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ABSTRACT

Multimedia has become an integral part of computing and communications environment, and networks are carrying ever-increasing volume of multimedia information. The main characteristics of multimedia information are high-volume and bursty traffic, with low tolerance to delay and delay variance. The legacy networks (designed in 70s and 80s) are not able to meet these requirements. Enhancements to the older networking technologies have been developed to convert these into multimedia networks. Enhancements to LANs include Switched Ethernet, Isochronous Ethernet, Fast Ethernet, 100VGAnyLAN, FDDI-II, and Synchronous FDDI. WAN options for multimedia networking include digital leased lines and ISDN. The Internet has revolutionized business and personal communications, but falls short of being a genuine multimedia network. To make the Internet capable of carrying multimedia traffic, new protocols such as MBone, ST-II, RTP, and RSVP have been developed. Internet2 is a new initiative that is aimed at overcoming the problems of throughput, delay and jitter encountered on the original Internet. One technology that was developed with multimedia networking as one of its main applications, is the Asynchronous Transfer Mode (ATM) technology. Upcoming Gigabit Ethernet technology will provide a path for upgrading current Ethernet networks into multimedia networks.

Keywords: Multimedia networks, Multimedia communications, Multimedia information, Local Area Networks (LAN), Wide Area Networks (WAN), Ethernet, Token-ring, Network performance parameters (NPPs), Synchronization accuracy specification (SAS) factors, ATM, FDDI, CDDI, Internet, Internet2, ISDN, RSVP, RTP, ST-II.

I. INTRODUCTION

Computer-based multimedia systems started appearing in the early 1990s, and became popular by mid 1990s, when most personal computers became capable of handling multimedia information. In parallel with the development of multimedia systems came the proliferation of Internet-based services. Networks were then increasingly required to carry multimedia information. Unfortunately, most legacy networks were not designed to carry multimedia traffic. As the demand for transmitting multimedia information increased, networking technologies were enhanced to be able to transmit multimedia information. New networking technologies and protocols were developed specifically for carrying multimedia information [Agnew, 1996].

This tutorial paper presents an overview of multimedia networks. Section 2 presents the parameters used to characterize multimedia traffic and network performance. The concept of Quality of Service (QoS) and its application to Multimedia Networks are discussed in Section 3. The requirements for transmitting multimedia information on a network are presented in Section 4. Transmission of multimedia over wide area networks (WANs) and local area networks (LANs) is presented in Sections 5 and 6 respectively. The Asynchronous Transfer Mode and its application to transmission of multimedia information are given in Section 7. Future directions for multimedia networking are covered in Section 8.

II. PERFORMANCE PARAMETERS

Why can some networks carry multimedia information, while other cannot? This question can be answered by studying performance parameters described in the next section. Let us first look at a real-life, non-networking scenario. When does a car meet your needs? If it can accelerate and runs faster than you will ever want. If the air conditioning system can make the car cooler than you will ever need. If it can come to a complete halt from 100 mph in 10 sec. flat; and especially if it costs much less than what you are happy to pay. In other words, if the car provides feature better than what you need, it is good enough for you. Similarly if a network provides facilities that are equal to, or better than what multimedia information transmission requires, the network can be used for transmitting multimedia information.

The demands of the multimedia traffic being transmitted on a network depend upon the type of multimedia objects contained in the multimedia stream, e.g.,. text, audio, video, and animation. These needs are characterized mainly by the required bandwidth and four factors called the Synchronization Accuracy Specification (SAS) factors. The four SAS factors are:

1. delay,
2. delay jitter,
3. delay skew, and
4. error rate.

The performance of a network is characterized by similar parameters, called the Network Performance Parameters (NPPs) [Sharda, 1999, p. 358]:

1. network throughput,
2. networking delay,
3. delay variance, and
4. error rate.

By comparing the bandwidth and the SAS factors of an application with the NPPs of a network, we can determine whether or not the network is capable of carrying multimedia traffic.

SYNCHRONIZATION ACCURACY SPECIFICATION (SAS) FACTORS

Synchronization implies the occurrence of events at the same time. For example, lip synchronization in a video implies that the movement of lips in the video clip matches the words uttered by these lips.

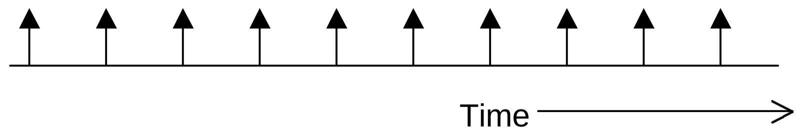
Three types of synchronization relationships can be considered:

1. asynchronous,
2. synchronous, and
3. isochronous.

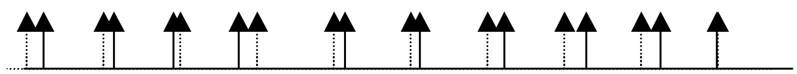
In an asynchronous system there is no well-defined timing relationship between the events. In a synchronous system, related events occur at the same time. In an isochronous system the events under consideration occur at regular intervals. Continuous multimedia elements such as audio and video comprise of a stream of elements. Digitized audio comprises a stream of samples of the audio signals, in which each sample may consist of 8-bit values for low quality audio, and 16-bit values for high quality audio. Video is transmitted as a sequence of frames. The timing relationship between consecutive frames - together with the resolution of the video frames - determines the quality of the video presentation. The SAS factors are used to specify the goodness of synchronization between the elements of the multimedia streams.

The delay factor refers to the acceptable time gap (latency) between transmission and reception of a multimedia element, such as audio sample or a video frame. The delay encountered in transmitting the elements of a multimedia object stream can vary from one element to the next. This delay variance can take two forms, namely, delay jitter and delay skew.

Jitter implies that in an object stream the actual presentation times of the various objects shift with respect to their desired presentation times. The effect of jitter on an object stream is shown in Figure 1. In Figure 1a each arrow represents the position of an object, and these objects are equally spaced in time. In Figure 1b the dotted arrows represent the desired positions of the objects and the solid arrows represent their actual positions. It can be seen in figure 1b that these objects are randomly displaced from their original positions. This effect is called jitter in the timing of the object stream. The effect of jitter, on a video clip, will be a shaky picture.



