HDTV application.

The compression rate for MPEG-2 is 30%. Video sampling dimension is 1920 pixel x 1080 line x 16 bits x 30 frames/second. The message size is 93.225MB or 745.8Mb, warmup length (specifies the time in the simulation when the application will begin to collect data) is 10 seconds and replication length (the number of seconds of simulated time during which statistics are collected for each report) is 50 seconds. The bandwidth used for HDTV application simulation is the same one used for Video simulation, OC-9, 466.560Mbps. Just as in the previous four applications, the distribution of the number of observations in an interval is Poisson distributed. And the parameter entered for the Poisson

distribution is the interarrival time (IAT, the time from the start of one message to the start of the next). The total number of simulation runs is 10. Each run time is approximately 6-14 hours. The simulation run statistics for compressed and uncompressed data are shown in **Tables 9.55** and **9.56**, respectively.

Examination of data in **Tables 9.55** and **9.56** show that as bandwidth increases:

1. IAT decreases
2. Packets/Second increase (see **Figures 9.23** and **9.24**)
3. Bits/Second increase
4. Each simulation run time is approximately 6-14 hours and the message size used for HDTV is 745.8MB.
5. The rate of transmission for compressed data at each bandwidth is less than for uncompressed data.

as was observed in E-mail, Audio, Image, and Video applications. As the tables show, when IAT decreases the number of packets/second increases linearly in both data types. This is depicted in **Figures 9.23** and **9.24**.

HDTV (OC-9)	Bandwidth Used for Simulation				
MPEG-2 (30%)	50%	60%	70%	80%	90%
InterArrival Time (IAT)	4.5	3.8	3.2	2.8	2.3
Poisson Distribution (poi)					
Frames/Second(fps)	0.22	0.26	0.3	0.35	0.37
MBits/Second(Mbps)	164.1	194.0	223.7	261.0	276.0
Simulation Time (hh:mm)	6:50	10:10	11:00	11:40	12:40

Table 9. 55: HDTV Simulation Run Statistics for Compressed Data
Message Size: 93.225MB = 745.8 Mbits, Bandwidth: OC-9, 466.560Mbps,
Replication Length: 50s

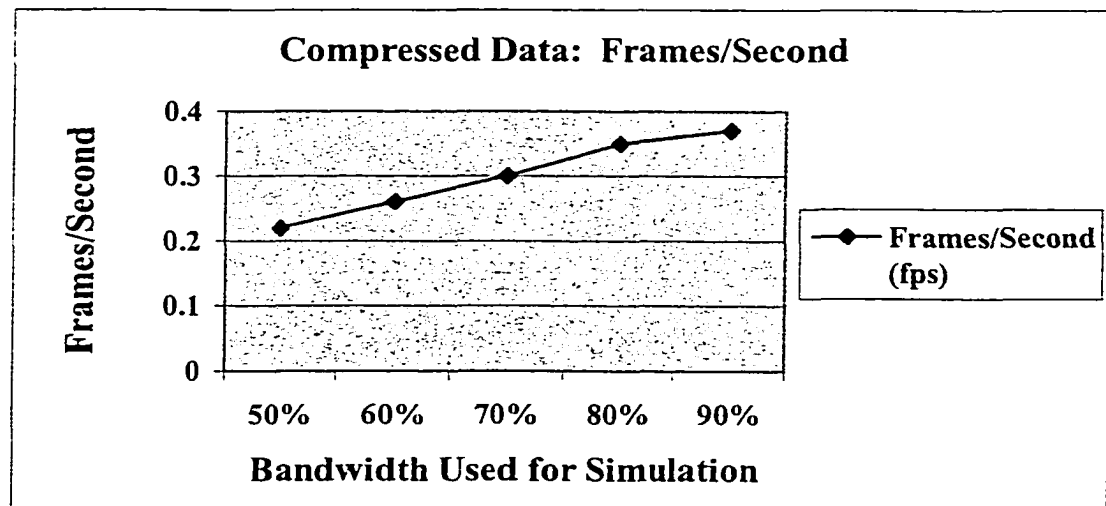


Figure 9. 23: HDTV - Compressed Data of HDTV: Frames/Second

HDTV (OC-9)	Bandwidth Used for Simulation				
MPEG-2 (30%)	50%	60%	70%	80%	90%
InterArrival Time (IAT)	3.2	2.7	2.3	2.0	1.8
Poisson Distribution (poi)					
Frames/Second(pps)	0.31	0.37	0.43	0.496	0.55
Bits/Second(bps)	231.2	276.0	320.6	370.0	410.2
Simulation Run Time (hh:mm)	11:00	12:40	12:35	13:05	14:00

Table 9. 56: HDTV Simulation Run Statistics for Uncompressed Data
Message Size: 93.225MB = 745.8 Mbits, Bandwidth: OC-9, 466.560Mbps,
Replication Length: 50s

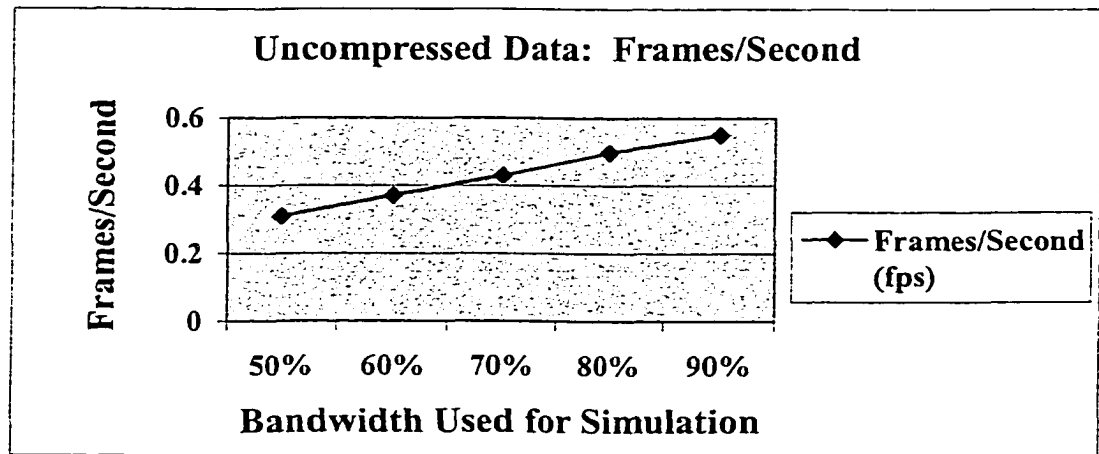


Figure 9. 24: HDTV - Uncompressed Data of HDTV: Frames/Second

MPEG-2

MPEG-2 (approved by ITU-T, November 1994) is chosen for HDTV compression because it is the standard for digital television. Reports are produced automatically at the end of each simulation run. COMNET III generates a statistical report of Link Utilization for various processing nodes as shown in Tables 9.57 and 9.58 for compressed and uncompressed data, respectively.

Analysis

- **Compressed Data**

The number of frames delivered from CUNY to Milano is 293857, 276887, 302949, 326672, and 356609 and the number of packets delivered by the same link is 31994, 30146, 32984, 35567, and 38828 at 50%, 60%, 70%, 80%, and 90% bandwidths, respectively. From 50% to 60% we see a decrease of 6% in the number of frames/packets delivered. We observe a linear increase in the number of frames/packets delivered from 60% to 90% bandwidths. From 60% to 70%, 70% to 80%, and 80% to 90% bandwidths, the number of frames/packets delivered increases by 9%, 8% and 10%, respectively. The slope of the curve is negative for 50% to 60% bandwidth simulation runs and positive for 60% to 90% bandwidths, in each category, packets/second, Bytes delivered, kilobits per second (kbps), and link utilization percentage. The number of frames/packets delivered

increases from 50% to 80% and decreases at 90% for links from CUNY to Princeton and from Milano to Montpelier. From Montpelier to Princeton link the number of frames/packets delivered increases from 50% to 60%, decreases at 70%, and again increases at 80% and 90% bandwidths. The same holds for the other categories. These statistics are shown in **Table 9.57**.

HDTV (OC-9)	Compressed Data				
MPEG-2 (30%)	Bandwidth Used For Simulation				
Link	50%	60%	70%	80%	90%
From CUNY-Milano					
Frames Delivered	293857	276887	302949	326672	356609
Packets Delivered	31994	30146	32984	35567	38828
Packets/second	639.880	602.920	659.680	711.340	776.560
Bytes Delivered	2046995000	1928769000	2110309000	2275575000	2484141000
KBPS Delivered	327519.2000	308603.040	337649.440	364092.000	397462.560
Utilization (%)	73.46	69.22	75.74	81.67	89.15
From CUNYPrincet					
Frames Delivered	245990	276891	279124	335768	327636
Packets Delivered	26782	30146	30390	36557	35763
Packets/second	535.640	602.920	607.800	731.140	713.460
Bytes Delivered	1713565000	1928792000	1944362000	2338981000	2282267000
KBPS Delivered	274170.400	308606.720	311097.920	374236.960	365162.720
Utilization (%)	61.50	69.22	69.78	83.94	81.91
From MontpelierPri					
Frames Delivered	232188	294652	288305	309522	336313
Packets Delivered	25279	32079	31389	33698	36615
Packets/second	505.580	641.580	627.780	673.960	732.300
Bytes Delivered	1617396000	2052527000	2008321000	2156074000	2342716000
KBPS Delivered	258783.36	328404.320	321331.360	344971.840	374834.560
Utilization (%)	58.05	73.66	72.08	77.38	84.08
From Milano-Mont					
Frames Delivered	236809	348841	352141	361399	345436
Packets Delivered	25781	37980	38338	39347	37608
Packets/second	515.620	759.600	766.760	786.940	752.160
Bytes Delivered	1649524000	2430007000	2452988000	2517472000	2406245000
KBPS Delivered	263923.840	388801.120	392478.080	402795.520	384999.200
Utilization (%)	59.20	87.21	88.04	90.35	86.36

Table 9. 57: HDTV - Links: Channel Utilization/Utilization by Application for Compressed Data

- **Uncompressed Data**

In case of uncompressed data, the number of frames delivered from CUNY to Milano is 302949, 356609, 340338, 355224, and 336908 and the number of packets delivered is 32984, 38828, 37054, 38676, and 36682 at 50%, 60%, 70%, 80%, and 90% bandwidths, respectively. We see an increase of 18% at 60% and a decrease of 4% at 70% in the number of frames/packets delivered. At 80% the number of frames/packets delivered increases by approximately 4%. And at 90% the number of frames/packets delivered decreases by 5%. Similarly, the link utilization percentage is upward from 50% to 60% and from 70% to 80% and downward at 70% and 90%. In case of from CUNY to Princeton link, the number of frames/packets delivered goes up by from 50% to 60% bandwidths, decreases at 80% and again increases at 90%. From Montpelier to Princeton link, the number of frames/packets delivered increases by 16% and 9% at 60% and 80% bandwidths, respectively. The number of frames/packets delivered decreases by 1% and 2% at 70% and 90% bandwidths, respectively. From Milano to Montpelier link the decrease is 2% and 5% at 60% and 70% bandwidths, respectively. At 80% the number of frames/packets delivered increases by 12% and at 90% decreases by nearly 5%. These changes also reflect in other categories, packets/second, bytes delivered, kbps delivered, and utilization percentage as shown in **Table 9.58**.

HDTV (OC-9)	Uncompressed Data				
MPEG-2 (30%)	Bandwidth Used For Simulation				
Link	50%	60%	70%	80%	90%
From CUNY-Milano					
Frames Delivered	302949	356609	340338	355224	336908
Packets Delivered	32984	38828	37054	38676	36682
Packets/second	659.680	776.560	741.080	773.520	733.640
Bytes Delivered	2110309000	2484141000	2370743000	2474459000	2346866000
KBPS Delivered	337649.440	397462.560	379318.880	395913.440	375498.560
Utilization (%)	75.74	89.15	85.08	88.81	84.23
From CUNYPrincet					
Frames Delivered	279124	327636	339368	330929	331214
Packets Delivered	30390	35763	36949	36026	36059
Packets/second	607.800	713.460	738.980	720.520	721.180
Bytes Delivered	1944362000	2282267000	2364000000	2305227000	2307201000
KBPS Delivered	311097.920	365162.720	378240.000	368836.320	369152.160
Utilization (%)	69.78	81.91	84.84	82.73	82.80
From MontpelierPri					
Frames Delivered	288305	336313	333024	362241	355226
Packets Delivered	31389	36615	36256	39437	38675
Packets/second	627.780	732.300	725.120	788.740	773.500
Bytes Delivered	2008321000	2342716000	2319763000	2523324000	2474487000
KBPS Delivered	321331.360	374834.560	371162.080	403731.840	395917.920
Utilization (%)	72.08	84.08	83.26	90.56	88.81
From Milano-Mont					
Frames Delivered	352141	345436	327702	366304	347334
Packets Delivered	38338	37608	35678	39880	37814
Packets/second	766.760	752.160	713.560	797.600	756.280
Bytes Delivered	2452988000	2406245000	2282748000	2551630000	2419521000
KBPS Delivered	392478.080	384999.200	365239.680	408260.800	387123.360
Utilization (%)	88.04	86.36	81.93	91.58	86.83

Table 9. 58: HDTV - Links: Channel Utilization/Utilization by Application for Uncompressed Data

First row in **Tables 9.57** and **9.58** represents packets delivered during the simulation. They fluctuate for some links and increase as the bandwidth increases for others. This fluctuation could be a function of network delay and/or link/node failure.

Fluctuations in:

- Packets/Second (Packets Delivered/Replication Length, 50s);
- Bytes Delivered (Packets Delivered * Message Size);
- KBPS (kilobits/second, Packets/Second Message Size * 8bits); and
- Link Utilization

can be seen as reflections of the fluctuation in packets delivered.

Comparison of the results shown in **Table 9.59** demonstrates that the utilization rate for compressed data is consistently lower than for uncompressed data in most cases. This implies that compression permits transmission of larger volume of data in shorter intervals.

HDTV	Link Utilization (%)	
Link	Compressed Data	Uncompressed Data
From CUNY-Milano		
50%	73.46	75.74
60%	69.22	89.15
70%	75.74	85.08
80%	81.67	88.81
90%	89.15	84.23
From CUNYPrinceton		
50%	61.50	69.78
60%	69.22	81.91
70%	69.78	84.94
80%	83.94	82.73
90%	81.91	82.80
From MontpelierPri		
50%	58.05	72.08
60%	73.66	84.08
70%	72.08	83.26
80%	77.38	90.56
90%	84.08	88.81
From Milano-Mont		
50%	59.20	88.04
60%	87.21	86.36
70%	88.04	81.93
80%	90.35	91.58
90%	86.36	86.83

Table 9. 59: HDTV - Comparison of Utilization (%) for Compressed & Uncompressed Data

WAN Clouds: Access Link Statistics Analysis

Tables 9.60, 9.61, 9.62 and 9.63 present statistics for compressed data and uncompressed data for 7 access links: (1) CUNY-Montpelier, (2) CUNY-Milano, (3) CUNY-Bern, (4) CUNY-Bonn, (5) Princeton-Montpelier, (6) CUNY-Princeton, and (7) Montpelier-Milano. Only three of these links are chosen for analysis to demonstrate the pattern of variation over access links.

Compressed Data	Frames Accepted				
Access Link	50%	60%	70%	80%	90%
CUNY-Montpelier	24570	31163	31598	29888	34135
CUNY-Milano	28685	27066	29747	31980	34979
CUNY-Bern	30416	28713	33568	30471	35790
CUNY-Bonn	26610	30027	29677	31034	37337
Princeton-Montpelier	22704	28841	28247	30335	33022
CUNY-Princeton	24131	27176	27318	32806	32091
Montpelier-Milano	23168	34245	34465	35322	33807

Table 9. 60: HDTV - WAN Cloud Access Link Statistics: Frames Accepted (Compressed Data)

Compressed Data	Frames Dropped				
Access Link	50%	60%	70%	80%	90%
CUNY-Montpelier	2816	3384	3524	3274	3800
CUNY-Milano	3311	3080	3237	3587	3851
CUNY-Bern	3356	3175	3713	3359	3969
CUNY-Bonn	2894	3350	3245	3438	4243
Princeton-Montpelier	2575	3238	3142	3361	3593
CUNY-Princeton	2704	3037	3077	3675	3585
Montpelier-Milano	2612	3735	3873	4025	3799

Table 9. 61: HDTV - WAN Cloud Access Link Statistics: Frames Dropped (Compressed Data)

Uncompressed Data	Frames Accepted				
Access Link	50%	60%	70%	80%	90%
CUNY-Montpelier	31598	34135	33930	33490	33818
CUNY-Milano	29747	34979	33341	34815	33032
CUNY-Bern	33568	35790	34286	34395	37191
CUNY-Bonn	29677	37337	33428	33291	31076
Princeton-Montpelier	28247	33022	32717	35521	34901
CUNY-Princeton	27318	32091	33345	32341	32457
Montpelier-Milano	34465	33807	32146	35900	34061

Table 9. 62: HDTV - WAN Cloud Access Link Statistics: Frames Accepted (Uncompressed Data)

Uncompressed Data	Frames Dropped				
Access Link	50%	60%	70%	80%	90%
CUNY-Montpelier	3524	3800	3958	3818	3858
CUNY-Milano	3237	3851	3713	3861	3650
CUNY-Bern	3713	3969	3921	3749	4104
CUNY-Bonn	3245	4243	3687	3705	3579
Princeton-Montpelier	3142	3593	3540	3916	3774
CUNY-Princeton	3077	3585	3565	3656	3531
Montpelier-Milano	3873	3799	3532	3980	3753

Table 9. 63: HDTV - WAN Cloud Access Link Statistics: Frames Dropped (Uncompressed Data)

- **Compressed Data**

The number of frames accepted by CUNY-Montpelier access link are 24570, 31163, 31598, 29888, and 34135 at 50%, 60%, 70%, 80%, and 90% bandwidths, respectively. The number of dropped frames similarly is 2816, 3384, 3524, 3274, and 3800. The number of frames accepted by the link shows that the increase from 50% to 60% and 80% to 90% bandwidths is upward and it is downward at 70% and 80% bandwidths. Similarly, the number of frames dropped by the link shows that the increase is linear from 50% to 70%, goes down at 80%, and again increases at 90% bandwidth. The drop rate is 11% for all five different bandwidth simulation runs.

