COMMUNICATIONS REQUIREMENTS OF MULTIMEDIA APPLICATIONS: A PRELIMINARY STUDY

Timothy Kwok
Apple Computer, Inc.
20450 Stevens Creek Blvd., MS: 76-2H
Cupertino, CA 95014
Email: kwok@apple.com

ABSTRACT

A preliminary study is presented that provides a framework to explore the communications requirements of wireless applications in general, with emphasis on multimedia applications. An estimate of bandwidth requirements for various basic common applications are given, from which network bandwidth requirements in different scenario can be evaluated. The delay and error characteristics for certain multimedia applications are described, along with possible implications about the suitability of contention-based and reservation-based MAC protocols. This paper provides a basis for further investigating the networking requirements as well as the criteria for designing MAC protocols to support multimedia applications in the wireless environment.

I. INTRODUCTION

Communications networks are designed for connectivity as well as sharing resources. Connectivity refers to providing access to people, information and processing power [1] that are located remotely. Expensive resources that need to be shared include information processing devices (computers, printers, file servers, etc.) and interconnection bandwidth. For wireless communications, sharing of spectrum (bandwidth) is essential because spectrum is not only expensive, but also inherently limited.

After centuries of communications technology developments, the ultimate, utopian goal of telecommunications remains to be setting people free from the constraints of distance, location, time and the medium of communication.

Distance. The constraint of distance refers to the need of communication between individuals at fixed, geographically separated locations. Since the invention of the telephone many decades ago, the worldwide public switched telephone network (PSTN) has evolved to a nearly-ubiquitous presence, primarily through a wireline access infrastructure. The distance constraint is generally removed as far as voice communication as concerned.

Location. However, the wireline access infrastructure ties a person to a telephone at a fixed access point. To make outgoing calls, one has to find a location with a telephone connection. To reach a person, one has to know the location of that person as well as the telephone number at that location, provided that location has a telephone. Only during the last decade has the constraint of location (for voice communications) been addressed effectively, with the development of a wireless "cellular" telephony infrastructure. This allows a person to communicate wherever he is, independent of location (but at a premium price). The personal communications services under development (discussed below) promise to provide such service in a more ubiquitous manner by significantly reducing the cost to an affordable level for the general public and increasing the capacity (number of subscribers) for a given coverage area.

Time. For many years, to communicate with someone over the phone, we are restricted to having both party present at the same time for the conversation. However, during the past decade, the proliferation of answering machines has removed most of us from the constraint of time: the recipient of a voice message does not have to be present at the time of the initial communication.

Medium. The medium refers to the type of information (voice, data, images, video, etc.) that can be communicated. Traditionally, communications are mainly in the form of voice. More recently, data (electronic mail), low quality image (facsimile) and video (video conferencing) have also been introduced. However, some of these different new media of communications have been supported by the PSTN and the cellular network infrastructure in a very inefficient manner with limited capability, because the PSTN design has been optimized for voice communication. Ideally, a communication network should be capable flexible exchange of text, audio, image and video information to support multimedia and collaborative applications easily.

An enormous international effort has recently emerged to create a worldwide information infrastructure referred to as B-ISDN [2], based on a standardized set of networking protocols (yet to be completed). The goal is to flexibly and efficiently support a very wide range of applications with diverse communications characteristics and requirements (such as voice, data, video, image communications). This development is expected to guide the future of communications as well as computing in the 1990s and beyond. This is essential to ubiquitously support important future applications such as multimedia and collaborative applications.

Simultaneously, another development that has received significant attention, both in the U.S. and worldwide, is personal communications service (PCS). Initially, PCS was narrowly described as an evolution from the current analog cellular telephone service (by using microcellular digital technologies and lower power mobile phones) to allow more affordable provision of mobile services to the general public. Refined visions of PCS have been proposed [3] to allow a person to direct calls to either another specified person via a personal number or a specified place via a place number.

While PCS has the emphasis of voice telephony, Data-PCS as proposed by Apple [4] emphasizes the importance of providing data (which includes all information types, e.g., voice is a type of data) connectivity. Functionally and at its mature development phase, Data-PCS can be envisioned as providing wireless capabilities similar to those provided on wires by B-ISDN, i.e., a wireless infrastructure capable of supporting a very wide range of applications. More recently, FCC decided to broadly define PCS to include both voice and data communications and to make available an adequate amount of spectrum for development of innovative and competitive markets [15].