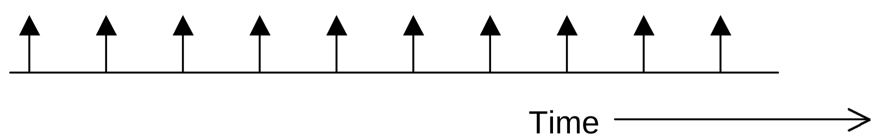
a) Object stream without Jitter



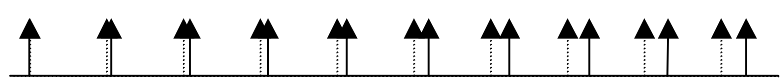
b) Object stream with Jitter

Figure 1: The Effect of Jitter on an Object Stream

Skew implies constantly increasing difference between the desired presentation times and the actual presentation times of streamed multimedia objects [Furht, 1994]. This effect is shown in Figure 2. Figure 2a shows the original object stream, and the solid arrows in Figure 2b show the object stream with skew. The effect of skew in the presentation times of consecutive frames in a video will be a slow (or fast) moving picture.



a) Object stream without Skew



b) Object stream with Skew

Figure 2: The Effect of Skew on an Object Stream

Delay jitter and skew are caused by variations in the end-to-end delay over the transmission media. Delay jitter and skew can be encountered within a medium (intramedia) as well as between media (intermedia). Intramedia delay jitter or skew refers to the changes in the presentation times of the objects in a single object stream. Intermedia delay jitter or skew occurs between two (or more) object streams.

When errors are encountered in data or multimedia objects being transmitted over a communication channel, they can be handled in one of two ways.

1. If the information being transmitted is pure text then the data packets with errors can be retransmitted.
2. If the corrupted packets carry audio or video information then these packets may have to be discarded, because retransmitted packets arrive out-of-order, and do not improve the presentation quality. They can, in fact, adversely effect the presentation quality due the additional delays caused by retransmission.

A communication channel with high error rates will lead to increased lack of synchronization in the final object stream.

The level of errors encountered on a communication channel is specified as its bit error rate (BER), and is defined as:

BER = Bit errors detected / Total number of bits transmitted in a given period

TRAFFIC CHARACTERIZATION PARAMETERS

Another important aspect of multimedia traffic is the character of the traffic in terms of the variability of the bit rate. The actual bit rate that an application generates can be either constant or variable; which gives two types of applications, namely: constant bit rate (CBR) applications, and variable bit rate (VBR) applications. Uncompressed digital voice transmission generates CBR traffic; but when it is subjected to compression it generates VBR traffic. In fact, most multimedia applications - such as, video transmission using compression - generate VBR traffic. In an application with VBR traffic the variation in the traffic volume over time is called its burstiness. The burstiness of a data stream is specified in terms of peak bit rate (PBR), and mean bit rate (MBR). A measure of burstiness is specified as the burstiness ratio, defined as [Fluckiger, 1995, p. 324]:

$$\text{Burstiness Ratio} = \text{MBR} / \text{PBR}$$

NETWORK PERFORMANCE PARAMETERS (NPP)

As listed at the beginning of Section II, the four Network Performance Parameters (NPPs) are throughput, delay, delay variance, and error rate.

Throughput is the effective rate of transmission of information bits. Often the raw bit rate of various networking technologies is specified. But, the throughput is lower than the raw bit rate because of factors such as protocol-overhead, queuing delays, congestion, and errors. For example, the throughput of an Ethernet network with 10 Mbps raw bit rate may be as low as 3 Mbps.

Delay – also called latency – is the time it takes a bit to traverse the network. It is the time difference between the instant when it is ready for transmission over the network and the instant when it is ready for use at the other end of the network. End-to-end networking delay consists of many factors, such as packetization (including depacketization) delay, transmission delay, and propagation delay. Large delays are unacceptable for most multimedia applications. The maximum acceptable delay for voice and video is 250 ms. More than the delay, variations in this delay cause problems with multimedia traffic. Delay variance can take the form of delay jitter and delay skew. Up to 10 ms of delay variance can be tolerated for voice and TV quality video signals. For compressed video and high quality stereo music the delay variance should be less than 1 ms.

By the very nature of most communication media, errors can occur on communication channels. Errors in the transmitted multimedia information lead to reduction in the quality of the multimedia presentations. The level of error expected on a network is specified in terms of three rates:

1. Bit error rate (BER) is the number of bits that get corrupted per unit time.
2. Packet Error Rate or Cell Error Rate (PER or CER) is the number of packets or cells corrupted per unit time.
3. Packet Loss Rate or Cell Loss Rate (PLR or CLR) gives the number of packets or cells lost per unit time.

The values of these NPPs, as compared to the SAS factor values required by the transmitted multimedia information, determines the quality of service provided by the network.

III.QUALITY OF SERVICE (QOS)

Users require different quality presentations at different times [ITU-T, 1988]. Different quality presentations map onto different quality of service (QoS) parameter values required of the network and the end-systems. When a multimedia presentation is transmitted via a network it translates into some requirements of the network, in terms of the NPPs. A complete QoS specification must consider all aspects of presentation, hardware and software components in the system. The various aspects of QoS are usually grouped under some well-defined categories. An example of grouping QoS parameters is the the five categories given by Vogel [1995] and listed in Table 1.

The SAS factors described earlier are also called QoS parameters by some authors [Furht, 1995, p.52]. In fact, the SAS factors are a set of quantitative parameters that determine the QoS. The QoS is a wider concept that includes qualitative, as well as quantitative aspects of networked multimedia systems [Sharda, 1999, p. 218].