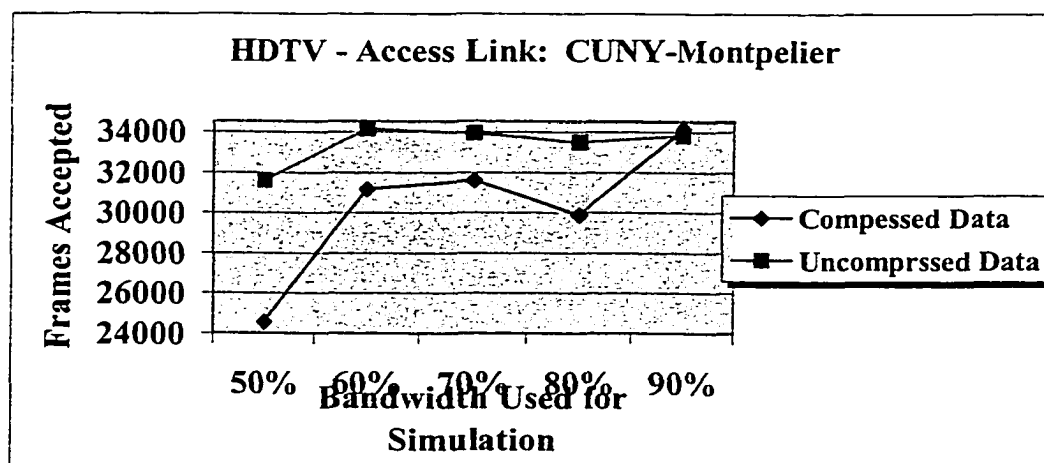


Figure 9. 25: HDTV – Selected Access Link: CUNY-Montpelier (Frames Accepted)

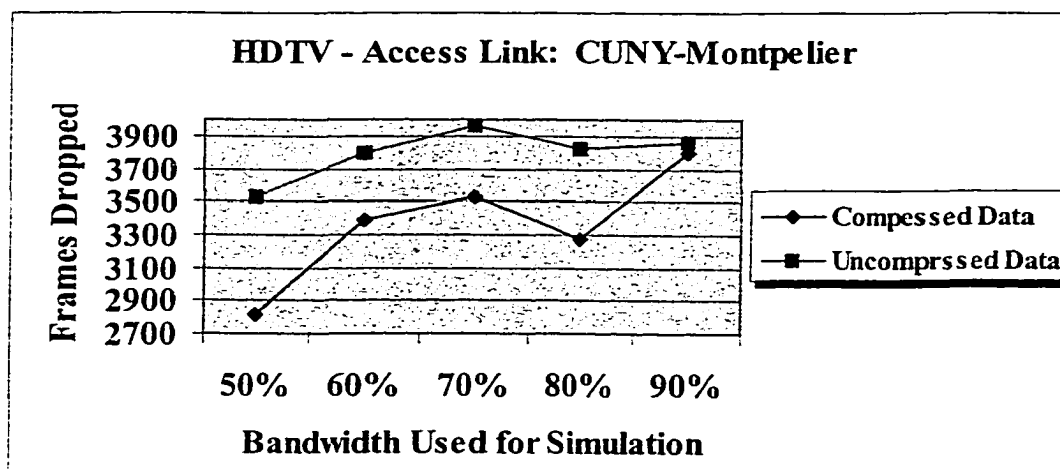


Figure 9. 26: HDTV – Selected Access Link: CUNY-Montpelier (Frames Dropped)

CUNY-Bonn access link accepted 26610 frames at 50% and 30027 frames at 60% bandwidths. At 70% the number of frames accepted by the link went down to 29677, a decrease of 1%. The number of frames accepted at 80% and 90% bandwidth is 31034 and 37337, respectively. We see an increase of nearly 5% and 12% at 80% and 90% bandwidths, respectively. The number of dropped frames at each bandwidth percentage is 2894 at 50%, 3350 at 60%, 3245 at 70%, 3438 at 80% and 4243 at 90%. The drop rate is 11% or all five different bandwidth simulation runs.

The numbers of frames accepted by Montpelier-Milano access link for compressed data are 23168, 34245, 34465, 35322, and 33807 at 50%, 60%, 70%, 80%, and 90% bandwidths, respectively. The number of frames accepted increases linearly from 50% to 80% bandwidths and decreases at 90% by nearly 4%. The number of frames dropped similarly is 2612, 3735, 3873, 4025, and 3799. The drop rate is 11% for all five bandwidths.

- **Uncompressed Data**

The number of frames accepted by CUNY-Montpelier link is 31598, 34135, 33930, 33490, and 33818 at 50%, 60%, 70%, 80%, and 90% bandwidths, respectively, as presented in **Table 9.62**. From 50% to 60% bandwidths, the numbers indicate that the increase is linear. However, from 70% to 80% the number of frames accepted by the link goes down, a decrease of 0.6% at 70% and 1% at 80%. We observe an increase of nearly 1% at 90% bandwidth. The number of frames dropped is 3524 at 50%, 3800 at 60%, 3958 at 70%, 3818 at 80% and 3358 at 90%, as presented in **Table 9.64**. The number of frames dropped decreases by 3% and 12% at 80% and 90% bandwidths, respectively. The drop rate is 11%, 11%, 12%, 11%, and 11% for all five different bandwidth simulation runs, respectively.

The number of frames accepted by CUNY-Bonn access link are 29677, 37337, 33428, 33291, and 31076 at 50%, 60%, 70%, 80%, and 90% bandwidth simulation runs, respectively. These numbers indicate that at 70% the number of frames accepted by the link decreases by 10%. From 70% to 90% the increase is downward. The number of frames dropped is 3245 at 50%, 4243 at 60%, 3687 at 70%, 3705 at 80% and 3579 at 90%. At 70% the number of frames dropped by the link decreases by 13%. At 80% the number is 3705, an increase of 0.5% and at 90% it is 3579, a decrease of 3%. The drop rate is 11% for all five different bandwidth simulation runs.

The number of frames accepted by Montpelier-Milano access link for uncompressed data is 34465, 33807, 32146, 35900, and 34061 at 50%, 60%, 70%, 80%, and 90% bandwidths, respectively. We observe that at 60% and 70% the number of frames accepted by the link decreases by approximately 2% and 5%, respectively. At 80% there is an increase of 12% and a decrease of 5% at 90% bandwidth. The number of frames dropped is 3873 at 50%, 3799 at 60%, 3532 at 70%, 3980 at 80% and 3753 at 90%. The decrease is 2% and 7% at 60% and 70% bandwidths, respectively. At 80% the number of frames dropped is 3980, an increase of nearly 13%. However, at 90% there is a decrease of 6%. The drop rate is 11% for all five different bandwidth simulation runs.

Comparison of Frames-Dropped

The drop rate of compressed and uncompressed data and their differences are shown in **Table 9.64**. Also shown in the table is the comparison of the link utilization rate for both data types.

The drop rate of CUNY-Montpelier access link for 50%, 60%, 80%, and 90% bandwidths of compressed data is equal to that of uncompressed data, 11%. At 70% bandwidth the drop rate is 11% and 12% for compressed and uncompressed data, respectively. So the difference is 1 which is within the margin of error, $\pm 5\%$. The link utilization rate for compressed data at 50%-80% bandwidths is less than for uncompressed data. However, the link utilization rate for compressed data at 90% is more than for uncompressed data.

The drop rate of CUNY-Bonn access link at all bandwidths for compressed and uncompressed data is 11%. The link utilization rate of compressed data is 64.75%, 73.25%, 72.25%, 75.65% and 91.25% at 50%-90% bandwidths, respectively. The link utilization rate of uncompressed data similarly is 72.25%, 91.25%, 81.46% and 81.19%, and 76.06%. We note that the link utilization of uncompressed data at 50% and 60% bandwidths is equal to that of compressed data at 70% and 90% bandwidths, respectively. Clearly, the link

utilization rate for compressed data is less than for uncompressed data except at 90% bandwidth.

The drop rate of Montpelier-Milano access link of compressed and uncompressed data is 11%. The link utilization rate for compressed data at 50% bandwidth is 56.58% compared to 84.14% for uncompressed data. At 60% bandwidth the utilization of compressed data 83.35% and for uncompressed data it is 82.53%. We observe that the link utilization rate for compressed data at 70% is more than for uncompressed data. The link utilization rate for compressed data at 80% and 90% bandwidths is less than for uncompressed data as shown in **Table 9.64**.

This demonstrates that while the drop rate is more or less the same for both compressed and uncompressed data, compressing data has a clear and substantive advantage in terms of the link utilization rate. Thus, data compression allows for transmission of substantially larger volume of data in shorter intervals.

HDTV (OC-9)	WAN CLOUDS: ACCESS LINK STATS (ENTRY)									
MPEG-2 (30%)	Comparison of Compressed & Uncompressed HDTV Data									
Msg Size: 93.225MB	Frames Accepted/Dropped/Utilization									
Cloud: Access Link	Cmp 50%	Uncmp 50%	Cmp 60%	Uncmp 60%	Cmp 70%	Uncmp 70%	Cmp 80%	Uncmp 80%	Cmp 90%	Uncmp 90%
CUNY-Montpelier										
Frames Accepted	24570	31598	31163	34135	31598	33930	29888	33490	34135	33818
Frames Dropped	2816	3524	3384	3800	3524	3958	3274	3818	3800	3858
Drop Rate	0.1146	0.112	0.1085	0.111	0.112	0.117	0.110	0.114	0.111	0.114
Drop Rate (%)	11%	11%	11%	11%	11%	12%	11%	11%	11%	11%
Difference	0%		-0%		-1%		-0%		-0%	
Link Utilization (%)	60.10	77.08	75.81	83.26	77.08	83.15	72.78	81.88	83.26	82.68
CUNY-Milano										
Frames Accepted	28685	29747	27066	34979	29747	33341	31980	34815	34979	33032
Frames Dropped	3311	3237	3080	3851	3237	3713	3587	3861	3851	3650
Drop Rate	0.1154	0.109	0.1137	0.110	0.109	0.111	0.1121	0.111	0.110	0.110
Drop Rate (%)	12%	11%	11%	11%	11%	11%	11%	11%	11%	11%
Difference	1%		0%		-0%		0%		-0%	
Link Utilization (%)	70.22	72.38	66.16	85.21	72.38	81.32	78.05	84.87	85.21	80.50

HDTV (OC-9)	Frames Accepted/Dropped/Utilization (Cont'd.)									
Cloud: Access Link	Cmp 50%	Uncmp 50%	Cmp 60%	Uncmp 60%	Cmp 70%	Uncmp 70%	Cmp 80%	Uncmp 80%	Cmp 90%	Uncmp 90%
CUNY-Bern										
Frames Accepted	30416	33568	28713	35790	33568	34286	30471	34395	35790	37191
Frames Dropped	3356	3713	3175	3969	3713	3921	3359	3749	3969	4104
Drop Rate	0.1103	0.111	0.1105	0.111	0.111	0.114	0.110	0.109	0.111	0.110
Drop Rate (%)	11%	11%	11%	11%	11%	11%	11%	11%	11%	11%
Difference	-0%		-0%		-0%		0%		0%	
Link Utilization (%)	74.11	81.82	69.98	87.25	81.82	83.85	74.24	83.71	87.25	90.62
CUNY-Bonn										
Frames Accepted	26610	29677	30027	37337	29677	33428	31034	37337	37337	31076
Frames Dropped	2894	3245	3350	4243	3245	3687	3438	4243	4243	3579
Drop Rate	0.1087	0.109	0.1115	0.114	0.109	0.110	0.1107	0.114	0.114	0.115
Drop Rate (%)	11%	11%	11%	11%	11%	11%	11%	11%	11%	12%
Difference	-0%		-0%		-0%		-0%		-1%	
Link Utilization (%)	64.75	72.25	73.25	91.25	72.25	81.46	75.65	81.19	91.25	76.06
Princeton-Montpelier										
Frames Accepted	22704	28247	28841	33022	28247	32717	30335	35521	33022	34901
Frames Dropped	2575	3142	3238	3593	3142	3540	3361	3916	3593	3774
Drop Rate	0.1134	0.111	0.1122	0.109	0.111	0.108	0.1107	0.110	0.109	0.108
Drop Rate (%)	11%	11%	11%	11%	11%	11%	11%	11%	11%	11%
Difference	0%		0%		0%		0%		0%	
Link Utilization (%)	55.48	68.88	70.4	80.36	68.88	79.57	73.95	86.55	80.36	84.47

HDTV (OC-9)	Frames Accepted/Dropped/Utilization (Cont'd.)									
Cloud: Access Link	Cmp 50%	Uncmp 50%	Cmp 60%	Uncmp 60%	Cmp 70%	Uncmp 70%	Cmp 80%	Uncmp 80%	Cmp 90%	Uncmp 90%
CUNY-Princeton										
Frames Accepted	24131	27318	27176	32091	27318	33345	32806	32341	32091	32457
Frames Dropped	2704	3077	3037	3585	3077	3565	3675	3656	3585	3531
Drop Rate	0.1120	0.113	0.1117	0.112	0.113	0.107	0.1120	0.113	0.112	0.10879
Drop Rate (%)	11%	11%	11%	11%	11%	11%	11%	11%	11%	11%
Difference	-0%		0%		1%		-0%		0%	
Link Utilization (%)	58.89	66.70	66.30	78.29	66.70	81.00	80.06	79.01	78.29	78.98
Montpelier-Milano										
Frames Accepted	23168	34465	34245	33807	34465	32146	35322	35900	33807	34061
Frames Dropped	2612	3873	3735	3799	3873	3532	4025	3980	3799	3753
Drop Rate	0.1127	0.112	0.1090	0.112	0.1123	0.110	0.1139	0.111	0.112	0.110
Drop Rate (%)	11%	11%	11%	11%	11%	11%	11%	11%	11%	11%
Difference	0%		-0%		0%		0%		-0%	
Link Utilization (%)	56.58	84.14	83.35	82.53	84.14	78.30	86.35	87.52	82.53	82.99

Table 9. 64: HDTV - WAN Cloud: Access Link Statistics- Frames Accepted, Frames Dropped & Link Utilization

Message & Response Sources: Packets Delivered/Delay

Table 9.65 shows data for 6 distinct origin nodes that send messages to 6 different destination nodes. Of these, only two nodes are chosen for analysis that follows:.

CUNY-Milano/CUNY-Princeton (Origin/Destination)

Compressed Data

CUNY-Milano, origin node, created 39339, 40796, 55366, 65565, and 61194 to send to CUNY-Princeton, destination node, at 50%, 60%, 70%, 80%, and 90% bandwidth, respectively. The number of packets created increases linearly from 50% to 80% bandwidths and decreases by 7% at 90% bandwidth. The number of packets delivered at the destination is 7211, 5140, 5928, 6022, and 6550 at the five simulation runs, respectively. And the number of packets dropped similarly is 12218, 13516, 25528, 34157, and 27743. The number of packets delivered and packets dropped show a pattern of fluctuation, as may be observed from **Table 9.65**. The average packet delay (in milliseconds) also fluctuates over the range of the bandwidths.

Uncompressed Data

CUNY-Milano, origin node, created 55366, 61194, 87420, 101990, and 84506 to send to CUNY-Princeton, destination node, at 50%, 60%, 70%, 80%, and 90% bandwidth, respectively. The number of packets created increases from 50% to 80% bandwidths linearly and decreases at 90% by 17%. The number of packets delivered at the destination similarly is 5928, 6550, 5385, 5066, and 5882. These numbers show a decrease from 6550 packets delivered at 60% to 5385 packets delivered at 70%, a decrease of nearly 18%. The number further decreases at 80% bandwidth by 6%. We observe a 16% increase at 90% bandwidth. And the number of packets dropped is 25528, 27743, 54571, 67753, and 52980 at the five simulation runs, respectively. The average packet delay (in milliseconds) increases from 50% bandwidth to 80% bandwidth and decreases at 90%.

Comparison of Compressed and Uncompressed Data

The number of packets created for compressed data is less than for uncompressed data for all simulation runs, as shown in **Table 9.65**. The number of packets delivered for compressed data at 60% bandwidth is less than for uncompressed data but higher at the other bandwidths. The number of packets dropped for compressed data is less than for uncompressed data for all five

bandwidths. The ratio of packets dropped over packets delivered is 31%, 33%, 46%, 52%, and 45% for compressed data and for uncompressed data, it is 46%, 45%, 62%, 66%, and 63%. We see high percentages at 70%-90% bandwidths for uncompressed data. This can be attributed to network saturation. The average packet delay (in milliseconds) for compressed data is more than for uncompressed data at 50% bandwidth and less for the rest of the bandwidths.

HDTV (OC-9)	Message & Response Sources: Packets Delivered/Packet Delay									
MPEG-2 (30%)	Compressed Data						Uncompressed Data			
Msg Size: 93.225MB	Bandwidth Used For Simulation									
Origin/Destination	50%	60%	70%	80%	90%	50%	60%	70%	80%	90%
CUNY-Milano/ CUNY-Princeton										
Packets Created	39339	40796	55366	65565	61194	55366	61194	87420	101990	84506
Packets Delivered	7211	5140	5928	6022	6550	5928	6550	5385	5066	5882
Packets Dropped	12218	13516	25528	34157	27743	25528	27743	54571	67753	52980
Dropped/Created	31%	33%	46%	52%	45%	46%	45%	62%	66%	63%
Dropped/Delivered	169%	263%	431%	567%	424%	431%	424%	1013%	1337%	901%
Packet Delay (MS)										
Average	27729.297	23811.083	26411.686	30235.394	28085.098	26411.686	28085.098	29301.589	32917.091	30436.173
Maximum	44751.389	40218.244	48754.201	47756.175	48750.264	48754.201	48750.264	48754.340	49756.388	49750.564
MontpelierP/ Montpelier-Milano										
Packets Created	30597	53909	55366	62651	75764	55366	75764	72850	110732	107818
Packets Delivered	4756	6143	6054	4398	4969	6054	4969	5665	5387	6295
Packets Dropped	7402	26729	27812	31726	44060	27812	44060	41612	76561	74486
Dropped/Created	24%	50%	50%	51%	58%	50%	58%	57%	69%	69%
Dropped/Delivered	156%	435%	459%	721%	887%	459%	887%	735%	1421%	1183%
Packet Delay (MS)										
Average	24978.969	26740.676	28679.710	21768.347	28709.967	28679.710	28709.967	29351.221	29992.719	29145.087
Maximum	42285.289	47528.289	48936.250	35953.092	48527.852	48936.250	48527.852	48526.498	49530.185	49529.396

HDTV (OC-9)	Message & Response Sources: Packets Delivered/Packet Delay (Cont'd.)									
MPEG-2 (30%)	Compressed Data						Uncompressed Data			
Msg Size: 93.225MB	Bandwidth Used For Simulation									
Origin/Destination	50%	60%	70%	80%	90%	50%	60%	70%	80%	90%
CUNY-Bern/ Bern-Bonn										
Packets Created	50995	40796	69936	61194	93248	69936	93248	78678	101990	129673
Packets Delivered	6578	6001	6725	6014	5265	6725	5265	4599	6694	4535
Packets Dropped	20425	12639	36217	31316	58035	36217	58035	44785	69250	92207
Dropped/Created	40%	31%	52%	51%	62%	52%	62%	57%	68%	71%
Dropped/Delivered	311%	211%	539%	521%	1102%	539%	1102%	974%	1035%	2033%
Packet Delay (MS)										
Average	28542.876	28723.187	27372.258	26732.030	30395.374	27372.258	30395.374	28124.773	28923.630	28573.510
Maximum	44523.256	47721.743	48050.444	47497.133	48498.569	48050.444	48498.569	47500.507	49018.751	47998.118
CUNY-Bonn/ Milano-Bonn										
Packets Created	43710	65565	55366	65565	83049	55366	83049	77221	69936	83049
Packets Delivered	5931	5284	5764	4786	7019	5764	7019	6732	6790	5805
Packets Dropped	16886	36116	27043	35028	46499	27043	46499	45396	38307	52608
Dropped/Created	39%	55%	49%	53%	56%	49%	56%	59%	55%	63%
Dropped/Delivered	285%	683%	469%	732%	662%	469%	662%	674%	564%	906%
Packet Delay (MS)										
Average	23135.941	32307.930	29194.081	30767.924	28435.771	29194.081	28435.771	28781.375	29663.008	30351.040
Maximum	41561.363	48331.866	48778.794	47777.354	48782.083	48778.794	48782.083	48780.728	49534.977	49538.579

HDTV (OC-9)	Message & Response Sources: Packets Delivered/Packet Delay (Cont'd.)									
MPEG-2 (30%)	Compressed Data						Uncompressed Data			
Msg Size: 93.225MB	Bandwidth Used For Simulation									
Origin/Destination	50%	60%	70%	80%	90%	50%	60%	70%	80%	90%
CUNY-Montpelier/ Montpelier-Milano										
Packets Created	37882	53909	59737	74307	85963	59737	85963	72850	74307	99076
Packets Delivered	5238	6418	5700	7087	7188	5700	7188	6729	4693	6018
Packets Dropped	14159	24180	29818	44308	52624	29818	52624	40283	40721	65929
Dropped/Created	37%	45%	50%	60%	61%	50%	61%	55%	55%	67%
Dropped/Delivered	270%	377%	523%	625%	732%	523%	732%	599%	868%	1096%
Packet Delay (MS)										
Average	26371.283	29759.445	29964.840	28427.087	29525.057	29964.840	29525.057	26348.427	22705.085	25569.784
Maximum	44342.415	47943.619	49048.554	48184.007	48994.674	49048.554	48994.674	45145.309	36840.205	47995.121
CUNYPrinceton/ Princeton-Mont										
Packets Created	32054	42253	50995	53909	74307	50995	74307	84506	90334	93248
Packets Delivered	6466	6342	3997	5389	5828	3997	5828	6230	5803	5570
Packets Dropped	9738	15870	25482	21645	42040	25482	42040	51934	58585	62221
Dropped/Created	30%	38%	50%	40%	57%	50%	57%	61%	65%	67%
Dropped/Delivered	151%	250%	638%	402%	721%	638%	721%	834%	1010%	1117%
Packet Delay (MS)										
Average	24239.332	29159.845	30701.311	31655.681	27422.337	30701.311	27422.337	28005.731	28762.795	29290.829
Maximum	44494.159	48813.128	49141.975	49109.803	48140.874	49141.975	48140.874	48347.756	49238.785	49175.039

Table 9. 65: HDTV - Message & Response Sources: Packets Delivered/Dropped/Delay

MontpelierP/Montpelier-Milano (Origin/Destination)

Compressed Data

MontpelierP, origin node, created packets 30597, 53909, 55366, 62651, and 75764 to send to Montpelier-Milano, destination node, at 50%, 60%, 70%, 80%, and 90% bandwidths, respectively. The number of packets delivered at the destination is 4756, 6143, 6054, 4398, and 4969 at the five simulation runs, respectively. And the number of packets dropped similarly is 7402, 26729, 27812, 31726, and 44060. The number of packets created increases linearly from 50% to 90% bandwidths. The number of packets delivered increases at 60% by 29%. However, the number of packets delivered decreases linearly from 60% to 80% bandwidths and again increases at 90% bandwidth. The number of packets dropped by the network increases linearly from lower bandwidth to higher bandwidth. The average packet delay (in milliseconds) increases linearly from 50% to 70%, decreases at 80%, and again increases at 90% bandwidths

Uncompressed Data

MontpelierP, origin node, created 55366, 75764, 72850, 110732, and 107818 to send to Montpelier-Milano, destination node, at 50%, 60%, 70%, 80%, and 90% bandwidths, respectively. The number of packets delivered at the destination is 6054, 4969, 5665, 5387, and 6295 at the five simulation runs, respectively. And the number of packets dropped similarly is 27812, 44060,

41612, 76561, and 74486. The number of packets created by the origin node decreases at 70% and 90% bandwidths by nearly 4% and 3%, respectively. The ratio of packets dropped over packets delivered is 50%, 58%, 57%, 69%, and 69% at 50%-90% bandwidths. The average packet delay (in milliseconds) increases linearly from 50% to 80% and decreases at 90% bandwidth.