Some parties have suggested that as such, PCS may well share similar technology employed by B-ISDN, for example, ATM (asynchronous transfer mode) as the transport mechanism for all applications. However, there are three fundamental attributes in the wireless environment that are significantly different from the B-ISDN: a finite supply (and thus scarcity) of bandwidth, unreliable transport and interference from both unintended and intended users. ATM technologies have been developed assuming operation in the fiber optics environment, for which bandwidth is abundant, transport is reliable, and interference is virtually nonexistent. To use the ATM protocol in a wireless environment, significant modifications in the ATM protocol may be needed to account for the changed assumptions. However, this topic merits further study, because PCS eventually needs to be communicating with the B-ISDN network and protocol compatibility is essential.

Designing a universal network such as B-ISDN and PCS to support all envisioned applications requires an in-depth understanding of applications in general. The objective of this paper is to suggest a framework to analyze and characterize applications so that networking schemes can be designed to effectively support them, both in the wireline and wireless environments. In particular, discussion on the implications in the wireless environment is emphasized, because it introduces an additional dimension of issues not present in the wireline case. In section II of this paper, we will define what an application is. In section III, the networking assumptions for this paper are specified. Then, section IV will explain how an application in general can be characterized based on its traffic generation characteristics and its corresponding communication requirements. In particular, the bandwidth requirements for a set of basic applications and the implications in supporting these applications in the wireless environment are presented. Finally, section V discusses the implications of supporting multimedia applications on media access control (MAC) protocol design.

II. WHAT IS AN APPLICATION ?

An application is defined as a task that requires communication of one or more information streams between two or more parties that are geographically separated. More specifically, an application can be characterized by the following attributes:

(1) INFORMATION TYPES

In general, the information to be communicated can be classified as time-based or nontime-based. Time-based information is defined as those that must be presented at specific instants to convey its meaning, i.e., timing (synchronization between various parts of the information) is an integral part of the information. Typical time-based information are video and audio, while nontime-based information includes still images, graphics and text. Also, an application can in general include both time-based and nontime-based information. When an application involves multiple information sources (possibly of various information types), synchronization among them is an important issue [5].

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(2) DELIVERY REQUIREMENTS

Applications can also be classified according to its information delivery requirements into real-time or nonreal-time applications. A real-time application is defined as one that requires information delivery for immediate consumption. In contrast, for nonreal-time applications, their information are to be stored at the receiving parties for later consumption. The former requires sufficient bandwidth, while the latter requires sufficient storage. For example, a telephone conversation is considered a real-time application, while sending an electronic mail is a nonreal-time application. In other words, communicating parties for a real-time application participate at the same time, while, for a nonreal-time application, they participate at different times.

However, it is important to distinguish between the delivery requirement (real-time or nonreal-time) of an application from the intrinsic time dependency of its information content, which can be time-based or nontime-based. Video conferencing and image browsing are typical examples of real-time applications, while downloading digitized movies and electronic mail belong to nonreal-time applications. In the case of applications such as image browsing, even though the image is itself nontime-based information, to ensure interactive response for the user, a maximum response time constraint is required to satisfy this application; so it is considered as a real-time application. On the other hand, for nonreal-time applications like downloading a digitized movie, even though the information content is a time-based, the entire movie can be treated as a single file transfer like electronic mail, because the movie is not being displayed in real-time at the receiver. In general, the communications requirements for supporting an application depends on both the information types as well as the delivery requirement of the application.

(3) SYMMETRY OF CONNECTION

In general, a communication application involves a two-way information transfer between communicating parties. Such bidirectional connections can be classified as either symmetric or asymmetric connections. A symmetric connection is one that involves transfer of information of similar traffic characteristics in both directions, otherwise, it is called an asymmetric connection. A voice call is an example of a symmetric connection, while video browsing is considered as an asymmetric connection, which involves sending control messages in one direction and video transfer in the other direction. Obviously, the communications requirements of an application depend on the symmetry of its connections; many networks are designed to take advantage of the (a)symmetry of the applications they intend to support. For example, the cable TV network is designed as a one-way broadcast type network without any switches.