Table 1: Five categories of QoS parameters

Category	Example Parameters
Performance-oriented:	End-to-end delay and bit rate.
Format-oriented:	Video resolution, frame rate, storage format, and compression scheme.
Synchronization-oriented:	Skew between the beginning of audio and video sequences.
Cost-oriented:	Connection and data transmission charges and copyright fees.
User-oriented:	Subjective image and sound quality.

IV. MULTIMEDIA TRANSMISSION REQUIREMENTS

The two most demanding types of multimedia object streams are audio and video. This section explores the requirements for carrying audio and video traffic over networks. The requirements for transmitting digital audio and video over networks depend upon the media levity; i.e. whether the transmission is captured in real-time or retrieved from a storage device. These requirements depend even more upon the type of the receiver. If the receiver is a human being then the requirements focus on making a good quality presentation, as perceived by this observer. If the receiver is a computer then the requirements focus on ensuring that the information is received and stored without any errors.

The qualitative and quantitative requirements for audio and video transmission depend upon the ability of the human senses. The auditory senses and vision work differently. Human auditory senses are more sensitive to variations, while human vision smoothes out variations. The transmission requirements are based on the following factors:

- **Response of the Human Ear:** 20 Hz - 20 KHz (at best), and sensitive to change in signal levels, rather than absolute value of the signal.
- **Response of the Human Eye:** Smoothes out variations in the images projected on the eye.

- **Tolerance to Errors:** Higher error rate can be tolerated in uncompressed signals.
- **Tolerance to Delay and Jitter:** Delay and jitter must be small for live applications.
- **Lip Synchronization:** Time gap between sound and its lip movement is the most critical aspects of video presentations.

Audio Bandwidth: The bandwidth of a digital transmission system refers to its data transmission capacity in bits per second (bps). The bandwidth required for various types of audio signals are given in Table 2.

Table 2: Bandwidth Requirements –Audio

Audio Quality	No.of Channels	Sampling Frequency	Amplitude Resolution	Uncompressed Bandwidth	Compressed Bandwidth
Phone	1	8 KHz	7-bit	56 Kbps	
Phone	1	8 KHz	8-bit	64 Kbps	4-32 Kbps
CD	2	44.1 KHz	16-bit	1.411 Mbps	64-192 Kbps

SAS Factors for Audio: The SAS factor values required for audio are listed below.

- **Delay:** 100 ms to 500 ms in an application involving conversation between two people.
- **Jitter:** If delay is less than 100 ms then the delay jitter must be less than 10 ms.
- **Lip synchronization:** Lip movement and the related audio should be presented within 80 ms of each other.
- **Error rate:** BER should be less than 0.01 for telephone quality audio, and less than 0.001 for uncompressed CD quality audio.

Video Bandwidth: The bandwidth required for video is much higher than the bandwidth required for audio [Elliot, 1993]. The bandwidth required for various types of video signals are given in Table 3.

Table 3: Bandwidth Requirements –Video

Video Quality	Horizontal Resolution	Vertical resolution	Color resolution	Frame Rate	Uncompressed Bandwidth	Compressed Bandwidth	Compression Standard
Units ->	*Lines*	*Lines*	*No.of colors*	*/second*	*Mbps*	*Mbps*	
Television							
HDTV	1920	1080	24	60	2000	25-34	MPEG-2
TV	720	576	24	25	150	3-6	MPEG-2
VCR	640	480	24	25	92	1.5	MPEG-2
Video Conference							
CIF Format	352	288	8	15	NA	0.112	px64 or MPEG-4
QCIF Format	176	144	8	5-10	NA	0.0048-.064	px64 or MPEG-4

SAS Factors for Video: The SAS factors applicable to video are similar to those for audio transmission. The delay and delay jitter values are dictated by the audio component of the presentation. In a good quality video presentation the need for lip synchronization demands that the delay and jitter values used for audio be applied to the video stream as well. The acceptable channel error rate depends upon the video quality.

V. MULTIMEDIA TRANSMISSION OVER WANS

A wide area network (WAN) is a network that can span distances from a few miles to thousands of miles. Traditional WAN technologies include:

- Circuit switched services
- Integrated Services Digital Network (ISDN)
- Packet Switched and residential services

In recent years, the Internet and other new services have been added. Each of these WAN services is discussed in this section.

CIRCUIT SWITCHED SERVICES

Circuit switched services use networks that are based upon hardware switches. The entire network consists of a mesh connection of a number of switching nodes. The telephone network is the oldest and one of the most widespread circuit switched network. Other switched circuit networks have also been developed. The bandwidths of the various circuit switched services are listed in Table 4.

POTS: The data transmission rate that can be obtained over a plain old telephone service (POTS) connection is always less than 64 Kbps. This rates is adequate for text and audio transmission, but falls short of the bandwidth required for good quality video transmission in real-time. Video telephones that work over POTS connections use low frame rate and small picture size, in conjunction with image compression.

Table 4: Bandwidths of Circuit Switched WAN Technologies

Type of Service	Bandwidth	Comments
Plain Old Telephone Service		**Limited by 3-4Khz bandwidth**
Using V.32 model	4.8 Kbps	Low cost
Using V.34 modem	28.8 Kbps	Widely available Uses TCM modulation
Using V.42 bis modem	38.4 Kbps	Not very reliable at higher data rates
Switched 56	**56 Kbps**	**Low resolution video conference**
Analog Leased Lines		**Low Delay, jitter, and error rate**
ISDN		**Integrated Services Digital Network**
Basic-rate interface	144-192 Kbps	Digital voice and video conference
Primary-rate interface	1.54 – 2 Mbps	Compressed VCR quality video
Digital Leased Lines		**Good for transmitting following video**
Fractional T-1 to T-4	384 Kbps to 274 Mbps	Video conference, VCR/Broadcast/HDTV
SONET		**Synchronous Optical Network**
OC-1 to OC-48	51.84 Mbps to 2488 Mbps	Available in multiples of 51.84 Mbps

Switched 56 Service: The switched 56 service provides a bandwidth of 56 Kbps over switched lines [Szuprowicz, 1995, p. 192]. This service can be used for low-resolution video conferencing as an alternative to the POTS network. It can be used for multi-point conferencing as well. A somewhat similar

service can be obtained by using 56 Kbps modems (using the ITU V.90 standard) over POTS lines.