Comparison of Compressed and Uncompressed Data

The number of packets created for compressed data is less than that of uncompressed data for all of the simulation runs. However, for compressed data the number packets delivered is more for 60%-70% bandwidths (6143/4969 and 6054/5665) than that of uncompressed data. The number of packets dropped for compressed data is less than for uncompressed data for all of the simulation runs. The ratio of packets dropped over packets delivered is 24%, 50%, 50%, 51%, and 58% for compressed data and for uncompressed data it is 50%, 58%, 57%, 69%, and 69%. We observe that the drop ratio for uncompressed data at 50% bandwidth is twice the drop ratio of compressed data and for the rest of the bandwidths the difference is small.

9.6. Discussion of Results

Five different applications have been used for the purpose of simulation study. They include text and multimedia applications. Text application uses small message size while multimedia applications demand large size. So the message size varies from 100 bytes for text to 93.225 MB for HDTV application. Bandwidth is determined according to the type of application. Smaller bandwidth for text and larger bandwidth for multimedia applications has been used. The number of packets/second is calculated using bandwidth and message size as well as the interarrival time (IAT). As the bandwidth and message size increase so does the number of packets/second. We also observe that the IAT decreases as the number of packets/second increases since they are inversely related. The simulation run time is observed for each run. It ranges from one or two minutes for text file to several hours for multimedia applications. Different compression algorithms are used for different applications. Comparisons between compressed and uncompressed data have been made.

The network data obtained in the simulation runs show that the compressed data has an impact on link and channel utilization and WAN cloud access link utilization. The results also show that the error/drop rate (frames

dropped/frames accepted) of compressed data is close to that of uncompressed data. The following discussion of results is categorized by **Links**, **WAN cloud** and **Message Sources**.

- **Links**

The data in each application for each link show that the number of frames/packets delivered of compressed data by a link is less than that of uncompressed data for each bandwidth percentage. The number of packets/second is also smaller than that of uncompressed data. Since the number of packets/second is small the time to transmit this data is less. The channel utilization of compressed data is less than that of uncompressed data. The time it takes to transmit uncompressed data is longer implying that the channel utilization rate is higher. That means that the number of times we can use the channel to transmit uncompressed data is less because of the amount of data involved in transmitting. On the other hand, the time it takes to transmit compressed data is shorter due to the compressed volume of data. Hence, the channel usage is more efficient for compressed data than for uncompressed data. We also observe that the channel utilization rate is close to 100% for uncompressed data implying that the channel is occupied for a longer period of time during the transmission of data. The time it takes to transmit compressed data is less so that the channel can be utilized more efficiently. We also note that the number of frames/packets delivered and

packets/second oscillates or fluctuates for some links and increases linearly with bandwidth for other links for both data types. The same fluctuation is noted on channel utilization rate. We ascribe such variation to the duration of the simulation run and also whether steady saturation conditions have been reached as the results are scanned.

The table below shows the ranges of packets delivered and utilization rate for compressed and uncompressed data by application. In case of each application, the ranges for compressed data are lower than that of uncompressed data. This demonstrates the improvement in the transmission of compressed data over uncompressed data as well as link usage. **Figures 9.15-9.28** depict the utilization rate by link over the range of the bandwidths for each of the five applications. The link utilization is chosen for graphing because of its importance.

Application	Compression Algorithm	Ranges	
		Compressed Data	Uncompressed Data
E-mail	Huffman		
Packets Delivered		1200 - 2500	1500 - 3700
Utilization Rate (%)		29.50 – 61.60	37.88 - 91.40
E-mail	LZW		
Packets Delivered		653 - 1612	1515 – 3655
Utilization Rate (%)		16.43 – 37.50	37.88 – 91.40
Audio	MPEG-1	Simulation Run1	
Packets Delivered		900 - 2200	1450 - 3600
Utilization Rate (%)		21.98 – 54.88	38.83 - 88.75
Audio	MPEG-1	Simulation Run2	
Packets Delivered		436 - 993	900 - 2200
Utilization Rate (%)		10.90 – 24.83	21.98 – 54.88
Image	MPEG-1		
Packets Delivered		130000 - 300000	170000 - 335000
Utilization Rate (%)		31.84 – 73.36	43.04 – 83.60
Video	MPEG-2		
Packets Delivered		137000 - 270000	175000 - 350000
Utilization Rate (%)		34.42 – 67.04	44.07 - 86.62
HDTV	MPEG-2		
Packets Delivered		230000 - 362000	275000 - 370000
Utilization Rate (%)		58.05 – 90.35	69.78 - 91.58

Ranges of Packets Delivered & Utilization Rate

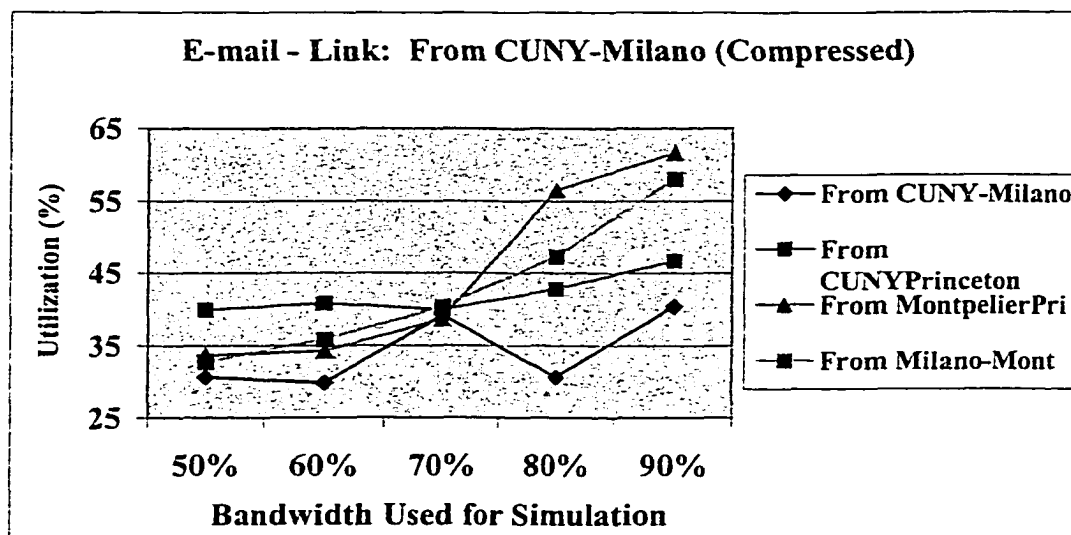


Figure 9. 27: E-mail (Huffman) – Link Utilization (%) (Compressed Data)

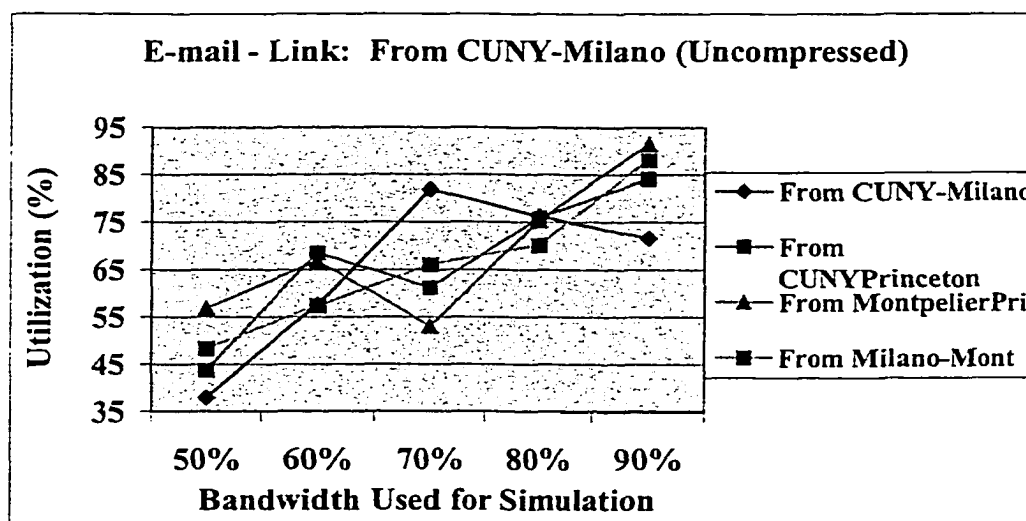


Figure 9. 28: E-mail (Huffman) – Link Utilization (%) (Uncompressed Data)

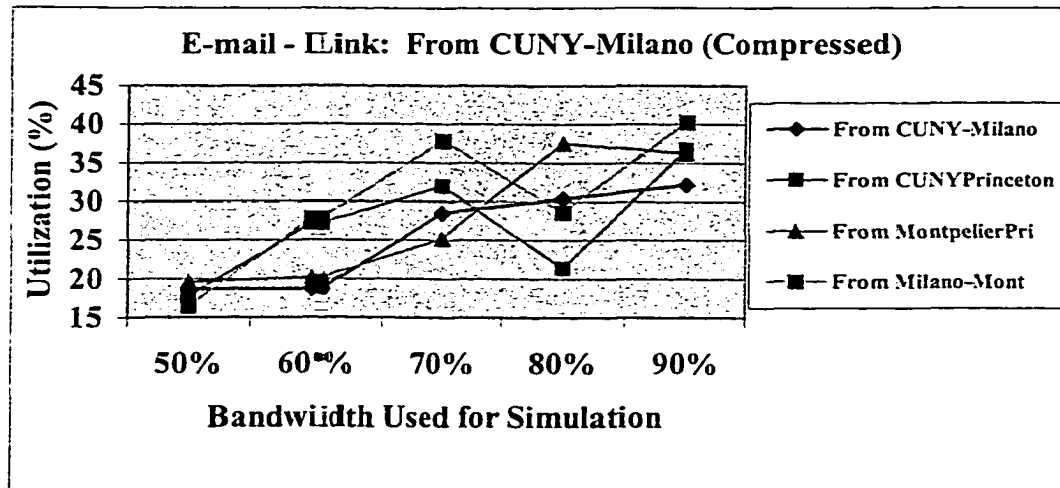


Figure 9. 29: E-mail (LZW) – Link Utilization (%) (Compressed Data)

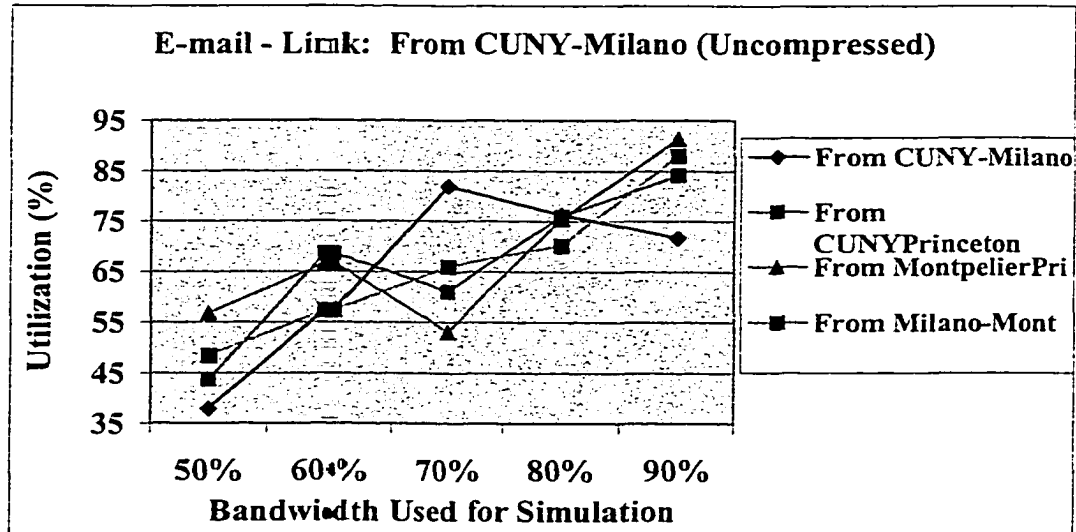


Figure 9. 30: E-mail (LZW) – Link Utilization (%) (Uncompressed Data)

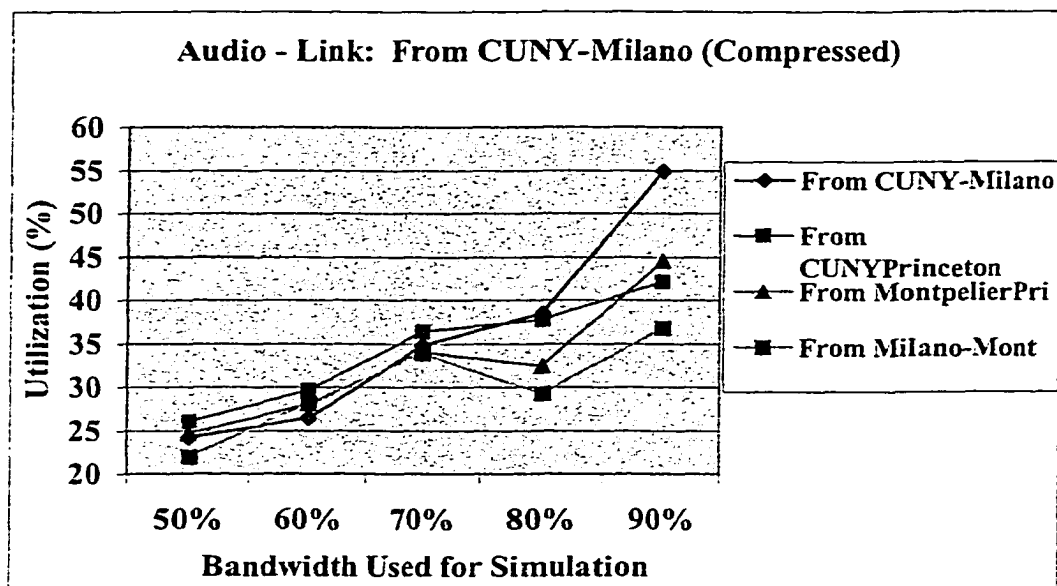


Figure 9. 31: Audio (SimRun1) – Link Utilization (%) (Compressed Data)

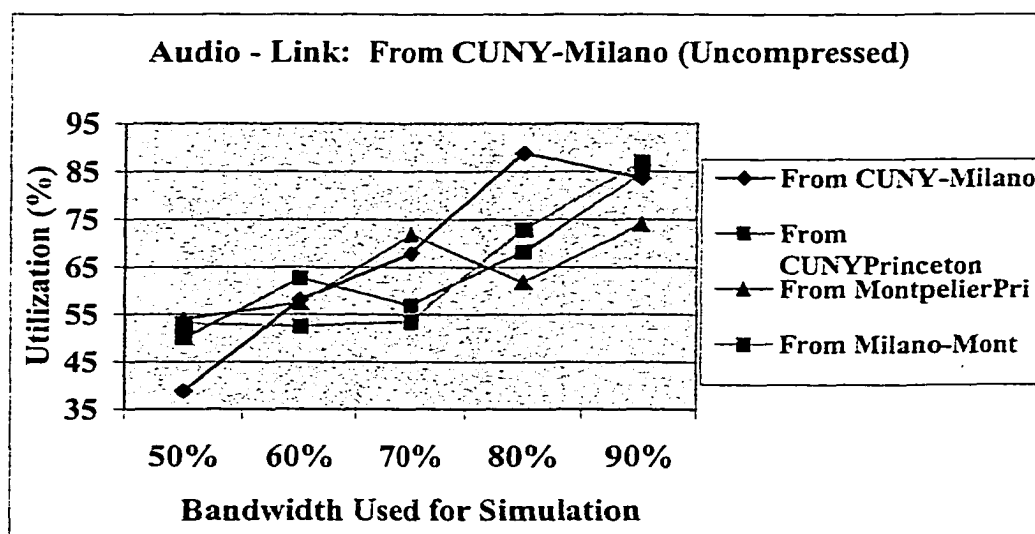


Figure 9. 32: Audio (SimRun1) – Link Utilization (%) (Uncompressed Data)

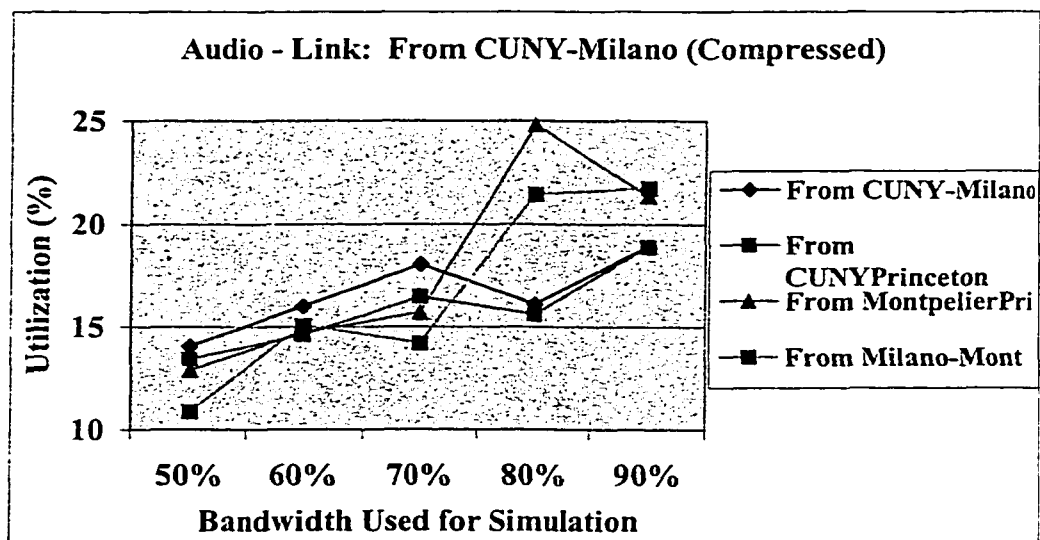


Figure 9. 33: Audio (SimRun2) – Link Utilization (%) (Compressed Data)