(4) POINT-TO-POINT VS MULTIPOINT

Applications can be considered as either point-to-point or multipoint depending on the number of parties involved in the communication; those involving two parties are called point-to-point applications, while those involving more than two parties are called multipoint applications. Traditional phone conversations are point-to-point applications, whereas teleconferencing, which involves three or more parties, are multipoint applications. Broadcast is the extreme case of multipoint application that involves sending information to all parties of the network. As a result, PSTN has been optimized for point-to-point (and some multipoint) connections by using a switch-based network architecture, while the cable TV network is based on a broadcast nonswitch-based architecture.

(5) HUMAN VS COMPUTING DEVICE

In general, the parties involved in the application may be either a human user (through a user terminal) or a information processing device. For instance, a voice call is a user-user application, while a user accessing a remote database is a user-device application, and finally two supercomputing performing parallel computation to solve the same problem by communicating intermediate results is considered as a device-device application. Collaboration [6] is an application between human users that shares an electronic space for communication. Collaboration can occur between two individuals present at the same (real-time) or different (nonreal-time) times. Real-time collaboration is a user-user application, while nonreal-time collaboration is a user-device applications, because the latter involves a user leaving a message at the other user's communication device.

(6) ACCESS NETWORK

A communication network can designed with both wireline and wireless access. Traditional access to PSTN has been through a wireline infrastructure. The need for wireless access stems from the need of communication independent of the location of the communicating parties; the parties are not limited to the fixed locations where the wireline access network terminate. An example of a wireless access network is a cellular telephone network.

It is important to distinguish two types of wireless access networks: with or without base stations. The main functions of a base station include (local) bandwidth management, routing between communicating devices and relaying to other base stations if necessary. Without a base station, an ad hoc network [4] can be set up among wireless devices, in which these devices communicate directly with each other. This implies the devices have to resolve the contention for the same spectrum themselves. Ad hoc networking is very convenient for users in close proximity (such as in the same meeting room), because they can start communication with each other anywhere without restricting to places that have a base station. However, it is still limited to local communication; because, without a base station, connection cannot be set up with remote devices.

(7) MOBILITY

To reach an individual over a network, one has to first determine the location of that individual within the network, and then more importantly, the network address of the individual's closest network access point (such as his home or work number). This essentially ties the the individual to particular network access points to receive his calls. To reach the branch of a multi-location organization closest to the caller, one has to determine which branch is closest as well as its network address (which differs from those of other branches). Mobility is a means to simplify the connection process with any desired entity. Mobility can be described as the ability to use a single logical label by any user (wherever he is) to access an entity wherever the latter is within the network (without the caller himself determining where that entity is and the corresponding network address). In general, mobility requires an intelligence within the network to determine the location of the destination entity, such as paging for the destination entity before setting up the call.

The mobility concept can be applied to a person, a terminal and a service [7]. Personal mobility allows connection to a person through the network independent of the location of that person. Although personal mobility is usually associated with wireless networks such as the cellular telephone network, personal mobility can also be implemented in wireline network. For instance, the wireline network can route the calls for a person to his office or home depending on the time of day, or to him or his secretary depending on the calling party. Terminal mobility allows the same capability for the physical terminal; in addition, more than one person can be assigned to same the same terminal. Service mobility allows communication to a service, which can be provided at multiple locations, that is closest to the caller. Service mobility exists today in the wireline network, such as the 911 number service, where calls are automatically routed to the emergency service center closest to the calling party. Recently, such service has been extended to pizza delivery service to allow a caller to be automatically connected to the closest pizza outlet by dialing the same number nationwide [16].

III. THE NETWORK ASSUMPTIONS

Before characterizing applications, we need to specify the assumptions on the network supporting the applications. Due to the flexibility in supporting a wide variety of application with diverse requirements [8], packet switching has emerged to be the preferred switching methodology (compared to circuit switching) in designing an universal network. So we assume in this paper the network used for supporting the applications is packet switched based. This implies that the information generated by an application is segmented into blocks, and each appended with a header of control information (such as routing and error control) to form a packet. From the network point of view, an application appears to be a sequence of packets destinated to one or more locations.

Also, the following framework on application characterization is independent of the types of access network, i.e., wireline or wireless. However, the implications for the wireless access environment will be emphasized.