Analog Leased Line: A leased line does not require any dial-up procedure, provides better quality connection, and higher signal to noise ratio (SNR) leading to higher data transmission rates as compared to those on dial-up lines.

Digital Leased Line: Time Division Multiplexed lines have been used as the inter-exchange trunks on the telephone networks since the 1950s. These lines have become available for leasing since the early 1980s [Szuprowicz, 1995, p. 197]. These trunk lines are called T-1, T-2, T-3 and T-4 in the U.S., Japan and Korea. A fractional T-1 (FT-1) line is also available at 384 Kbps. This service is also called the switched 384 service. The bandwidth and the multimedia applications of various digital leased lines are listed in Table 5.

Table 5: Digital Lease Lines and their Multimedia Applications

Line Type	**Speed (Mbps)**	**Multimedia applications, comments**
FT-1 (Fractional T-1)	0.384	Video conferencing. Also called switched 384.
T-1	1.544	VCR quality compressed video.
T-2	6.312	Broadcast TV quality compressed video.
T-3	44.70	HDTV-quality video transmission
T-4	274.0	Multiple video channels

QoS and Cost of Digital Leased Lines: Leased lines are technically excellent for providing the required QoS for the transmission of multimedia information. Because of the end-to-end switched connection, the SAS factors can be controlled very well. But the cost of leased lines can make them uneconomical for many applications. To mitigate the high cost associated with digital leased lines, the ISDN concept was developed to provide end-to-end digital transmission over dial-up connections, while providing the required QoS [ITU-T,1992].

INTEGRATED SERVICES DIGITAL NETWORK (ISDN)

The Integrated Services Digital Network (ISDN) was designed in the 1980s to provide data rates in the range of Kbps to Mbps over switched connections. To provide even higher data rates the original ISDN was extended to Broadband-ISDN (B-ISDN). The ISDN services are provided to the user as ISDN interfaces, where each interface comprises a number of ISDN channels.

ISDN Channels and Interfaces: ISDN channels, their bandwidths, and application areas are listed in Table 6 [Beyda, 1996, p. 211]. These channels are combined to provide standard interfaces called: Basic Rate Interface (BRI), Primary Rate Interface (PRI), and Hybrid Interface. The various ISDN interfaces are listed in Table 7.

Table 6. ISDN Channels

Channel Designation	Channel Type	Bandwidth	Application area
A	Analog	3- 4 KHz	Analog voice
B	Digital	64 Kbps	Digitized voice or data
C	Digital	< 16 Kbps	Low speed data
D	Digital	16 or 64 Kbps	Signaling or data

Table 7. ISDN Interfaces

Interface Name	Channels	Combined Bandwidth	Application area
Basic-rate interface	2B+D	144-192 Kbps	Digitized voice and data
Primary-rate interface	23B + D or 30B + D	1.544 Mbps or 2.048 Mbps	Multimedia including video. LAN to LAN connection.
Hybrid interface	A + C	Analog voice + 16 Kbps data	Hybrid connection for transition period.

Basic Rate Interface - BRI: The BRI interface is aimed at providing a simple interface to the desktop that includes a phone connection and a digital interface for the desktop computer. The D-channel is used for signaling; and the two B-channels provide a bandwidth of 128 Kbps for data transmission. The bandwidth of the BRI interface falls short for any serious multimedia application; it is barely enough for low-end video conferencing.

Primary Rate Interface - PRI: In the PRI interface 23 or 30 B-channels are combined. The D-channel in the PRI is used for out-of-band signaling. The 1.544 to 2.048 Mbps bandwidth provided by PRI is adequate for video transmission at VCR quality, apart from other multimedia applications.

Hybrid Interface: The Hybrid Interface allows connections that use a hybrid of analog and digital communication. The hybrid interface has been included in the ISDN systems to provide a transition path from the old POTS service to a full-fledged digital service.

PACKET SWITCHED AND RESIDENTIAL SERVICES

The transmission rates over the various packet switched and residential services are listed in Table 8. The ability of these services to carry multimedia traffic is discussed in the following subsections.

Table 8: Bandwidths of Packet Switching Networks and Residential Technologies

Type of Service	Bandwidth	Properties and Application
X.25	< 2 - 8 Mbps	Not suitable for audio and video traffic.
Frame Relay	56 Kbps to 1.544 Mbps	Better suited for multimedia than X.25.
SMDS	1.544 - 46 Mbps	Can carry multimedia traffic.
Internet	Depends upon implementation	TCP/IP not suitable for multimedia.
ADSL	1.544 Mbps to 6.1 Mbps	VoD and Internet access at home.

X.25 Service: This is one of the older packet switching services. The X.25 protocol includes error detection and correction over every hop. This slows the transmission process considerably, leading to delays that are not acceptable for real-time traffic. The bandwidth available on an X.25 connection is generally limited to 2 Mbps, though some 8 Mbps connections are also available. Bandwidth reservation is not readily available, nor is the facility of multicasting. Most X.25 implementations cannot provide bit rate guarantees. Thus, multimedia applications such as video conferencing cannot be run over X.25 networks.

Frame Relay: In a frame relay service, the packets are sent from the source node to the destination node without any node-to-node error and flow control. Packet acknowledgment is sent only from the destination node, and is relayed back without any processing at the intermediate nodes. This approach reduces protocol overhead, making it possible to transmit some types of multimedia traffic over frame relay service.

Switched Multimegabit Data Service (SMDS): SMDS is a connection-less service definition, designed primarily for interconnecting high speed LANs, over existing high-speed communication links. The delay and jitter depend upon the underlying technology; thus, there are no QoS guarantees. The ability of SMDS to carry real-time video or audio transmission is a function of the carrier technology used for providing the service.