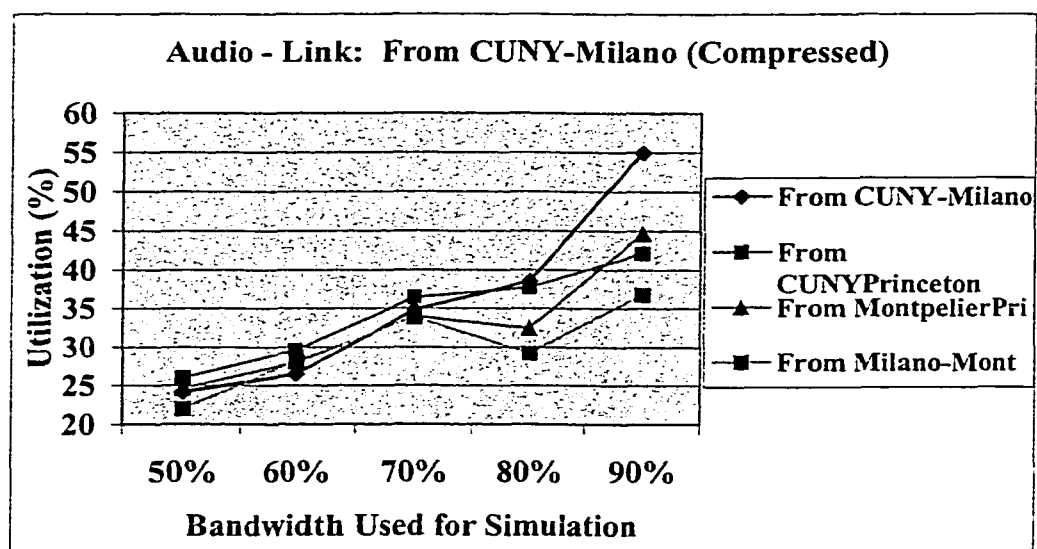


Figure 9. 34: Audio (SimRun2) – Link Utilization (%) (Uncompressed Data)

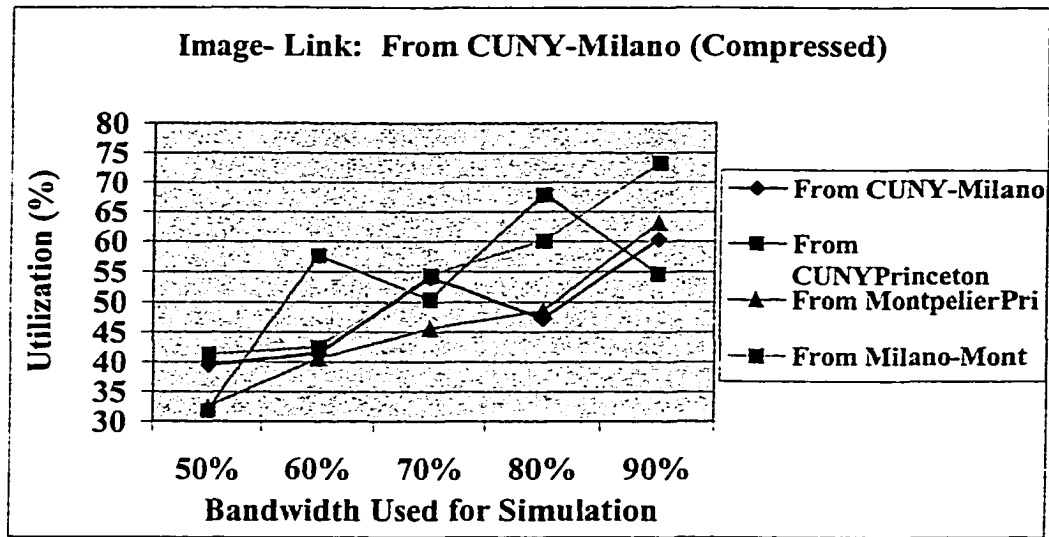


Figure 9. 35: Image – Link Utilization (%) (Compressed Data)

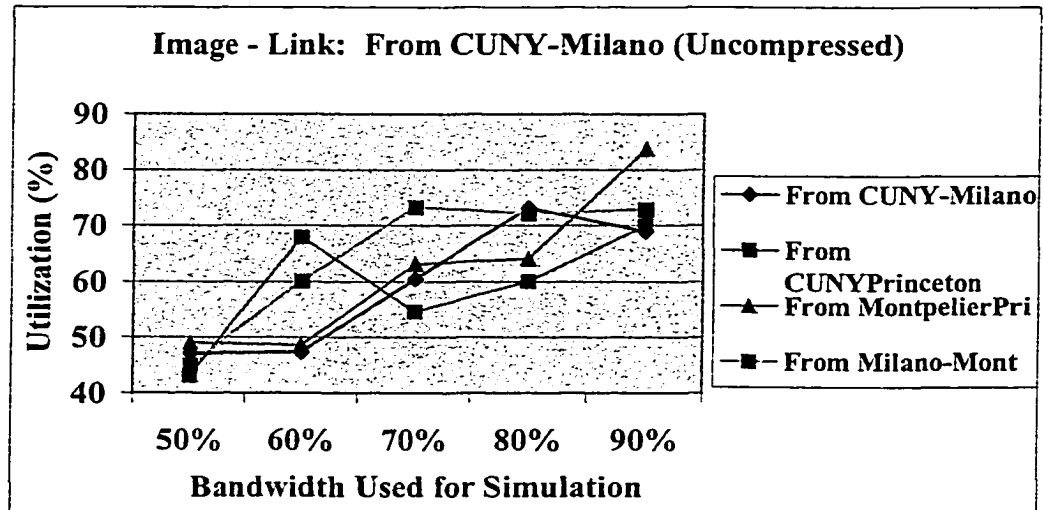


Figure 9. 36: Image – Link Utilization (%) (Uncompressed Data)

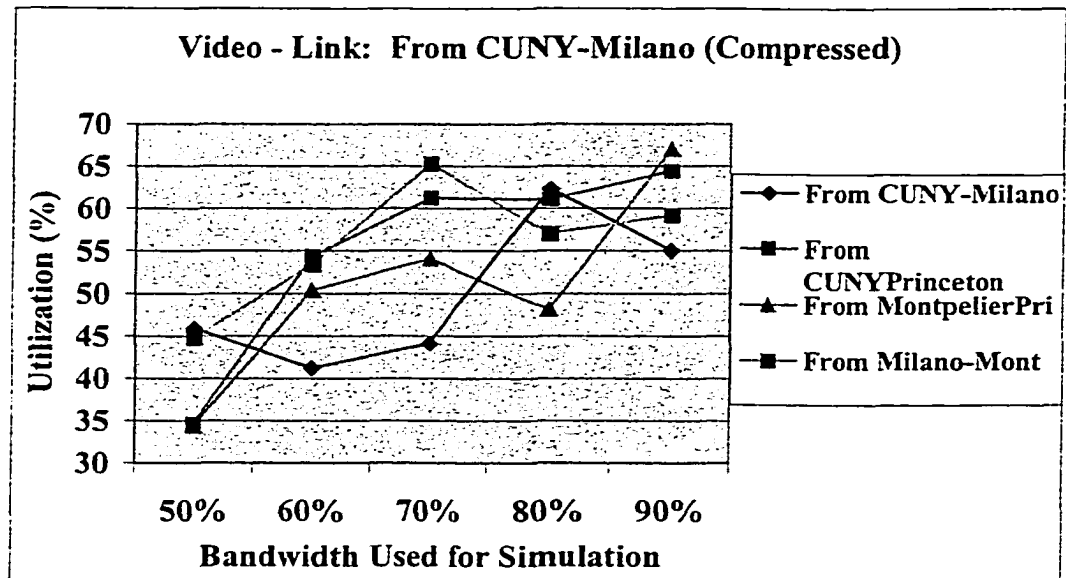


Figure 9. 37: Video – Link Utilization (%) (Compressed Data)

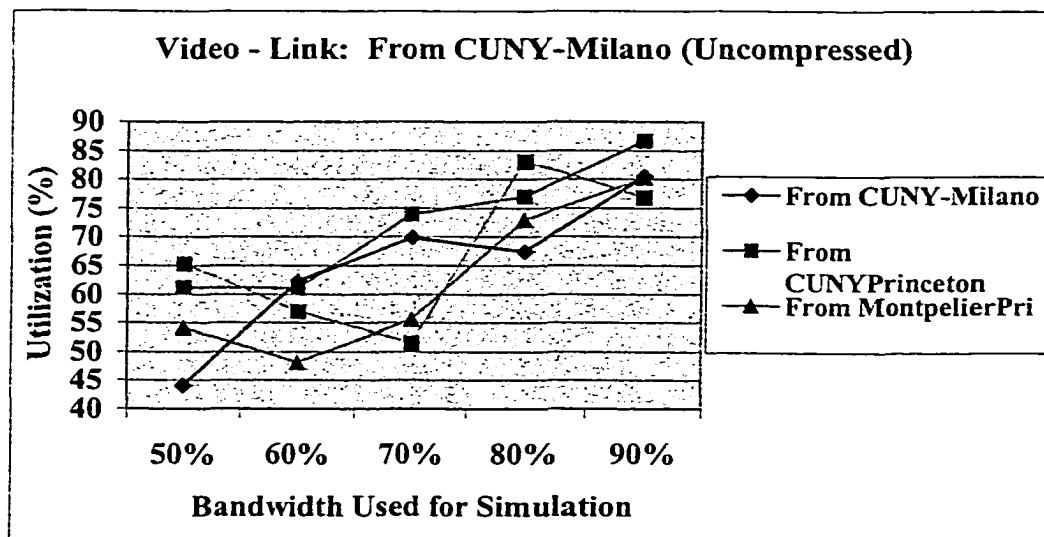


Figure 9. 38: Video – Link Utilization (%) (Uncompressed Data)

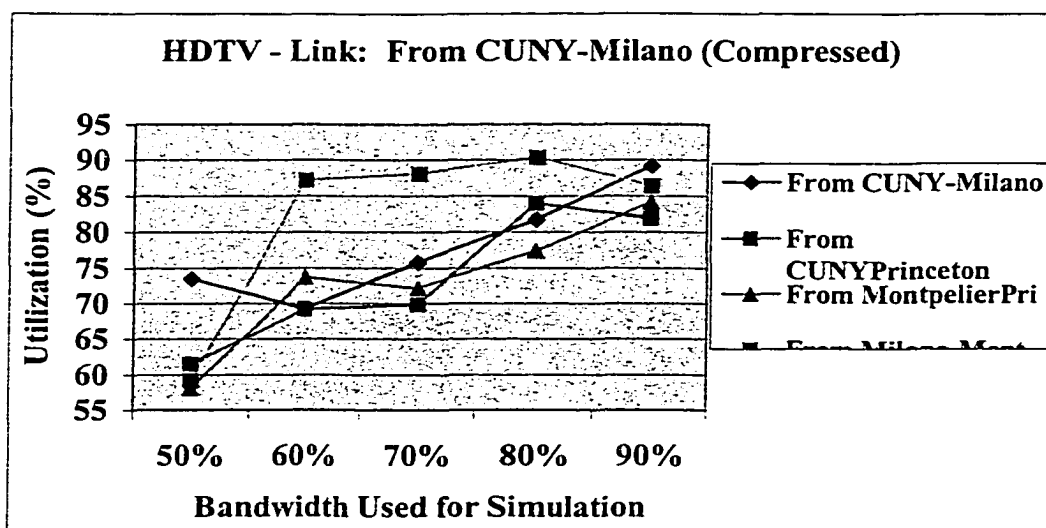


Figure 9. 39: HDTV – Link Utilization (%) (Compressed Data)

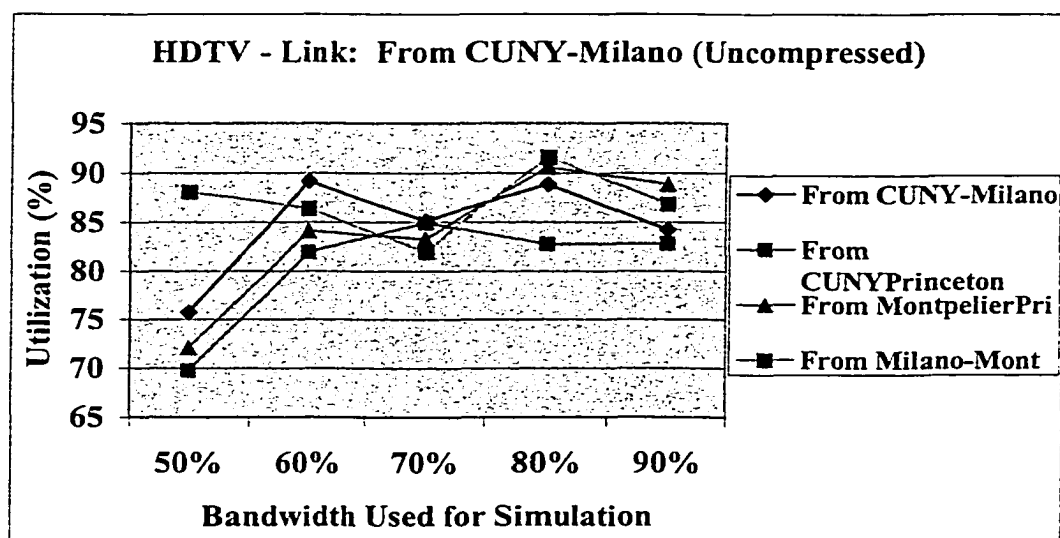


Figure 9. 40: HDTV – Link Utilization (%) (Uncompressed Data)

- **WAN Cloud**

For each application, the results indicate that the number of frames accepted by access links for compressed data is less than that for uncompressed data in most cases. The number of frames accepted of compressed data is more than that of uncompressed data for some links in some bandwidth simulation runs. This could be due to the network delay or link failure at the time the data are collected. Similarly the number of frames dropped of compressed data is smaller because of the smaller number of frames accepted while the drop rate (error) is approximately equal for both. The difference between the two drop-rates is zero or less than the margin of error of $\pm 5\%$. Hence, the error rate is the same for both data types.

The link utilization for E-mail, Audio, Video, and HDTV applications of compressed data is less than that of uncompressed data. However, the link utilization of Image is close to 100% for both data types. This can be attributed to the enormous amount of data being transmitted and/or network/transmission delay.

Table 9.66 below shows the ranges of frames accepted, frames dropped and the drop-rate for compressed and uncompressed data by application. In case

of each application, the ranges for compressed data are lower than that of uncompressed data. This demonstrates the improvement in the transmission of compressed data over uncompressed data. **Figures 9.41-9.54** depict the frames accepted by access link over the ranges of the bandwidths for each of the five applications.

Table 9.67 presents a comparison of the drop-rate by access link and application for compressed and uncompressed data. The drop-rates for compressed and uncompressed data are close to each other for each of the applications. The drop-rates, however, vary somewhat from one access link to another suggesting the possibility of variation in the number of packets sent from source to destination. The drop rates across applications, access-destination links and over the bandwidth percentages are on the low side suggesting that the error rate is small for both data types.

		Ranges	
Application	Compression Algorithm	Compressed Data	Uncompressed Data
E-mail	Huffman		
Frames Accepted		850 – 2700	1150 - 3150
Frames Dropped		91 – 269	136 – 365
Drop Rate (%)		8 – 13	9 - 13
E-mail	LZW		
Frames Accepted		593 - 1452	1150 - 3150
Frames Dropped		64 - 160	136 – 365
Drop Rate (%)		10 - 13	9 - 13
Audio	MPEG-1	Simulation Run1	
Frames Accepted		650 – 1650	1250 - 1870
Frames Dropped		66 – 641	257 – 1584
Drop Rate (%)		6 – 41	20 - 26
Audio	MPEG-1	Simulation Run2	
Frames Accepted		357 - 809	650 – 1650
Frames Dropped		37 - 96	66 – 641
Drop Rate (%)		7 - 25	6 – 41
Image	MPEG-1		
Frames Accepted		11800 – 14280	13900 – 14350
Frames Dropped		1280 – 1656	1499 – 1622
Drop Rate (%)		11 – 12	10 - 12
Video	MPEG-2		
Frames Accepted		13400 – 28000	17000 – 34500
Frames Dropped		1490 – 3146	1939 – 3896
Drop Rate (%)		11 – 12	11 - 12
HDTV	MPEG-2		
Frames Accepted		22700 – 37500	27000 – 37200
Frames Dropped		2612 – 4243	3077 – 4243
Drop Rate (%)		11 – 12	11 - 12

Table 9. 66: Ranges of Frames Accepted, Frames Dropped & Drop Rate

	Comparison of Drop Rate (%) for Compressed & Uncompressed Data									
	Bandwidth Used for Simulation									
Cloud: Access Link	Cmp 50%	Uncmp 50%	Cmp 60%	Uncmp 60%	Cmp 70%	Uncmp 70%	Cmp 80%	Uncmp 80%	Cmp 90%	Uncmp 90%
CUNY-Montpelier										
E-mail (Huffman, 40%)	9%	10%	8%	13%	10%	12%	13%	11%	12%	12%
E-mail (LZW, 60%)	10%	10%	13%	13%	13%	12%	13%	11%	12%	12%
Audio (SimRun1, 50%)	10%	26%	8%	39%	19%	72%	19%	57%	25%	60%
Audio (SimRun2, 50%)	12%	10%	9%	8%	10%	19%	13%	19%	25%	25%
Image (MPEG-1, 50%)	11%	11%	12%	11%	11%	12%	11%	11%	12%	11%
Video (MPEG-2, 30%)	12%	12%	11%	11%	12%	11%	11%	11%	11%	11%
HDTV MPEG-2, 30%)	11%	11%	11%	11%	11%	12%	11%	11%	11%	11%
CUNY-Milano										
E-mail (Huffman, 40%)	12%	10%	12%	12%	10%	12%	11%	11%	12%	12%
E-mail (LZW, 60%)	13%	10%	11%	12%	11%	12%	10%	11%	12%	12%
Audio (SimRun1, 50%)	9%	20%	8%	44%	11%	57%	19%	90%	41%	81%
Audio (SimRun2, 50%)	10%	9%	11%	8%	11%	11%	9%	19%	15%	41%
Image (MPEG-1, 50%)	11%	11%	11%	11%	11%	11%	11%	11%	11%	12%
Video (MPEG-2, 30%)	12%	11%	12%	11%	11%	11%	11%	11%	11%	11%
HDTV MPEG-2, 30%)	12%	11%	11%	11%	11%	11%	11%	11%	11%	11%
CUNY-Bern										
E-mail (Huffman, 40%)	12%	13%	11%	11%	10%	11%	12%	11%	11%	13%
E-mail (LZW, 60%)	11%	13%	12%	11%	11%	11%	10%	11%	10%	13%
Audio (SimRun1, 50%)	10%	28%	12%	48%	13%	48%	35%	75%	34%	74%
Audio (SimRun2, 50%)	10%	10%	10%	12%	10%	13%	10%	35%	11%	34%
Image (MPEG-1, 50%)	11%	11%	11%	12%	11%	11%	12%	11%	11%	11%
Video (MPEG-2, 30%)	11%	11%	11%	11%	11%	11%	11%	11%	11%	11%
HDTV MPEG-2, 30%)	11%	11%	11%	11%	11%	11%	11%	11%	11%	11%

	Comparison of Drop Rate (%) for Compressed & Uncompressed Data (Cont'd.) Bandwidth Used for Simulation									
Cloud: Access Link	Cmp 50%	Uncmp 50%	Cmp 60%	Uncmp 60%	Cmp 70%	Uncmp 70%	Cmp 80%	Uncmp 80%	Cmp 90%	Uncmp 90%
CUNY-Bonn										
E-mail (Huffman, 40%)	11%	12%	11%	12%	11%	9%	12%	12%	12%	11%
E-mail (LZW, 60%)	12%	12%	13%	12%	10%	9%	12%	12%	12%	11%
Audio (SimRun1, 50%)	11%	24%	13%	44%	19%	42%	16%	75%	19%	74%
Audio (SimRun2, 50%)	12%	11%	9%	13%	11%	19%	10%	16%	9%	19%
Image (MPEG-1, 50%)	11%	11%	12%	11%	11%	11%	11%	11%	11%	11%
Video (MPEG-2, 30%)	11%	11%	12%	11%	11%	11%	11%	11%	11%	11%
HDTV MPEG-2, 30%)	11%	11%	11%	11%	11%	11%	11%	11%	11%	12%
Princeton-Montpelier										
E-mail (Huffman, 40%)	11%	11%	13%	11%	13%	12%	10%	11%	12%	11%
E-mail (LZW, 60%)	10%	11%	12%	11%	12%	12%	11%	11%	10%	11%
Audio (SimRun1, 50%)	12%	38%	9%	42%	16%	61%	13%	50%	23%	79%
Audio (SimRun2, 50%)	10%	12%	11%	9%	9%	16%	9%	13%	12%	23%
Image (MPEG-1, 50%)	11%	11%	11%	11%	11%	11%	11%	11%	11%	10%
Video (MPEG-2, 30%)	11%	11%	11%	11%	11%	11%	11%	11%	12%	11%
HDTV MPEG-2, 30%)	11%	11%	11%	11%	11%	11%	11%	11%	11%	11%
CUNY-Princeton										
E-mail (Huffman, 40%)	13%	10%	11%	11%	12%	11%	12%	12%	10%	11%
E-mail (LZW, 60%)	12%	10%	12%	11%	12%	11%	10%	12%	12%	11%
Audio (SimRun1, 50%)	13%	33%	14%	44%	21%	39%	22%	54%	17%	79%
Audio (SimRun2, 50%)	10%	13%	13%	14%	10%	21%	9%	22%	8%	17%
Image (MPEG-1, 50%)	11%	11%	11%	11%	11%	11%	11%	11%	11%	11%
Video (MPEG-2, 30%)	11%	11%	11%	11%	11%	11%	11%	11%	11%	11%
HDTV MPEG-2, 30%)	11%	11%	11%	11%	11%	11%	11%	11%	11%	11%

	Comparison of Drop Rate (%) for Compressed & Uncompressed Data (Cont'd.)									
	Bandwidth Used for Simulation									
Cloud: Access Link	Cmp 50%	Uncmp 50%	Cmp 60%	Uncmp 60%	Cmp 70%	Uncmp 70%	Cmp 80%	Uncmp 80%	Cmp 90%	Uncmp 90%
Montpelier-Milano										
E-mail (Huffman, 40%)	12%	12%	12%	10%	12%	12%	10%	12%	11%	11%
E-mail (LZW, 60%)	11%	12%	10%	10%	10%	12%	11%	12%	11%	11%
Audio (SimRun1, 50%)	9%	32%	8%	31%	14%	39%	6%	65%	15%	86%
Audio (SimRun2, 50%)	13%	9%	12%	8%	7%	14%	12%	6%	12%	15%
Image (MPEG-1, 50%)	11%	11%	12%	12%	11%	12%	12%	11%	12%	11%
Video (MPEG-2, 30%)	12%	11%	12%	11%	11%	11%	11%	11%	11%	11%
HDTV MPEG-2, 30%)	11%	11%	11%	11%	11%	11%	11%	11%	11%	11%

Table 9. 67: Comparison of Drop Rate (%) for Compressed and Uncompressed Data

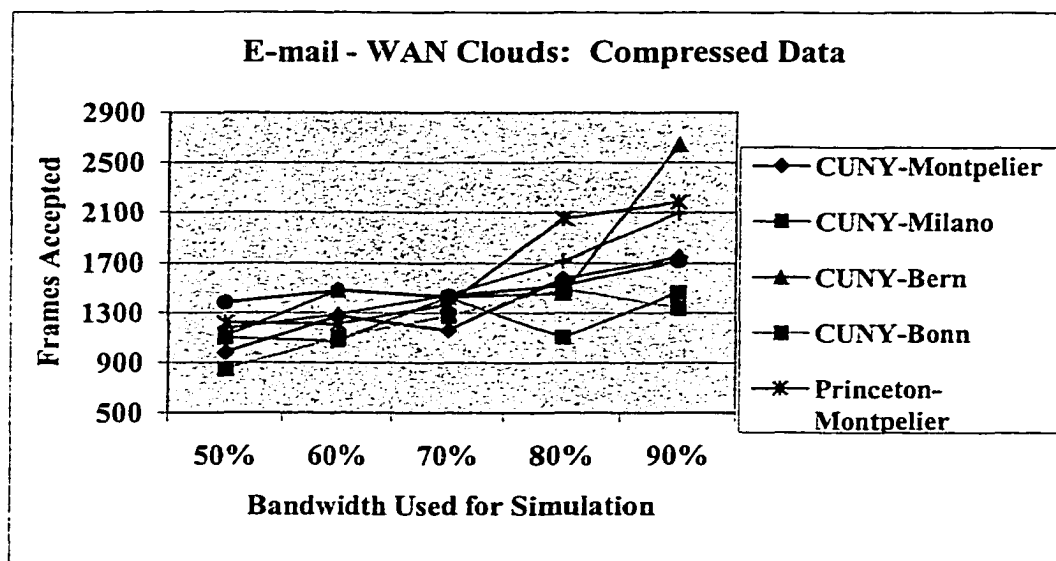


Figure 9. 41: E-mail (Huffman) - WAN Cloud: Frames Accepted (Compressed Data)

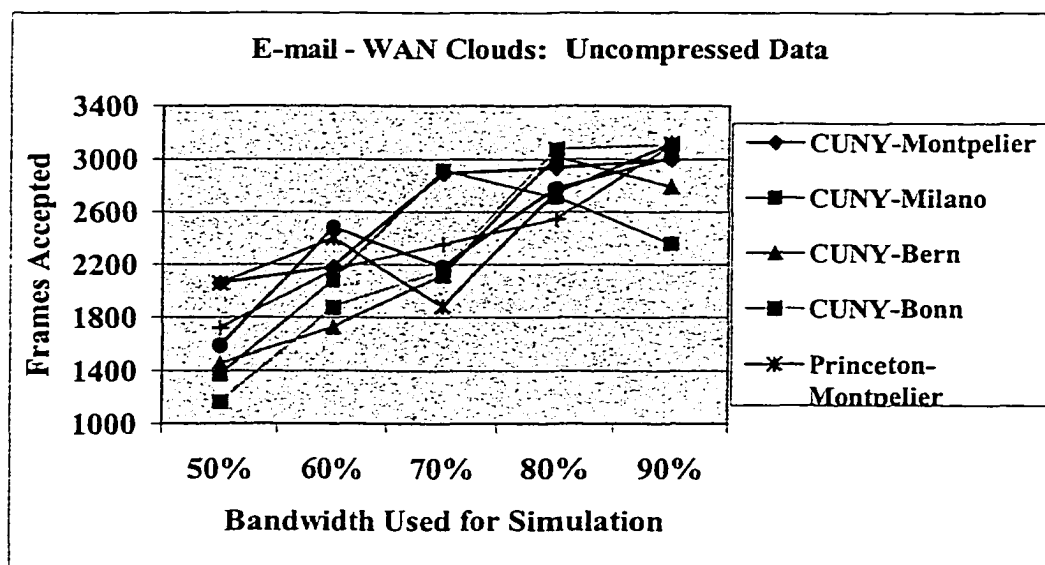


Figure 9. 42: E-mail (Huffman) - WAN Clouds: Frames Accepted (Uncompressed Data)

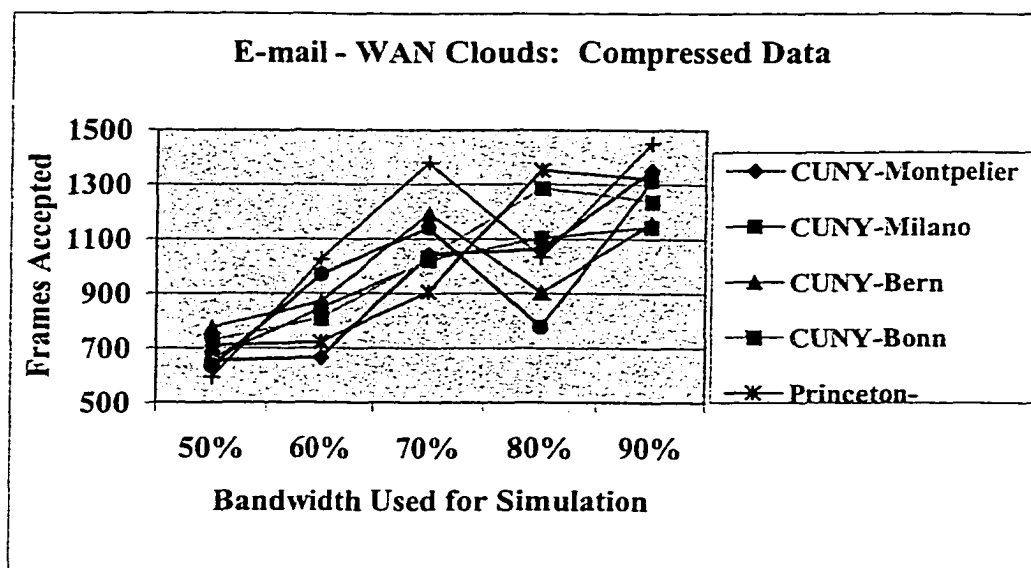


Figure 9. 43: E-mail (LZW) - WAN Cloud: Frames Accepted (Compressed Data)

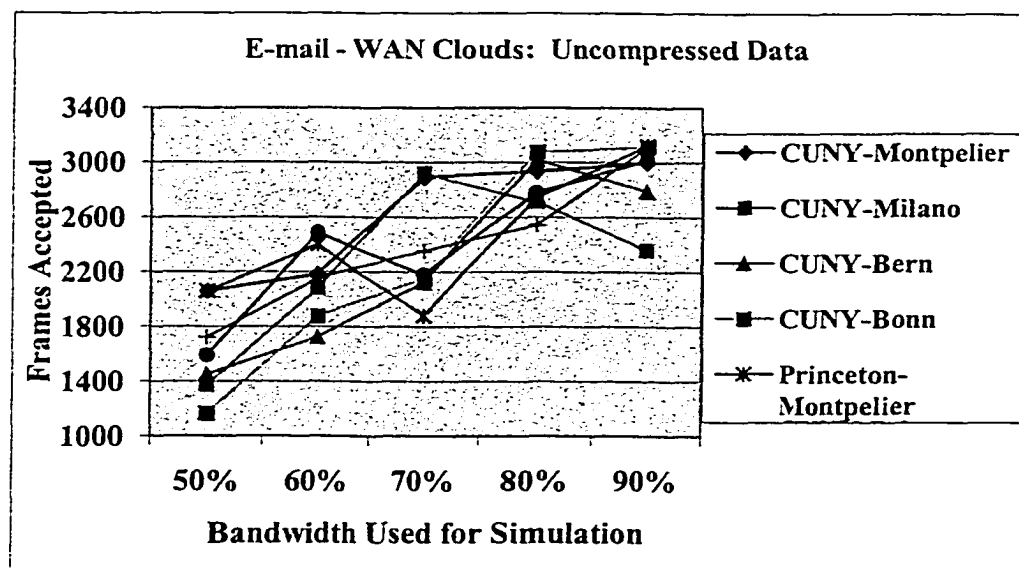


Figure 9. 44: E-mail (LZW) - WAN Clouds: Frames Accepted (Uncompressed Data)

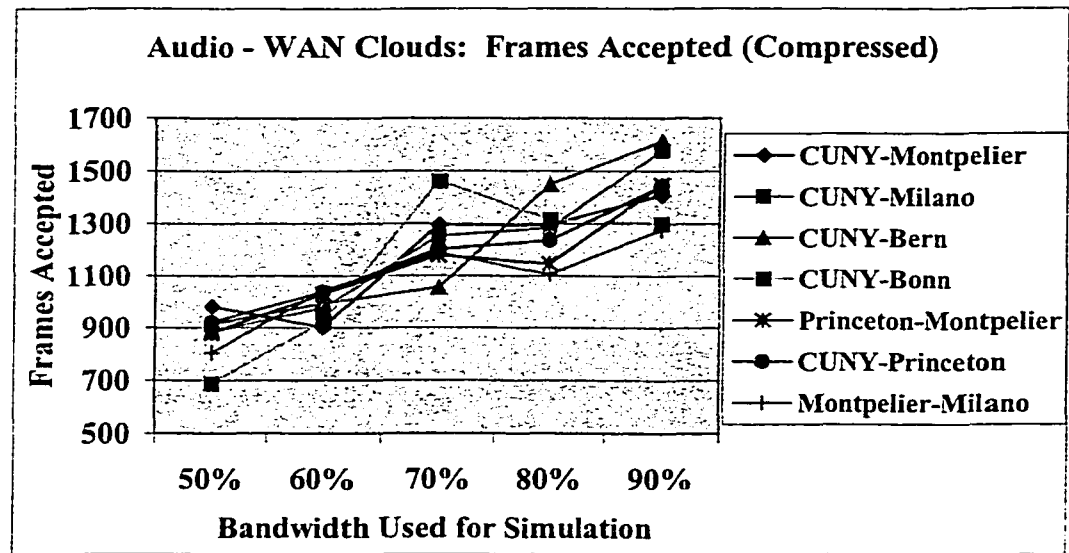


Figure 9. 45: Audio (SimRun1) - WAN Clouds: Frames Accepted (Compressed Data)

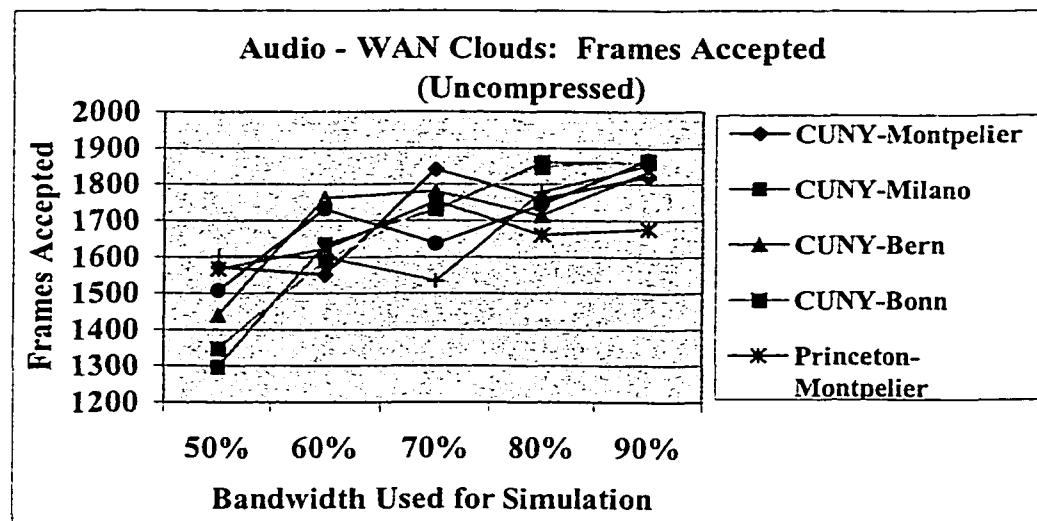


Figure 9. 46: Audio (SimRun1) - WAN Clouds: Frames Accepted (Uncompressed Data)

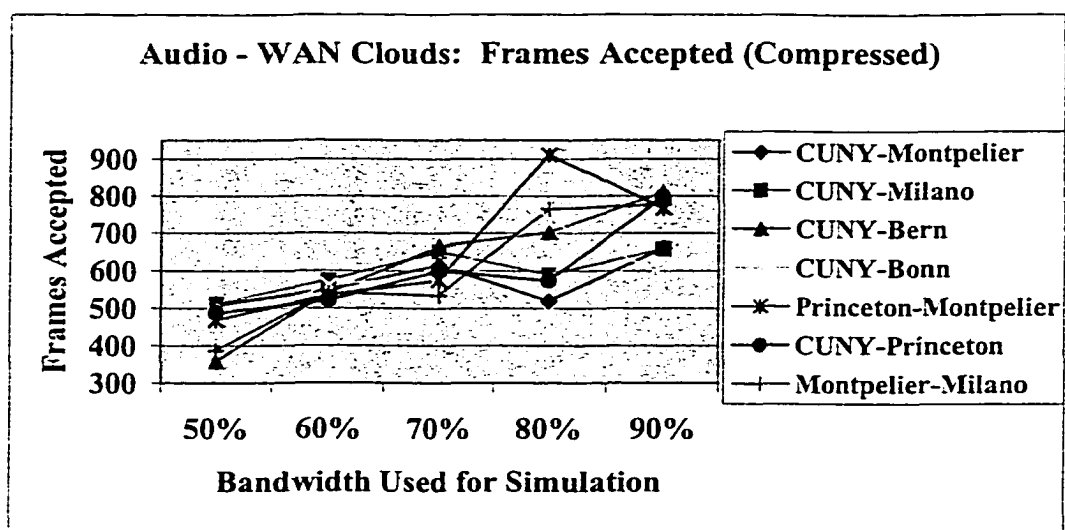


Figure 9. 47: Audio (SimRun2) - WAN Clouds: Frames Accepted (Compressed Data)

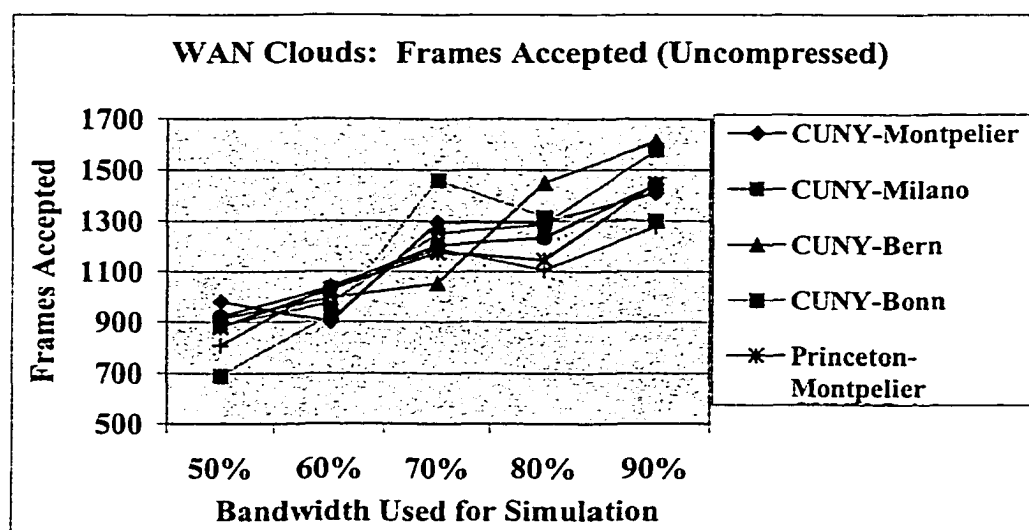


Figure 9. 48: Audio (SimRun2) - WAN Clouds: Frames Accepted (Uncompressed Data)