Asymmetric Digital Subscriber Loop (ADSL): ADSL technology was developed to deliver multimedia information to the home. Over good quality local loops (shorter than 5.5 Km), ADSL technology can carry 1.544 Mbps, the data rate of a T-1 line. If the length of the local loop is restricted to less than about 3 Km, then the data rate can be increased to 6.1 Mbps. VoD is one of the main applications of the ADSL technology. Access to the Internet, and other computer services are also promising application of ADSL.

Synchronous Optical Network (SONET): The media used in SONET is optical fiber. It provides much higher bandwidths than the copper-based circuits. SONET is a transport service that can be used for advanced network services such as Broadband-ISDN (B-ISDN), Fiber Distributed Data Interface (FDDI), and High Definition Television (HDTV) transmission. SONET is used in conjunction with the Asynchronous Transmission Mode (ATM) protocols for the B-ISDN systems. These broadband services are discussed further in Sections VI and VII.

THE INTERNET

How well does the Internet handle multimedia information? This question can be answered by studying the communication protocols used over the Internet. The rapid growth in the number of Internet users in the recent years has put too much demand on its bandwidth, making it difficult to transmit real-time multimedia information successfully. A new initiative called the Internet2 (see below) was initiated in 1996-97 to overcome these difficulties.

TCP/IP: The communications protocols used over the Internet are called the TCP/IP suite of protocols. The two main protocols in the TCP/IP suite are the Transmission Control Protocol (TCP), and the Internet Protocol (IP). The TCP/IP protocol stack consists of five layers [Stalling, 1997, p. 522]:

1. Application Layer,
2. Transport Layer,
3. Internet Layer,
4. Network Access Layer, and
5. Physical Layer.

Application layer: The application layer is used for interfacing the user applications to the communications environment. Some of the most commonly used TCP/IP application layer protocols include:

- Simple Mail Transfer Protocol (SMTP),
- Multipurpose Internet Mail Extension (MIME),
- File Transfer Protocol (FTP),
- TELNET protocol for remote login,
- Hyper-Text Transfer Protocol (HTTP), and
- Simple Network Management Protocol (SNMP).

Transport layer: The transport layer is also called the host-to-host layer. It is responsible for the end-to-end data transfer. Two protocols are used at this layer: TCP and UDP.

TCP: The Transmission Control Protocol (TCP) is the main protocol used at this layer. It is a connection oriented transport protocol, and includes mechanisms for establishing a connection between the two end-systems, as well as for data transmission. The TCP protocol includes error control and flow control procedures. Thus the data received by the application layer is error free and the packets arrive in the correct order.

UDP: The User Datagram Protocol (UDP) is a connection-less transport protocol, and thus the packet can arrive out-of-order. It includes options for multicasting as well broadcasting.

Internet layer: The Internet layer protocol performs the function of routing data over the internet, and is called the Internet Protocol (IP). IP protocol (version 4) is a 'best-effort' protocol, because it does not guarantee any average or minimum bit rate [Fluckiger, 1996, p. 583]. Various IP implementations suitable for different network access mechanisms, and physical layers are available. Static as well as dynamic IP routing protocols exist. Systems called IP routers are used to handle the routing function.

Network Access layer: The Network Access Layer is used for routing data between two end-systems attached to the same network. If a data packet has to go from one network to another, it must make use of the Internet layer. Different implementations of the Network access layer exist for different types of networks.

Physical layer: The physical layer is the layer responsible for the transmission of data bits. It includes options for conducted, radiated and optical media. The physical layer standards are derived form existing network implementations, and their physical layer standards.

Multimedia Capability of the Internet: The Internet, like many other older technologies, was not designed to carry multimedia traffic. The IP version 4 (IPv4) protocol, being a 'best-effort' protocol, does not guarantee service levels for bandwidth and SAS parameters such as delay, and delay jitter. The TCP protocol builds a reliable transmission service over the IP protocol by including error control and flow control. These aspects make the protocol 'heavy' and fundamentally unsuitable for multimedia traffic. New Internet protocols, such as the Real-time Transport Protocol (RTP) and extensions to the IP protocol, such as ST-II and RSVP (see below) can support real-time multimedia traffic.

MULTIMEDIA OVER INTERNET

The capability of the Internet to carry real-time multimedia traffic depends upon many factors and must be evaluated for specific sites and applications. In the past the Internet could not be used for real-time multimedia traffic. Enhancements to the existing network infrastructure, and development of new protocols are making it possible to use the Internet for multimedia information networking.

IPv6: IP version 6 protocol (IPv6), also called IP Next Generation (IPng) protocol is an enhancement to the IPv4 protocol. It supports efficient multicast transmission, dynamic and distributed group membership, multiple group membership, and multiple send/receive modes [Fluckiger, 1995, p. 435].

Multicast Backbone (MBone): The MBone (Virtual Internet Backbone for Multicast IP, or Multicast Backbone) is a real-world implementation of the IP multicast protocol. The MBone system started in 1992 and has continued to grow. MBone can be used for group communication applications such as video conferencing and collaborative conferencing.

Stream Protocol (ST-II): Stream Protocol ST-II supports transmission of streams of packets delivered to either a single destination, or to multiple destinations with controlled delay characteristics. Every node through which ST traffic passes maintains state information about the stream. ST-II includes facility for resource reservation to support real-time traffic. Transport protocols that can use the ST protocol include TCP, packet video and voice protocols.

Real-time Transport Protocol (RTP): RTP is an Application-layer protocol, developed for supporting real-time applications. RTP provides end-to-end transmission service to real-time data such as, audio stream, video stream, and simulation data, over unicast or multicast network-layer services. It does not provide QoS guarantees or facilities for resource reservation. RTP often runs over the IP and the UDP protocols.

Resource ReSerVation Protocol - RSVP: The main aim of the resource ReSerVation Protocol (RSVP) is to provide QoS guarantees for multimedia traffic over the Internet. It is aimed primarily at multicast communications; though it can work with unicast communication as well. RSVP can run over IPv4 and IPv6. RSVP allows dynamic group memberships, and can adapt to routing changes.