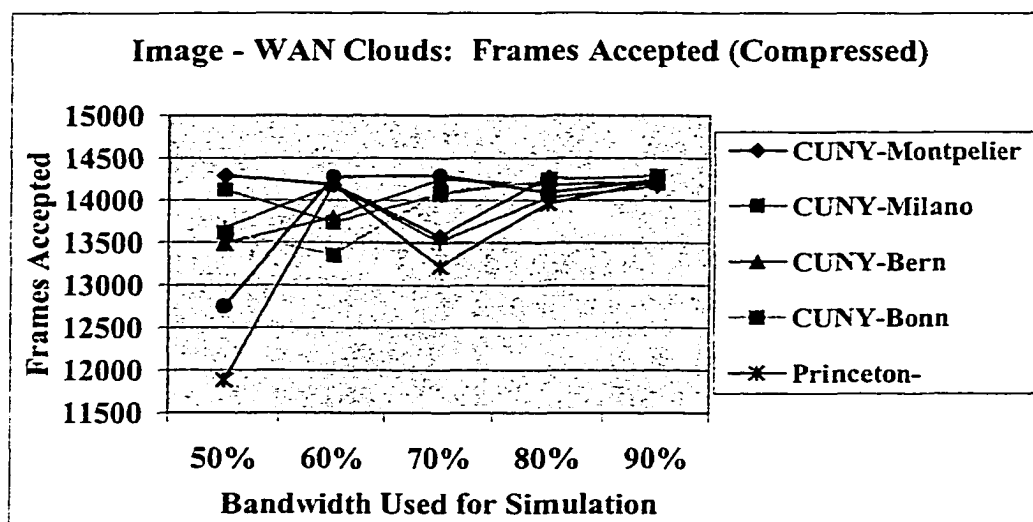


Figure 9. 49: Image - WAN Clouds: Frames Accepted (Compressed Data)

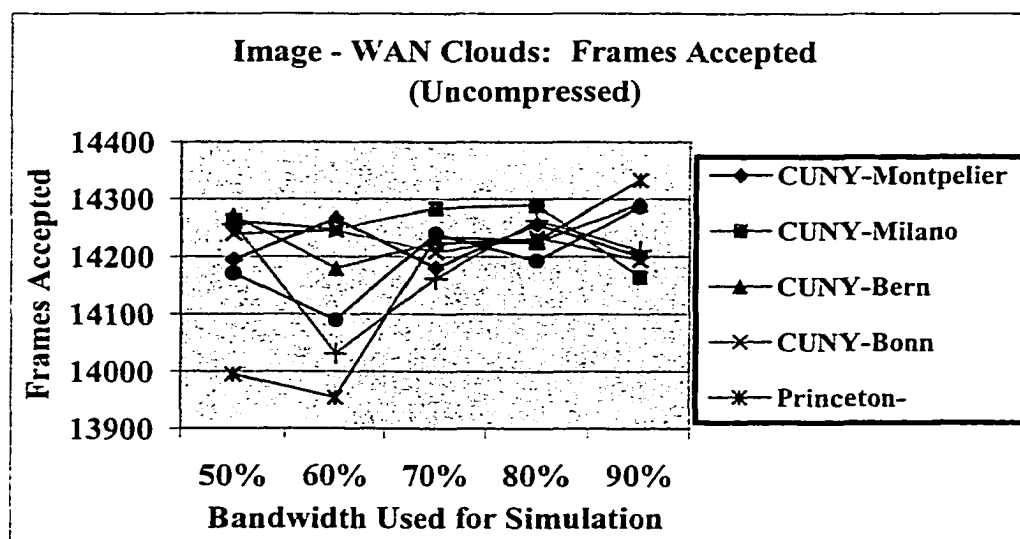


Figure 9. 50: Image - WAN Clouds: Frames Accepted (Uncompressed Data)

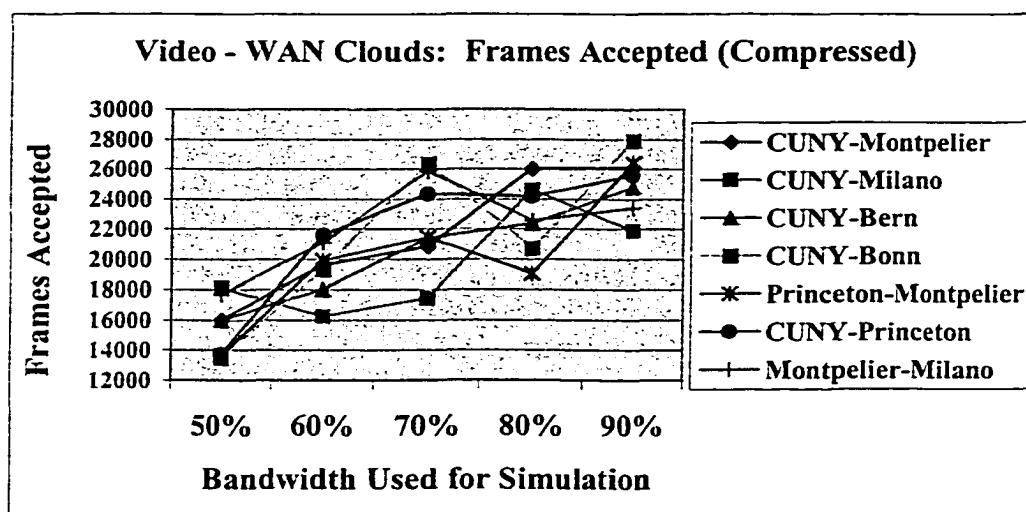


Figure 9. 51: Video - WAN Clouds: Frames Accepted (Compressed Data)

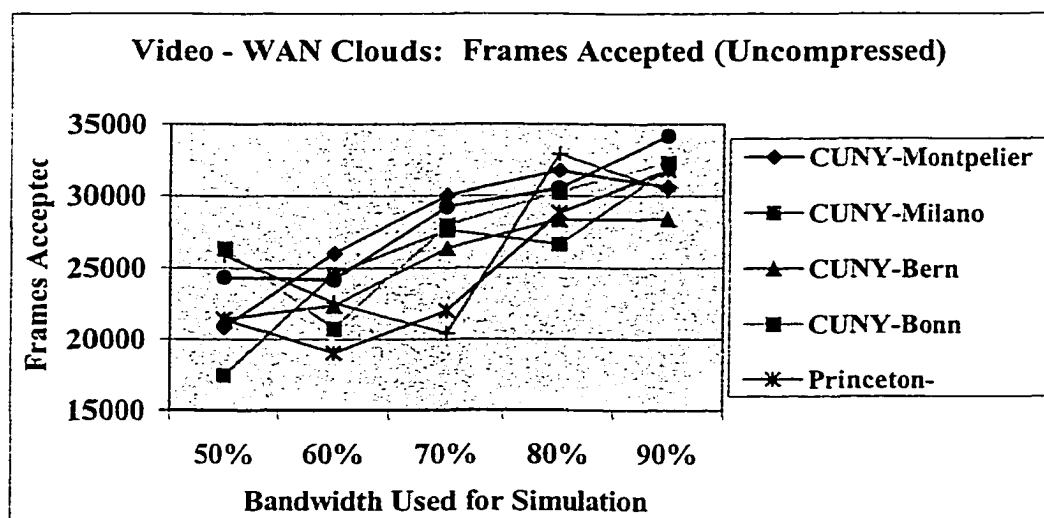


Figure 9. 52: Video - WAN Clouds: Frames Accepted (Uncompressed Data)

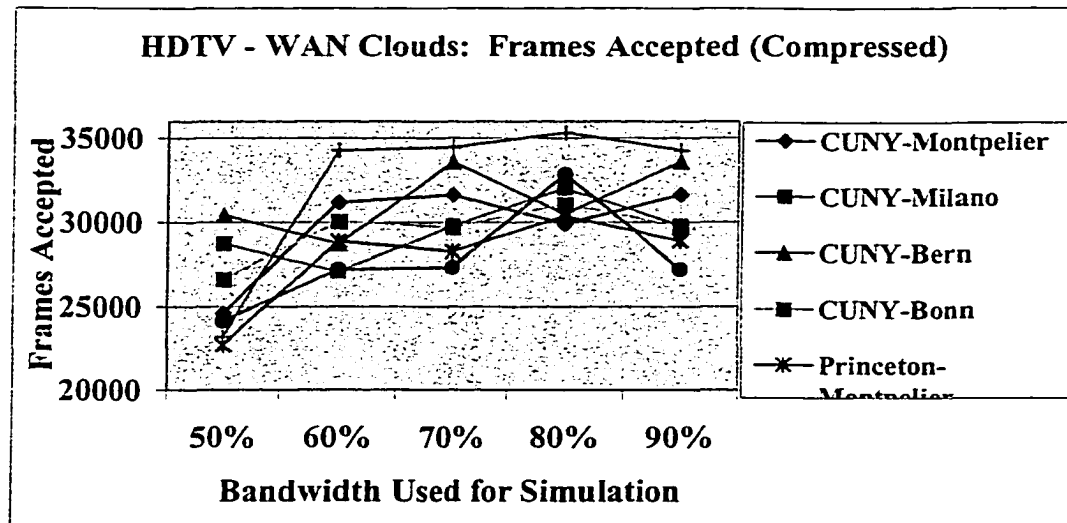


Figure 9. 53: HDTV - WAN Clouds: Frames Accepted (Compressed Data)

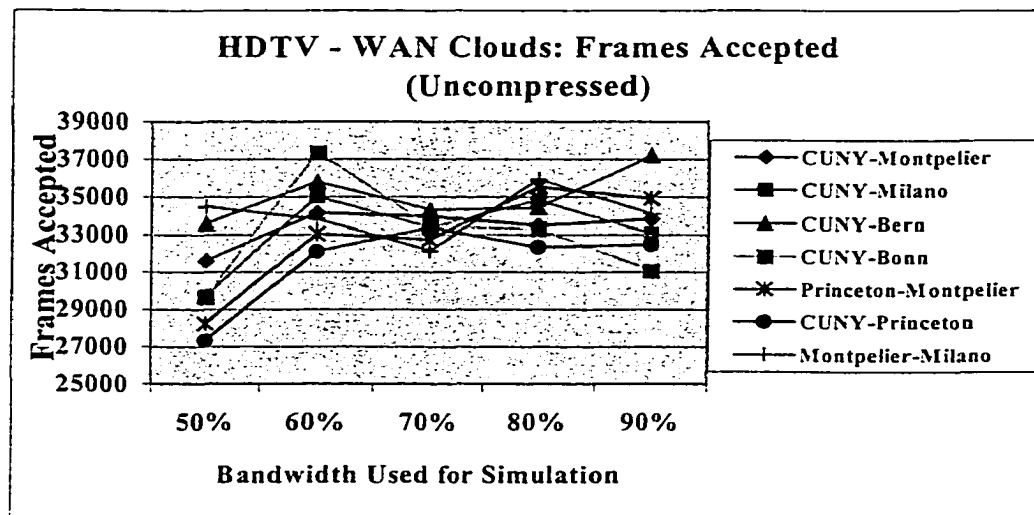


Figure 9. 54: HDTV - WAN Clouds: Frames Accepted (Uncompressed Data)

- **Message Source: Packets Delivered/Delay**

Packets Created: The number of packets/frames created of compressed data is less than that of uncompressed data. This number increases with message size. Since the message size of E-mail and Audio applications is not large, the number of packets created is considerably less for both data types. For Image, Video, and HDTV applications the number of packets created is very large.

Packets Delivered: The number of packets delivered may differ from the number of packets created depending on the number of packets that are in transit at the instant the data are collected. The number of packets delivered, however, is less for compressed data.

Packets Dropped: The percentage of packets dropped over created (dropped/created) is below 100% for both data types. However, the percentage of dropped over delivered (dropped/delivered) is very high for both, but the percentage varies for different applications. This shows that more packets are dropped than delivered because of network delay or link/node failure. Link failure can be considered the absence of a transmission link between origin and destination nodes. From the results of

simulation runs, it is evident that a link failure has a negative impact on throughput.

Average Packet Delay: Average packet delay is a measure of network performance as far as packet delivery is concerned. The results indicate that for low traffic intensities the average packet delay is low. The data show that the average time of packet delay is less for compressed data.

Table 9.68 below shows the ranges of packets delivered, packets dropped and the drop-rate (packets dropped/packets delivered) for compressed and uncompressed data by application. In case of each application, the ranges for compressed data are lower than that of uncompressed data. This demonstrates the improvement in the transmission of compressed data over uncompressed data. **Figures 9.55-9.68** depict the packets delivered by link over the ranges of the bandwidths for each of the five applications.

		Ranges	
Application	Compression Algorithm	Compressed Data	Uncompressed Data
E-mail	Huffman		
Packets Delivered		330 – 750	225 – 610
Packets Dropped		129 – 347	161 – 435
Drop Rate (%)		30 – 75	41 – 142
E-mail	LZW		
Packets Delivered		323 - 633	330 – 750
Packets Dropped		113 - 235	129 – 347
Drop Rate (%)		24 - 76	30 – 75
Audio	MPEG-1	Simulation Run1	
Packets Delivered		300 – 700	260 – 635
Packets Dropped		121 – 667	308 – 1731
Drop Rate (%)		24 – 219	55 - 505
Audio	MPEG-1	Simulation Run2	
Packets Delivered		319 - 680	300 – 700
Packets Dropped		75 - 404	121 – 667
Drop Rate (%)		24 - 109	24 – 219
Image	MPEG-1		
Packets Delivered		1000 – 2300	1250 – 2250
Packets Dropped		1726 – 22888	5537 – 33671
Drop Rate (%)		77 – 1308	290 – 2119
Video	MPEG-2		
Packets Delivered		3940 – 8500	3900 - 9000
Packets Dropped		1994 – 10755	2648 – 16368
Drop Rate (%)		37 – 207	40 - 251
HDTV	MPEG-2		
Packets Delivered		3950 – 7200	3950 – 7200
Packets Dropped		7402 – 58035	25482 – 92207
Drop Rate (%)		151 – 1102	424 - 2033

Table 9. 68: Ranges of Packets Delivered, Packets Dropped & Drop Rate

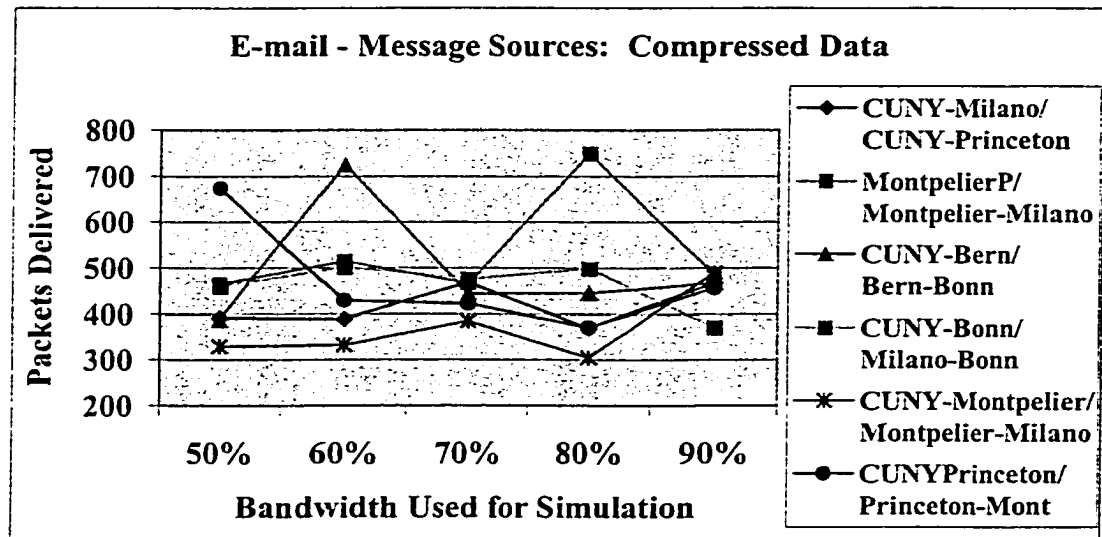


Figure 9. 55: E-mail (Huffman) - Message Sources: Packets Delivered (Compressed Data)

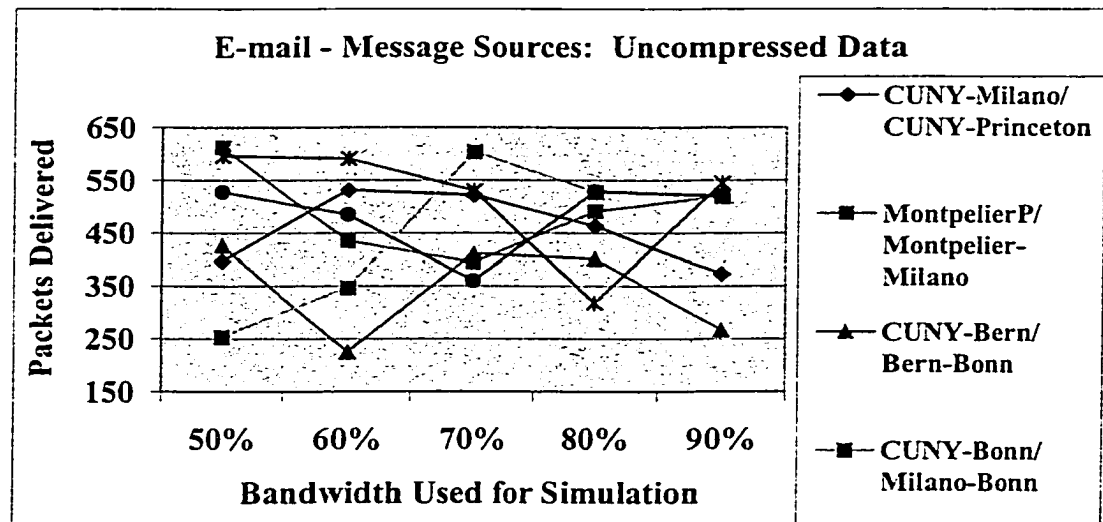


Figure 9. 56: E-mail (Huffman) - Message Sources: Packets Delivered (Uncompressed Data)

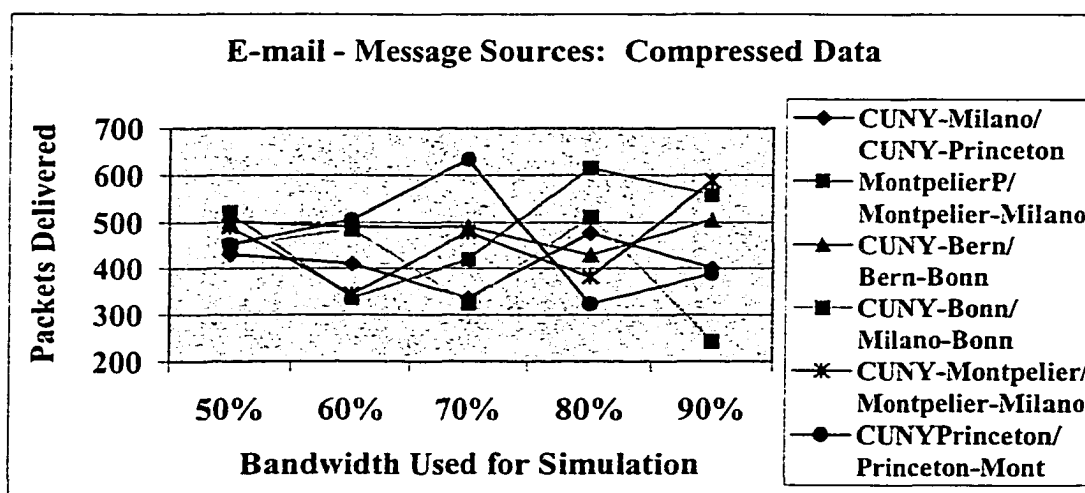


Figure 9. 57: E-mail (LZW) - Message Sources: Packets Delivered (Compressed Data)

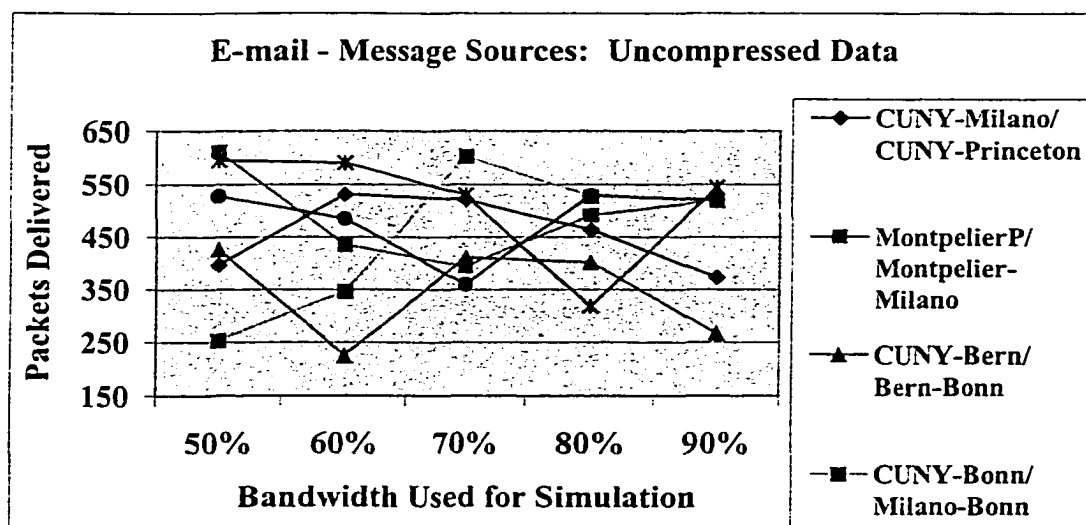


Figure 9. 58: E-mail (LZW) - Message Sources: Packets Delivered (Uncompressed Data)

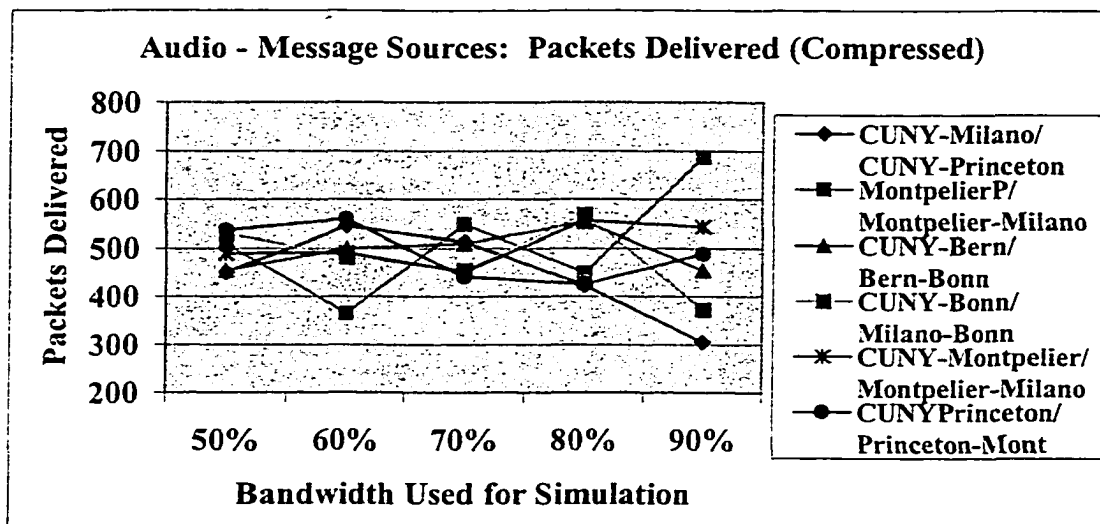


Figure 9. 59: Audio (SimRun1) - Message Sources: Packets Delivered (Compressed Data)

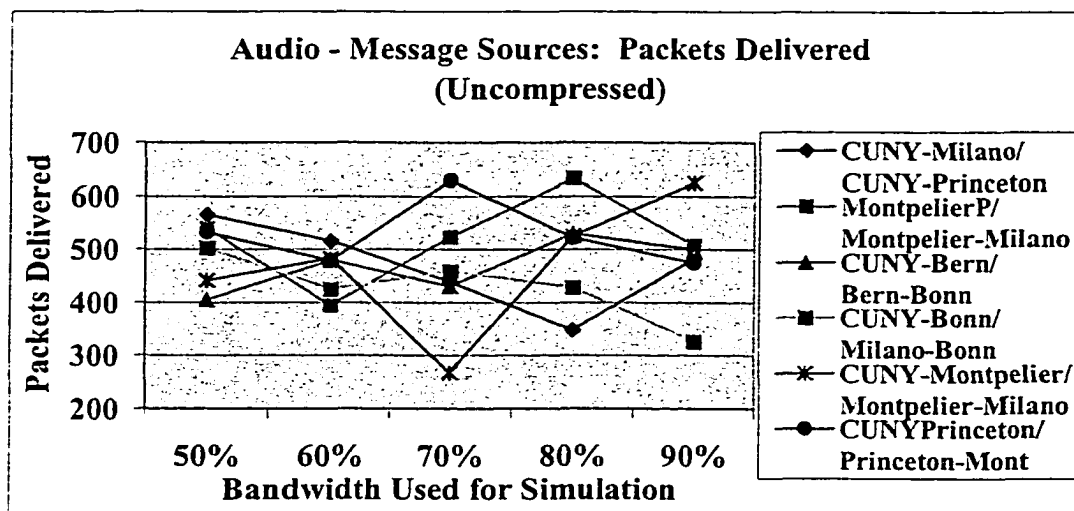


Figure 9. 60: Audio (SimRun1) - Message Sources: Packets Delivered (Uncompressed Data)

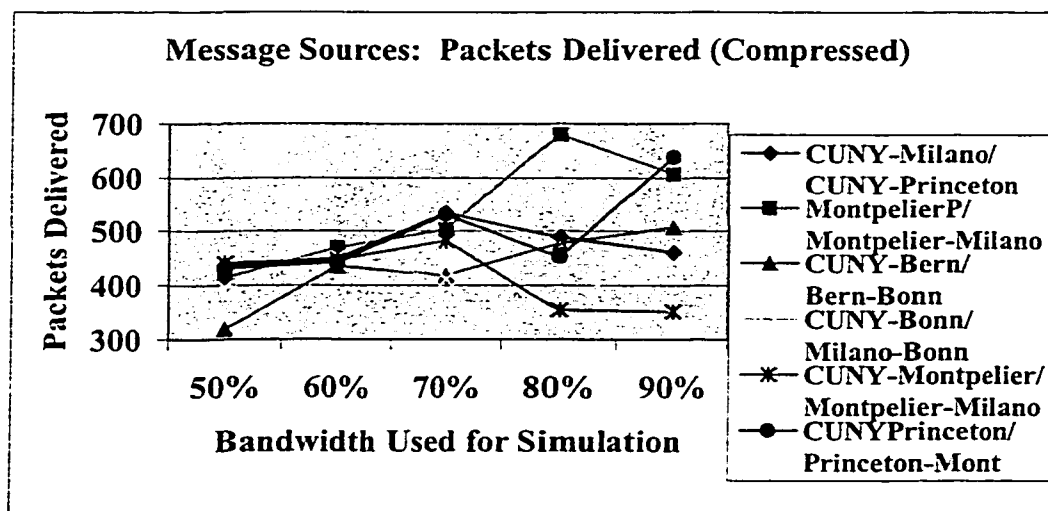


Figure 9. 61: Audio (SimRun2) - Message Sources: Packets Delivered (Compressed Data)

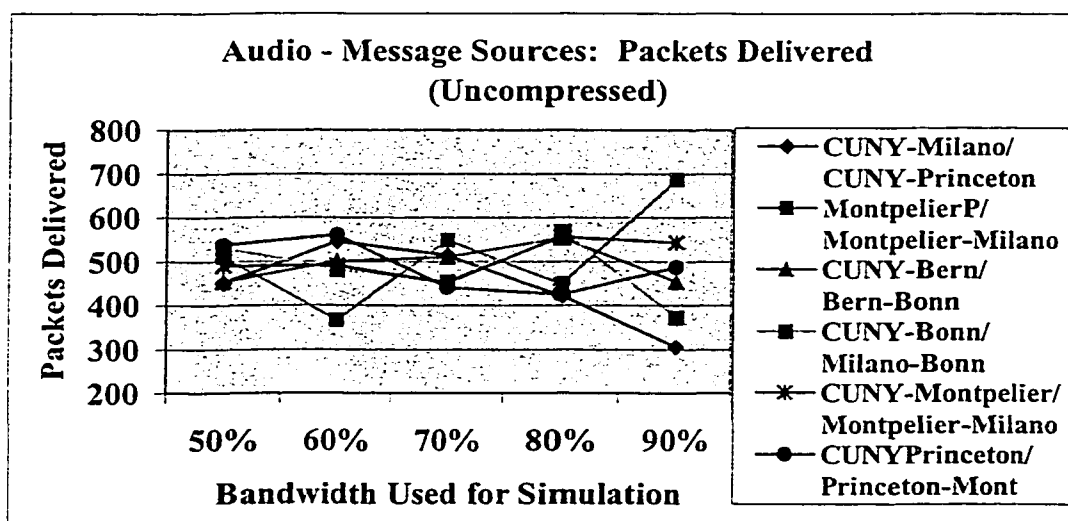


Figure 9. 62: Audio (SimRun2) - Message Sources: Packets Delivered (Uncompressed Data)

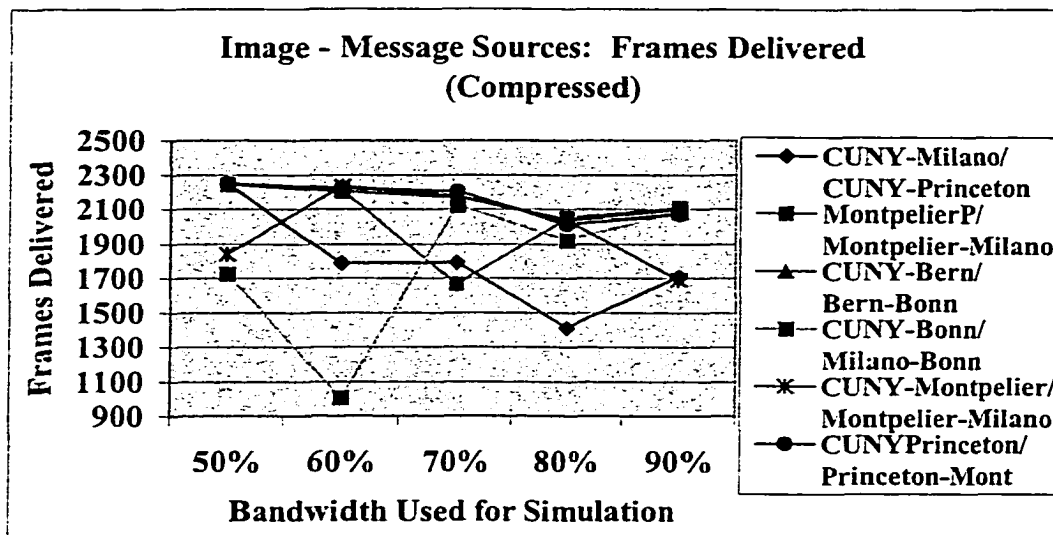


Figure 9. 63: Image - Message Sources: Frames Delivered (Compressed Data)

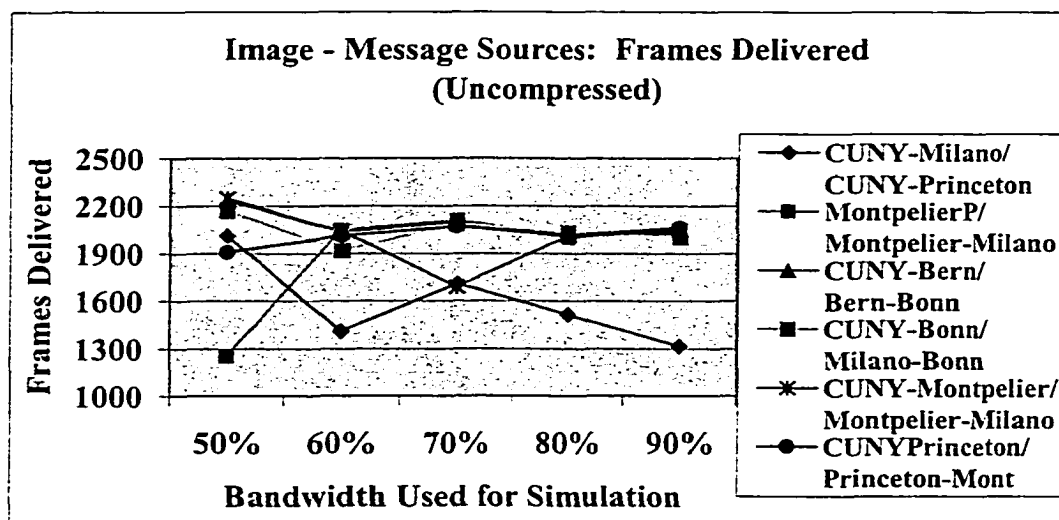


Figure 9. 64: Image - Message Sources: Frames Delivered (Uncompressed Data)

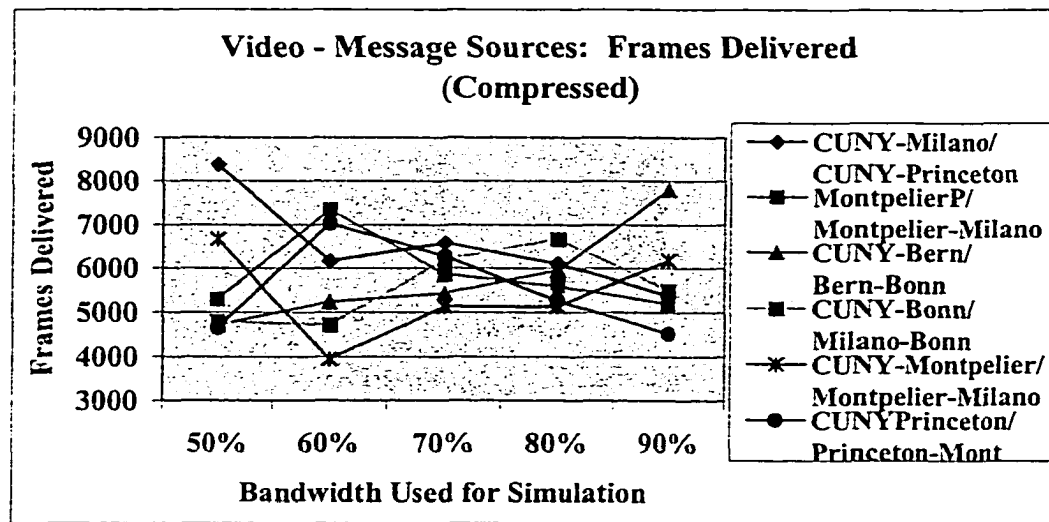


Figure 9. 65: Video - Message Sources: Frames Delivered (Compressed Data)

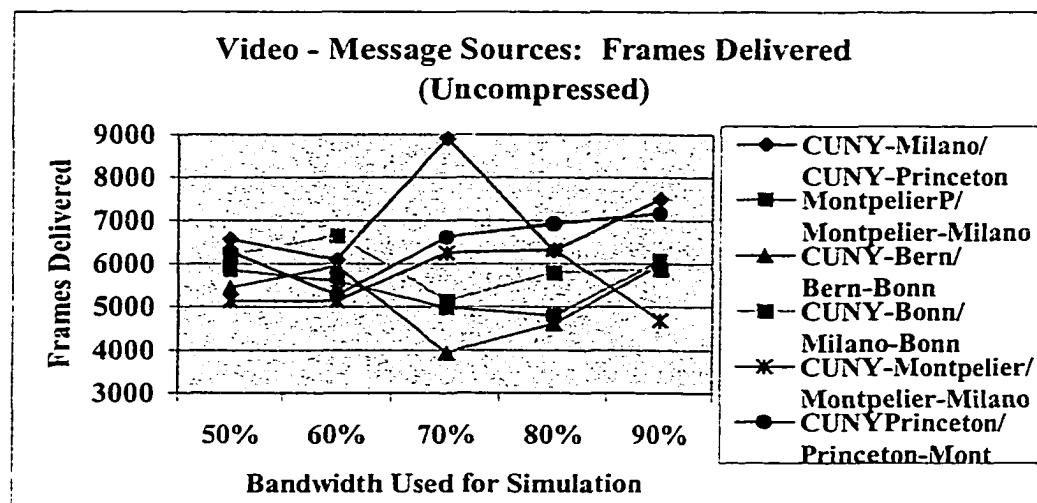


Figure 9. 66: Video - Message Sources: Frames Delivered (Uncompressed Data)

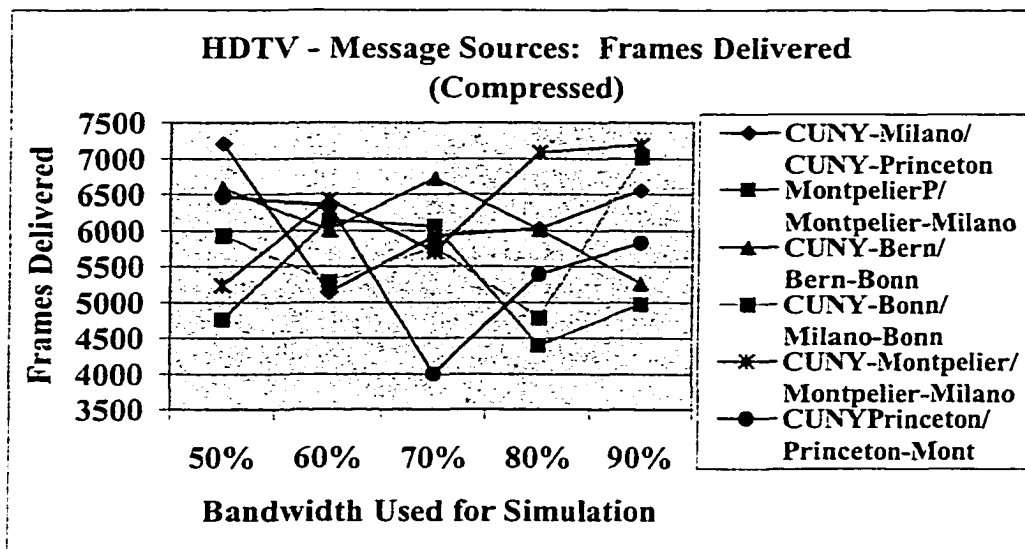


Figure 9. 67: HDTV - Message Sources: Frames Delivered (Compressed Data)

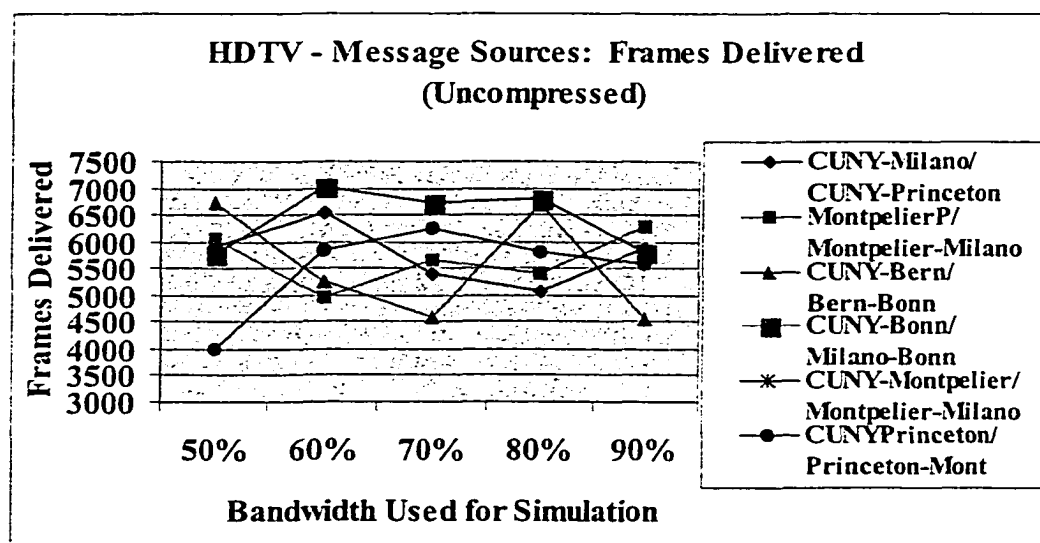


Figure 9. 68: HDTV - Message Sources: Frames Delivered (Uncompressed Data)

Chapter 10

10. SUMMARY/CONCLUSIONS

10.1. Summary (Dissertation)

This study represents an exploration of the advantages of compression of data for storage and transmission. The significance of compressing data cannot be overestimated in light of the mushrooming applications and globalization of communication. The advantages of compressing data manifest themselves in the form of efficiency/cost effectiveness. Compression of data is of particular significance or an imperative for the developing countries (DCs) because of their relatively underdeveloped telecommunications infrastructure. Data compression

in the case of DCs may overcome the varied infrastructure bottlenecks as well as represent a case of 'leap-frogging.'