THE INTERNET2 INITIATIVE

The Internet2 is an initiative designed to overcome the fundamental problems of throughput, delay and delay jitter encountered on the original Internet. It was announced in October 1996. By November 1997 it had grown from 34 universities to over 100 academic and commercial participants in the United States [Internet2]. Internet2 is a well-managed, Giga-bit network, serving the academic and the research communities. Education-on-demand, video conferencing, and virtual classrooms will be some of its main applications.

VI. MULTIMEDIA TRANSMISSION OVER LANS

A network restricted to a single site is called a local area network (LAN). The original LAN technologies, when developed in 70s and 80s were not designed to carry multimedia traffic. Some of the current enhancements to these LAN technologies enable them to transmit real-time multimedia traffic. Speed, throughput, and the ability of various LAN technologies to carry multimedia traffic are listed in Table 9.

ETHERNET LANS

Ethernet is one of the most widely used networking technologies. Although the raw bit rate specified for Ethernet LANs is 10 or 100 Mbps, the throughput is well below these bit rate values. Many efforts, such as Switched Ethernet, IsoENET, Fast Ethernet, and 100VGAnyLAN, make Ethernet systems capable of carrying multimedia traffic to some extent.

Gigabit Ethernet is a newer technology, currently under development. When this technology becomes affordable for widespread deployment, it will make Ethernet networks highly suitable for carrying multimedia traffic.

Table 9: Multimedia Capability of LAN Technologies

LAN TYPE	SPEED	THROUGHPUT	MULTIMEDIA CAPABILITY
ETHERNET			**Random delay**
Ethernet	10 Mbps	3 - 9 Mbps	Not suitable for multimedia.
Switched Ethernet	10 Mbps	9 Mbps	Can be used for stored multimedia.
Isochronous Ethernet	6.144 Mbps	Isochronous.	Suitable for videoconference.
Fast Ethernet	100 Mbps	40-90 Mbps	Can be used for stored and live multimedia.
100VGAnyLAN	100 Mbps		Suitable for interactive multimedia and video conferencing.
TOKEN RING			**Deterministic delay**
Token Ring at	4 Mbps	3.8 Mbps	Inadequate throughput per station.
Token Ring at	16 Mbps	15.5 Mbps	Can carry audio and video traffic.
FDDI			**Guaranteed delay & throughput**
FDDI / CDDI	100 Mbps	50-60 Mbps	Can carry multimedia traffic.
Synchronous FDDI	100Mbps	As allocated	Can carry multimedia traffic.
FDDI-II			6.144 Mbps wideband channels can carry multimedia traffic.
ATM	**34-155 Mbps**		**Designed to carry multimedia.**

Standard Ethernet: Ethernet LANs are based on bus topology and the CSMA/CD (Carrier-sense multiple access with collision detection) protocol. Even though the raw speed of the physical layer components used in the standard Ethernet technology is 10 Mbps, the network throughput can drop down to around 3.6 Mbps. If 100 stations are connected to the LAN, then the nominal throughput of each station will be only 3600/100 = 36 Kbps. Moreover, because of the occurrence of collisions in the Ethernet system, the delays encountered by the individual packets can vary beyond the jitter limits specified for acceptable audio and video quality. Thus a network based on the standard Ethernet technology is not suitable for multimedia traffic.

Switched Ethernet: Switched Ethernet uses a star topology. Each of the workstations, server(s), printer(s), and other devices are connected to a central switching hub via point-to-point links, using the Ethernet physical layer standard. 9 Mbps throughput is theoretically possible for each link. The overall throughput will depend upon switching speed and architecture of the hub. But without explicit support for isochronous traffic, a switched Ethernet system cannot guarantee QoS for live audio and video transmissions.

Isochronous Ethernet: Isochronous Ethernet (IsoENET) can transmit isochronous traffic over switched Ethernet LANs. It adds an isochronous channel of 6.144 Mbps capacity to the standard 10 Mbps Ethernet capacity. This capacity can be broken into 96 sub-channels of 64 Kbps each. These channels work as circuit switched channels with low transmission delay. IsoENET can support multimedia applications such as video conferencing. The original 10 Mbps Ethernet channel cannot be used for multimedia traffic even if it is idle.

Fast Ethernet: Fast Ethernet is a set of physical layer standards designed for a 100 Mbps transmission rate. Additional options in the fast Ethernet range are called: 100BaseTX, 100BaseFX, and 100BaseT4. The 100BaseTX and 100BaseFX networks use separate links for data transmission and reception. 100BaseTX works over shielded twisted pair (STP) copper wires, or category-5 unshielded twisted pair (UTP-5) wires. A 100BaseFX network works with a pair of optical fibers. The 100BaseT4 specification allows the use of (four) voice grade UTP-3 cables. Even though the bit rate is 100 Mbps on a fast Ethernet, due to collisions on a network. With as many as 100 stations the effective throughput drops to about 40 Mbps. This throughput is adequate for transmitting non-live multimedia information, such as VoD. Because of the delay variance experienced by the CSMA/CD protocol even the Fast Ethernet system is not suitable for real-time multimedia traffic.

Switched Fast Ethernet: If collisions are avoided by using a 100 Mbps switch as the hub of a switched Ethernet system then the network can be used to carry real-time multimedia traffic. A switched Ethernet system using a hybrid of 100 Mbps and 10 Mbps Ethernet links is a cost-effective solution for building a multimedia LAN. End systems requiring high throughput are connected to the hub via 100 Mbps links, while most workstations are connected to the hub via less expensive 10 Mbps links.

100VG-AnyLAN: A standard called 100VG-AnyLAN is designed to support 100 Mbps transmission over voice grade lines, and multiple MAC (media access control) layer frame types. It uses a hierarchical Star topology. The media access protocol for the 100VGAnyLAN uses a round-robin scheme. Two priority levels are available, normal and high. Any station wishing to transmit data sends a request to the hub. Only on receiving permission from the hub does the station transmit data. Real-time traffic can be transmitted with minimum delay, by using its demand priority scheme. Interactive multimedia and video conferencing traffic can be supported by the 100VGAnyLAN technology.

RING NETWORKS

Ring topology networks use a deterministic media access scheme called token passing. The token-based media access schemes include priority and reservation mechanisms. These priority mechanisms make it possible to allocate high priority to real-time multimedia traffic.

Token Ring: The original ring technology is based on the IEEE 802.5 token passing MAC protocol. It can be operated at either 4 Mbps or at 16 Mbps. The 4 Mbps token ring network cannot support live video transmission, even for the lowest quality video. The 16 Mbps operating speed helps by increasing the throughput allocated to individual stations. The priority and reservation features of the 802.5 protocol facilitate successful transmission of isochronous traffic. Up to ten simultaneous sessions of VCR quality audio and video have been demonstrated on the 16 Mbps Priority Token Ring [Fluckiger, 1995 p. 413] by reserving 80% of the bandwidth for the real-time multimedia traffic.

FDDI and CDDI: The Fiber Distributed Data Interface (FDDI) standard uses optical fibers and a dual ring topology to form a local area network or a metropolitan area network (MAN). The current version of the FDDI standard can also be used over twisted pair copper wires, giving the Copper Distributed Data Interface (CDDI) standard. The FDDI communication protocol uses a capacity allocation scheme that breaks traffic into synchronous and asynchronous components. A finite transmission capacity is allocated to each station. The traffic that makes use of this reserved capacity is called synchronous traffic. The remaining capacity is used for asynchronous traffic. By keeping the volume of asynchronous traffic low, the synchronous traffic allocation can be used for transmitting multimedia information. FDDI provides a deterministic guarantee for delay and a statistical guarantee for bandwidth. Thus, FDDI is suitable for multimedia information networking.

FDDI-II: This enhancement to the fiber based networking technology aims to carry isochronous traffic with guarantee of delay and delay jitter values. Thus, it is also called isochronous FDDI. The 100 Mbps bandwidth is broken into 16 channels, called wideband channels (WBC), each capable of carrying traffic at 6.144 Mbps. These channels behave like circuit switched channels, and can be divided into sub-channels of 8 Kbps each, or multiples of this value. These channels are suitable for carrying traffic at a constant bit rate (CBR). FDDI-II is capable of carrying broadcast TV quality video traffic.

VII.ASYNCHRONOUS TRANSFER MODE (ATM)

The Asynchronous Transfer Mode (ATM) technology was conceived and designed as a service that can support multimedia traffic [Goralski, 1994]. ATM is based on the principle of cell switching. Data is transmitted as small (53 byte), fixed size cells. Switching of small cells in hardware allows the traffic to be transmitted with low delay and delay jitter. ATM technology allows bandwidths generally ranging from 34 to 155 Mbps and going up to 622 Mbps. The idea of developing large-scale networks based on the Asynchronous Transfer Mode (ATM) emerged in the middle of 1980s. ATM was chosen as the technique that could be used to implement networks for all types of applications: applications requiring low bandwidth, applications requiring high bandwidth, applications requiring low delay and jitter, as well as applications with no restrictions on delay and jitter [Stalling, 1997].

The ATM protocol has been designed to meet the different QoS needs of different applications. Specific procedures exist for the user to specify the required QoS [Jung, 1996a]. The QoS specified by the user is translated into values required of the network performance parameters [Jung, 1996b]. The network provides deterministic guarantees for some of the parameters, while for some other parameters the network gives only statistical guarantees.

One of the main benefits of the ATM technology is that it can be used for WANs as well as for LANs [Szuprowicz, 1995]. ATM is used in LANs in two different modes: ATM switch-based LANs, and LAN emulation.

Switch-Based ATM LAN: To create an ATM switch based LAN, all nodes are connected directly to the ATM switches, and the individual stations transmit and receive data under the ATM protocol. Therefore, sessions involving multimedia traffic can be established between any two stations.

In another LAN scenario, ATM switches are used to form a backbone for a multi-segment network. The individual LAN segments connected to the ATM backbone can be Ethernet, Token Ring, or others. These segments connect to the ATM switches through bridges. The ability to transmit multimedia information between two stations now depends upon the capability of the LANs to which each of the segments is connected.

LAN Emulation: ATM can also be used for building LANs in a mode called LAN emulation (LANE). In this mode, the ATM network transports the conventional LAN frames. LAN emulation for various conventional LAN technologies, such as Ethernet and Token Ring, are available. LAN emulation provides a migration path from the current LAN technology to fully ATM based networks. The multimedia capability of a LAN emulation system depends upon the LAN being emulated.

VIII. FUTURE DIRECTIONS

Future efforts will focus on improving the existing networking technologies to provide better multimedia transmission services at lower cost. These efforts will be based on a three-prong approach. The first prong will focus on enhancing the network infrastructure. The second prong will tackle networking protocols, and the third prong will aim to reduce the traffic, by using better compression techniques.

Bandwidth of the current networking technologies range from Kbps to Mbps. If the bandwidth of a network is enhanced to Giga bits per second (Gbps) range, then this network can carry multiple audio and live video streams. Of course, the networking service must also provide low delay and delay variance required for good quality multimedia presentations. A number of enhancements to existing technologies were described in earlier sections. Gigabit Ethernet is a technology that is being developed to take Ethernet systems into the multimedia networking era.

A vast majority of networked computers use the Ethernet technology for their LAN connections. Towards the end of 1996, Ethernet had 83 percent of the market [Gigabit, 1997]. The Fast Ethernet technologies have already provided an upgrade path to 100 Mbps range. The aim of the Gigabit Ethernet technologies is to provide 1000 Mbps bandwidth channels. The ATM technology already provides bandwidth ranging from Mbps to Gbps, then why develop Gigabit Ethernet systems? Because, when two Ethernet LANs are connected via an ATM backbone, the Ethernet packets must be converted into ATM cells on the sending end, and the ATM cells must be converted back to Ethernet packets on the receiving end. The main advantage of the Gigabit Ethernet technology is that it will use the same packet format as used by the old Ethernet and the newer Fast Ethernet systems.