The study traces the development of telecommunications networks in all their breath taking diversity and breakthroughs in Chapters 2, 3, and 4. The significance of the advances in the telecommunications/computer networks is measured in terms of their diversity of applications in myriad ways that we feel in our daily lives.

Chapter 5 builds a case for the relevance of data compression in DCs in the backdrop of their poor telecommunication infrastructure. The study uses India as an illustrative case providing a historical perspective on the development of telecommunications and data networks in the country. India presents a clear case of the usefulness of data compression. The country also becomes, with its recent developments in the field, an example of 'leap-frogging.'

Chapter 6 outlines the different data compression algorithms and discusses their distinguishing features in terms of applications. Techniques discussed are Run-Length Coding, Huffman Coding, Predictive Coding, Block Coding, Lossy Image Compression, Sequential Coding, motion compression algorithms (MPEG-1, MPEG-2) and Image Synthesis.

Data compression applications to communication networks are discussed in Chapter 7. The applications covered are e-mail, audio, image, video, and HDTV. The chapter highlights the significance of data compression to communication networks in light of the ever growing user demand for both storage and transmission of data/information. This is particularly important in the face of computer access to libraries and databases and the impact of the Internet.

The importance of simulation as a tool for modeling a network to determine its performance characteristics is stressed in Chapter 8. The components of the simulation topology, i.e., nodes, links and arcs and simulation methodology pertaining to link utilization, WAN cloud, message sources are presented in the chapter, with graphical illustrations.

Analysis and discussion of simulation results is the refrain of Chapter 9. Tables showing simulation results for e-mail, audio, image, video, and HDTV using Huffman Coding, LZW, MPEG-1, and MPEG-2 are presented with graphical illustrations. Tables also compare the simulation results in terms of link utilization rate (Links), frames accepted/dropped/link utilization rate (WAN cloud), and packets dropped/delay (Message and Response Sources) for

compressed and uncompressed data over the range of 50% to 90% of bandwidth used for simulation.

10.2. Conclusions (Results)

Simulation runs, presented in Chapter 9, ranged over five different applications, namely e-mail, audio, image, video, and HDTV. Links, WAN Clouds, and Message and Response Sources are used in simulating the results. Five bandwidth percentages are used to observe the variation in utilization rate (Links), drop rate and access link utilization rate (WAN Clouds), and packets-dropped/packets-delivered ratio (Message and Response Sources). Channel utilization rate, in the case of links, is an accurate reflection of the difference in efficiency between transmission of compressed and uncompressed data. Drop rate and access link utilization rate, in the case of WAN Clouds, similarly reflects the difference in transmission efficiency between compressed and uncompressed data. Packets-dropped/packets-delivered ratio similarly is a measure of the comparative efficiency between the two data types. The conclusions that follow reflect these three measures of comparison, namely utilization rate, drop rate and access link utilization rate, and packets-dropped/packets-delivered ratio.

Simulation results presented with analysis and discussion (Ch 9), yield the following overall conclusions:

1. The utilization rate for compressed data, in case of Links, is less than that of uncompressed data over the range of the five applications. The utilization rate increases for both compressed and uncompressed data as the bandwidth percentage (traffic intensity) increases. The utilization rate for compressed data is still less than for uncompressed data demonstrating the advantage of compressing data for transmission.
2. The drop rate for compressed data, in case of WAN Clouds, is less than for uncompressed data over the range of applications. However, the difference in the drop rate between the compressed and uncompressed data narrows as the bandwidth utilization increases except in the case of audio. The drop rate for uncompressed data, in case of audio, increases substantially as the bandwidth utilization increases, thus widening the difference in the drop rate between compressed and uncompressed data. The access link utilization rate, another measure of comparison between compressed and uncompressed data, follows the pattern observed in the case of links, i.e., less for compressed than for uncompressed data. And access link utilization rate increases as the bandwidth utilization increases just as in the case of links.
3. The packets-dropped/packets-delivered ratio, in case of Message and Response Sources, follows the same pattern as in the case of Links and WAN

Clouds, over the five different applications. The ratio increases as the bandwidth utilization increases similar to the case with Links and WAN Clouds. For the same message rate transmitted the received message is likely to have far less error than the uncompressed mode of transmission.

4. The comparative efficiency in the transmission of compressed over uncompressed data narrows as the bandwidth percentage (traffic intensity) increases. Stated alternatively the effect of compression loses its advantage as the network becomes extremely saturated.
5. The comparative efficiency of transmitting compressed over uncompressed data in link utilization is higher in the case of e-mail, audio, image and video applications and narrows as we move to HDTV application. This is because of the type of compression algorithm used for the different applications. The details are documented in **Table 7.1** in Chapter 7.
6. Access link utilization rate (WAN Cloud) for both compressed and uncompressed data reaches close to 100% in the case of image application. However, the utilization rate is lower (more efficient) for compressed data over uncompressed data in the other four applications. This is a corollary of conclusion 5.
7. As the message size increases substantially (100B for e-mail to 93.225MB for HDTV) more than half the packets get dropped through the network for both compressed and uncompressed data for the corresponding simulations.

However, the range of percentage for compressed data is still lower than for uncompressed data.

Whereas conclusion 1 is evident to a network designer, conclusions 2-7 have considerable quantitative information that can be inferred by reading the results in Chapter 9 (see **Tables 9.66** and **9.67**). This information is valuable in choosing the right algorithm for any given application depending upon the network conditions and the saturation in the backbone of the network. Conclusion 5 is especially valuable for the new HDTV industry in developing countries (DCs) in view of the fact that the compression algorithms are close to being optimal.

The conclusions drawn above, reflecting the overall efficiency rates of transmitting compressed and uncompressed data, confirm the improvement in the efficiency of transmitting compressed data. The methodology used in the dissertation can be used for evaluating new compression algorithms through the evolving modern communication networks and data storage systems of the future. Further, the quantitative data indicates cost effectiveness of using these compression algorithms in the DCs where the networks tend to be more congested. In the case of HDTV the marginal gain by using compression algorithm is low and for the other four applications the gain can be substantial.

This process of improvement has been spurred on by the ever-growing user demand for transmission (and storage) of information and the imperative of cost effectiveness.

The Postwar II period ushered in a new multifaceted era of advances in science and technology; emergence of multinational corporations; and globalization of economic activity. This change has and will continue to put heavy user demand on communication systems and networks. Increased competition globally in the production and transport of goods and delivery of services underlines the need for effective communication. For data storage and transmission, compression becomes imperative for cost effectiveness.

Glossary

#

1ESS, 1A ESS, 2B ESS, 3ESS	precursor ESS systems introduced before the current 4ESS and 5ESS
2BIQ	two binary to one quaternary code for BRISDN
4ESS	Number 4 electronic switching system for toll and trunk switching
5ESS	a generalized electronic switching system for most types of CS, PS, ISDN, subscriber loop carrier and trunk switching
5ESS 2000	a new 5ESS platform and concept to contend with wireless, intelligent network functions and almost all of the routine 5ESS functions

A

AAL	ATM adaptation layer
ABS	alternate billing service
ACP	action control point
ADPCM	adaptive delta pulse code modulation
ADSN	analog derived services digital network.
AFD	A type of digital switching; exchange or center
AIN	Advanced Intelligent Network
AL	application layer

A-law, μ-law	Two different systems used to encode voice or audio signals which involve compression of the sampled signal according to a fixed formula.
ANI	automatic number identification
ANSI	American National Standards Institute
AP	adjunct processors
APS	automatic protection switch
ARPANET	Advanced Research Project Agency Network
ASN	adjunct service node
ASP	adjunct service point
ASST	Azienda di Stato per I Servizi Telefonici
AT&T	American Telephone and Telegraph
ATM	asynchronous transfer mode
AWC	area wide centrex
AWG	American wire gauge

B

B Channel	A 64-kbps information bearing channel in ISDN
B1 Channel	the number one B channel of BRISDN service that is used to carry 64 kbps PCM voice data for telephone service
B2 Channel	the number two B channel of BRISDN service that can be used to carry packet-switched data
BER	bit error rate

BIP	bit interleaved parity
BISDN	broadband integrated services digital network
BRISDN	basic rate integrated services digital network
BW	bandwidth
BX.25	a specialized Bellcore protocol to communicate service management information and provides an interface for maintenance, security, and operations of the SCP

C

CA&O	centralized administration and coordination
CATV	cable television
CCAF	call control access function
CCF	call control function
CCIS	Common Channel Interoffice Signaling
CCIS6	Common Channel Interoffice Signaling 6
CCISS	Common Channel Interoffice Signaling System
CCISS6	Common Channel Interoffice Signaling System 6
CCISS7	Common Channel Interoffice Signaling System 7
CCITT	International Telegraph and Telephone Consultative Committee, now called ITU-T
CCSS	Common Channel Signaling System.
CCS7	Common Channel Signaling 7
CCSS7	Common Channel Signaling System 7

CD	compact disk
Centrex	a widespread telephone-company switching service that uses Central Office switching equipment and to which customers connect via individual extension access lines
CF	control function
Channel	A transmission path between two points
Channel Bank	Channel terminal equipment used for combining (multiplexing) channels on a frequency-division or time-division basis
Circuit-mode	Type of switching that causes a one-to-one correspondence between a call and a circuit
CLASS	customized local area special services
CLP	cell loss priority
CO	Central Office; a telephone company switching facility or center, at which a diversity of transmission channels, including subscribers' local loops terminate
CPE	customer premises equipment
CPU	central processing unit
CSMA/CA	carrier sense multiple access/collision avoidance
CSMA/CD	carrier sense multiple access/collision detect
CT	computed tomography
CU	computer unit

D

D-channel	delta channel; in ISDN a 16 kbps (BRISDN) signaling channel for basic rate access, or a 64 kbps (PRISDN) signaling channel with other access rates
DAP	data access point
DCP	database control point
DCs	developing countries
DCT	Discrete Cosine Transform. A mathematical process that converts a set of values into an equivalent representation as the sum of cosine functions.
DDD	direct distance dialing
DDSN	Digital Derived Services Network; the British Telecom IN
DE	discard eligible
DLL	data link layer
DOT	Department of Telecommunication
DPCM	differential pulse code modulation
DQDB	distributed queue dual bus (IEEE 802.6).
DS1	Digital Signal 1, a formatted signal transmitted at 1.544 Mbps, usually referred to as T1 signal
DS2	Digital Signal 2, a formatted signal transmitted at 6.312 Mbps, usually referred to as T2 signal
DS3	standard third level digital carrier system used in the North America at 44.736 Mbps; also known as T3 signal

DS4	fourth level digital carrier system used in the North America at 274.176 Mbps; also known as T4 signal (Japanese equivalency at 397.2 Mbps)
DSP	digital signal processing
DTE	data terminal equipment
DTV	Digital TV

E

EI	European transmission link with 2.048 Mbps of bandwidth capacity
EO	End office; the lowest level central office before distribution to customers or remote terminals
ERS	emergency response system/service
ESS	Electronic Switching System

F

FC	functional component, in IN/1+ and IN/2 environments
FDDI	Fiber Distributed Data Interface
FDM	frequency-division multiplexing
FDMA	frequency division multiple access
Frame	In JPEG, a group of one or more scans. For the JPEG modes in common use a frame is indistinguishable from an image.

G

Gbps	gigabits per second
GC	Group Center
GFC	generic flow control
GHz	gigahertz
GIF	Graphics interc format
GNS	Green Number Service; in Europe, the so-called "Freephone" service provided by a typical IN/1
Group 3	This apparatus uses a digitized binary representation of the facsimile. Because it is a digital scheme, it can and does use data compression and is capable of transmitting an A4-size (210 x 297 mm) document in about a minute. The apparatus is standardized in recommendation T.4.
Group 4	This apparatus has the same speed requirement as Group 3. The apparatus is standardized in recommendation T.6, T.503, T.521, and T.563.

H

HO-channel	in ISDN, a 384-kbps user-to-user information transfer (5B+D service; also know as C6)
HI-channel	in ISDN, a 1.536 Mbps user-to-user information transfer (23B+D service, also known as C24)
HDTV	high definition TV
HEC	header error check
HF	high frequency

Huffman Coding	A compression technique that uses variable-length codes to represent data.

I

IAT	Interarrival time- the time from the start of one message to the start of the next message
IDN	integrated digital network
IDTV	improved digital TV
IEC	International Electrotechnical Commission
IN	Intelligent Network
IN/1	The initial Bellcore release of the intelligent network
IN/I+	The evolution of the IN/ 1 architecture and some IN/1 elements.
IN/2	The final architecture of an intelligent network.
INAP	Intelligent network access point
INCM	intelligent networks conceptual model
INET	National Packet Switching Network
IP	intelligent peripheral
ISDN	integrated services digital network
ISMS	Integrated services management systems
ISO	International Organization for Standardization
ITU-T	International Telecommunications Union /Telecommunications Standardization Sector; formerly known as CCITT.

J

JBIG	Joint bilevel image experts; JBIG is an advanced facsimile standard that achieves performance improvements over the Group 3 and Group 4 standards.
JPEG	Joint Photographic Experts Group

K

KBPS	kilobits per second (1000 bits/second)

L

LAN	local area network. A type of high-speed (typically in the Mbps range) data communication arrangement, covering a limited geographical area, in which all parts of the transmission medium are under the control of a network administrator.
LAPD	link access procedure for the D channel
LEC	Local Exchange Carrier
LF	low frequency
LIDB	line information database

Lossy Compression A compression method that creates a close approximation of the original data. Lossy compression methods discard information that is considered less important in order to increase compression.

Lossless Compression A compression method that allows an exact copy of the original image to be retrieved.

LZ	A family of compression algorithms named after their creators, Abraham Lempel and Jacob Ziv.

LZW Lempel, Ziv, and Welch compression algorithm; the LZ variant used in GIF.

M

MAN metropolitan area network

MAS memory administration system

MB Megabyte

Mbps Megabits per second (1,000,000 bits/second)

MF multifrequency

MFOS multifunction operating system

MHz megahertz

MPEG Motion Picture Experts Group

MPEG-1 A video standard for CD-ROMs and certain TV signals

MPEG-2 A broadcast quality video standard

MTNL Mahanagar Telephone Nigam Limited

MUX Multiplexer

N

NA North America

NAP network access point

NCP network control point

NCS network control system

NETSTAR	Network subscriber transaction, administration and recording system
NGOs	Nongovernmental Organizations
NICNET	National Informatics Centre, Planning Commission of India
NIDB	network information database
NL	network layer
NM	node manager
NNI	network-to-network interface.
No. 1 ESSTM	The number one ESS switching System.
NRM	network resource manager
NSCX	network service complex
NSP	network services control point corresponding to a SCP
NSSP	network services support point
NTT	Nippon Telephone and Telegraph
NTSC	National Television Standards Committee; the NTSC signal is the de facto standard governing the format of television transmission signals.

O

OC	optical carrier
OC-n	optical carrier nth level of multiplex
OSI	open system interconnection
OSI-RM	open system interconnection- reference model

P

PABX	private automatic branch exchange
PAL	phase alternating line
PBX	private branch exchange
PCM	pulse code modulation
PCOs	Public Call Offices
PL	physical layer
PLT	payload type
PMS	physical medium sublayer
PNG	portable network group
PNSS	private network signaling system
POTS	plain old telephone service
PPS	packets per second
PPV	pay per view (with cable television)
PRISDN	primary rate ISDN
PRL	presentation layer
PSE	packet switching exchange
PSN	public/public switched network
PSTN	public switched telephone network
PVC	permanent virtual circuit

PVN private virtual networks

Q

Q.922 A standard Series of protocol definitions published by CCITT (now ITU-T).

Q.931 A standard Series of protocol definitions published by CCITT (now ITU-T).

QoS quality of service

R

RABMN remote area business message network

RAM random access memory

RBOC Regional Bell Operating Companies

RF radio frequency

RFD Rete Foni Dati, an Italian public network

S

SA service administration

SAR segmentation and reassembly

SCE service creation environment

SCF service control function

SCL service control logic

SCP service control point

SDF	service data function
SIP	Societa Italiana per l'Escercizo; the post and telecommunications administration in Italy.
SL	session layer
SLI	service logic interpreter
SLP	service logic program
SM	switching module
SMAF	service management access function
SMDS	switched multimegabit data service
SMF	service management function
SMS	service management system
SMS/2	service management system/2
SNA	Systems Network Architecture (from IBM).
SONET	synchronous optical network
SPC	stored program control
SPE	synchronous payload envelop
SRF	specialized resource function
SS	switching system
SS7	Signaling System 7
SSF	service switching function
SSP	service switching point

SSP/2	service switching point/2
STM-i	synchronous transport module level ($i = N/3 = 1, 4, 16, 64$)
STP	signaling transfer point
STS	synchronous transport signal/system
STS-n	synchronous transport signal level n , $n = 1, 3, 12, 48$, or 192).
SU	signaling unit; also Syndes Unit

T

T1	A digital carrier facility used to transmit a DS-1 formatted digital signal at 1.544 Mbps, also called T1 carrier system.
T1 or E1	primary rates for North American or European environment (see DS-n levels)
TDM	time division multiplexing
Throughput	X.25 concept indicating the average information rate of a particular virtual circuit
TL	transport layer

U

UCI	universal customer interface
UFD	A type of peripheral concentrator used with AFDs in the RFD
UHF	ultra high frequency
UISN	Universal Information Service Network

UNI user network interface

V

VBR variable bit rate

VC virtual circuit

VCI virtual channel identifier

VFN vendor feature node

VP virtual path

VPI virtual path interface

VPN virtual private network

VSAT very small aperture terminal

W

WAN wide area network

X

X.21 Interface between data terminal equipment (DTE) and data circuit-terminating equipment (DCE) for synchronous operation on public circuit-switched networks.

X.25 A CCITT (now ITU-T) recommendation that specifies the interface between user data terminal equipment (DTE) and packet-switching data circuit-terminating equipment (DCE).

X.28 DTE/DCE interface for start-stop mode data terminal equipment accessing the packet assembly/disassembly

facility (PAD) in a public data network situated in the same country.

- X.29** Procedures for the exchange of control information and user data between a packet assembly/disassembly facility (PAD) and a packet-mode DET or another PAD.
- X.75** Terminal and Transit call-control procedure and data transfer mechanisms on (typically) international circuits between packet-switched data networks.

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