Gigabit Ethernet standards have been developed by the Institute of Electrical and Electronic Engineers (IEEE) standards committee IEEE-802.3z. The main features of the new Gigabit Ethernet standard include.

- Half-duplex and full-duplex operations.
- 802.3 Ethernet standard frame format.
- CSMA/CD media access protocol, within each domain.
- Backward compatibility with 10BASE-T and 100BASE-T systems.

Another area where the Gigabit Ethernet technology will be compatible with the existing Ethernet systems is Network Management. The existing management objects will also be used in the Gigabit Ethernet systems. This will make it much easier for the network managers to adopt the Gigabit Ethernet technology. For the next few years it will be worthwhile keeping an eye on the development and deployment of the Gigabit Ethernet technology.

The CSMA/CD protocol used in the Ethernet systems is fundamentally ill suited to providing guaranteed delay and delay jitter values. But with sufficient bandwidth devoted to each session, the QoS could be guaranteed by using higher layer protocols such as RSVP. Development and widespread deployment of application layer communication protocols that guarantee QoS will be another important direction in the future.

The third important direction will be the development and deployment of better compression techniques. The two most widely used data compression standards are JPEG (Joint Photographic Experts Group) and MPEG (Motion Picture Experts Group) standards. The compression algorithms used in the JPEG and the MPEG standards work on 8x8 pixel blocks of the image. Thus any error encountered on the communication channel can lead to a complete 8x8 image block being corrupted. This leads to blocky artifacts introduced in images transmitted over networks. Therefore, an important issue will be the development of image compression and communication protocols hand-in-hand, so that the artifacts introduced in the image due to communication errors lead to graceful reduction in quality. That is, a few errors should go unnoticed, and as errors increase the image quality should deteriorate on a sliding scale, rather than jump from good quality to bad quality.

IX. CONCLUSIONS

Multimedia traffic requires high throughput, low delay and very low delay variance for real-time delivery. Most legacy networking systems were not designed for multimedia traffic. This paper presented the limitations of the legacy networks, and the fundamentals of advanced networks that can carry real-time multimedia traffic. The various wide area networking options for multimedia networking include the POTS connection, leased lines, ISDN and ATM.

Being able to use the Internet for transmitting real-time multimedia traffic is a very attractive proposition. But, the original Internet, like many other older technologies, was not designed to carry multimedia traffic. Multimedia enabling protocols, such as RTP and RSVP, have been developed to carry multimedia traffic over the Internet. A new effort called Internet2 aims to overcome the problems of throughput and delay by using Gigabit technologies as its backbone.

Enhancements to existing LAN technologies have been developed in an effort to make them capable of carrying multimedia information. Switched Ethernet, Isochronous Ethernet, Fast Ethernet, 100VGAnyLAN, FDDI-II, and Synchronous FDDI are some of these enhancements for LAN systems. Given the large installation base of Ethernet networks, a path for upgrading these into multimedia networks is being developed. This upgrade path - called the Gigabit Ethernet technology - is the technology to watch.

Of the various technologies, ATM is the one that was conceived, developed and designed to carry multimedia traffic. ATM technology can be used for WANs, as well as the backbone for LANs.

Development of new communication protocols in conjunction with data compression techniques will also be important in the future. This will lead to better multimedia communication services. Much research is in progress in the area of networked multimedia systems. It will be five to ten years before multimedia networking technologies become as well established as the technology used for standard phone systems.

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EDITOR'S NOTE: The following reference list contains hyperlinks to World Wide Web pages. Readers who have the ability to access the Web directly from their word processor or are reading the paper on the Web, can gain direct access to these linked references. Readers are warned, however, that

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LIST OF ACRONYMS

ATM	Asynchronous transfer mode
BER	Bits error rate
B-ISDN	Broadband-Integrated services digital network
BPS (bps)	Bits per second
BRI	Basic rate interface
CBR	Constant bit rate
CDDI	Copper distributed data interface
CER	Cell error rate
CLR	Cell loss rate
CSMA/CD	Carrier-sense multiple access with collision detection
FDDI	Fiber distributed data interface
FTP	File transfer protocol
Gbps	Giga bits per second
HDTV	High definition TV
IEEE	Institute of electrical and electronic engineers
IP	Internet protocol
ISDN	Integrated services digital network
ISO	International standards organization
IsoENET	Isochronous Ethernet
ITU	International telecommunications union
Kbps	Kilo bits per second
LAN	Local area network
LANE	LAN emulation
MAC	Medium access control
MAN	Metropolitan area network
Mbps	Mega bits per second
MBone	Multicast backbone
MBR	Mean bit rate
MPEG	Motion picture experts group
NC	Network computer
NPP	Network performance parameter

OSI	Open systems interconnection
PBR	Peak bit rate
PCM	Pulse code modulation
PER	Packet error rate
PLR	Packet loss rate
POTS	Plain old telephone service
PBR	Peak bit rate
PRI	Primary rate interface
PSDN	Packet-switched data network
QoS	Quality of service
RSVP	ReSerVation Protocol
RTP	Real-time Transport Protocol
SAS	Synchronization accuracy specification
SNR	Signal to noise ratio
SONET	Synchronous optical network
ST-II	Stream Protocol-II
STB	Set top box
STP	Shielded twisted pair
TCP	Transport control protocol
TCP/IP	Transmission control protocol / Internet protocol
UDP	User datagram protocol
UTP	Unshielded twisted pair
VBR	Variable bit rate
VC	Virtual circuit
VCR	Video cassette recorder
VoD	Video on demand
WAN	Wide area network
WBS	Wide band channel
WWW	Word Wide Web. Also called the Web