IV. APPLICATION CHARACTERIZATION

An application can be characterized by its **traffic characteristics** and the corresponding **communications requirements**. As mentioned above, an application can be described as a sequence of packets of arbitrary length generated at certain instants of time destined to one or more locations. These collectively specify the traffic characteristics of the application. Each packet of information has its associated set of communication requirements. The traffic characteristics together with the corresponding communication requirements determine the requirements on network resources (bandwidth and buffer) as well as the criteria for network protocol design.

(A) Traffic Characteristics

The traffic characteristics of an application can be specified by its traffic generation process. Since the traffic generation process (or traffic pattern) is basically a sequence of packets generated at arbitrary instants, each packets having an arbitrary length, it can be modelled as an on-off source. The traffic pattern can be characterized by two stochastic processes: (a) packet generation process (or packet arrival process); and (b) packet length distribution process.

Two very important traffic patterns are periodic and bursty traffic patterns. If packet generations occur at regular intervals, it is a periodic traffic pattern. This traffic pattern is important because all real-time applications that carry time-based information have periodic traffic patterns. For example, conventional 64 kbps PCM audio generates samples at 125 µs intervals, each sample consists of 8 bits. For uncompressed full motion NTSC video, video frames (each contains a fixed amount information) are generated at regular intervals 1/30th sec or 30 frames/s. Even for compressed video (such as those generated by the MPEG algorithm [9]), a video frame is still created at regular intervals of 1/30th sec for NTSC and 1/25th sec for PAL formats, except that the amount of information generated is variable at each instant depending on the degree of compression for each frame. (This implies compressed video intrinsically generates variable bit rate traffic, as opposed to the constant bit rate traffic generated by uncompressed video. Incidentally, one of the key reasons packet switching is the preferred switching methodology is that it can support applications with variable bandwidth requirements easily and efficiently.)

Bursty traffic pattern is characterized by packets of arbitrary lengths generated at random time instants, separated by gaps of silence of random duration. Moreover, the period of silence is long compared to the duration of packet generation, leading to the distinctive high peak to average data rate ratio. Conventional data communications are bursty because they are typically file transfer, remote login or more recently, traffic generated by diskless workstations, which are all generated randomly without the tight correlation as in time-based information of real-time applications. In general, bursty traffic is also typical for applications transferring nontime-based information or nonreal-time applications (carrying either time-based or nontime-based information). The unpredictability of bursty traffic, especially the instants at which packet is generated, is the main culprit that needs to be dealt with in designing the packet switched network. Hence, the requirements to support new multimedia applications such as image browsing that have much high burstiness (due to large amount of information transferred per burst) is even more demanding. (However, the statistical gain made possible by multiplexing many bursty traffic sources through the network makes packet switching more efficient than circuit switching, which was the original motivation of packet switching in the 1960's.)

Finally, since multimedia applications have both periodic and bursty types traffic, the need of supporting both classes of traffic with a guaranteed performance in a single integrated network has become the most important challenge.

(B) Communications Requirements

To completely characterize an application, not only does its traffic generation process needs to be specified, but also its corresponding communications requirements. These requirements fall into three categories: bandwidth, delay and error. Among these requirements, the bandwidth requirement is the most important because it immediately determines whether an application can be supported, as well as the quality and quatity of such application that are feasible.

(1) Bandwidth

The bandwidth requirement of an application is determined by one of two methods, depending on its information types and delivery requirements (see Table I).

For a real-time application that generates time-based information, the bandwidth requirement is equal to its natural traffic generation rate . The natural traffic generation rate is simply the amount of information generated by the application per unit time. Such bandwidth requirement may be constant or variable, referred to as constant bit rate or variable bit rate applications, respectively. Constant bit rate applications include traditional PCM coding of voice that generates 64kbps, and common video conferencing systems requiring 2x56 kbps. However, a lot of these traditional constant bit rate applications are intrinsically variable bit rate. For example, voice itself is variable bit rate intrinsically, because voice can be sampled only when someone is talking and nothing needs to be sent otherwise. Similarly, video information needs to be sent only for the changes in image content, thus video compression algorithms intrinsically generates variable bit rate traffic. The reason these applications, which are intrinsically variable bit rate, are encoded as constant bit rate applications is that current networks are primarily circuit switched based, which support only fixed bit rate connections. Thus today, applications are encoded to satisfy the constraint of existing networks, while a more ideal case should be networks designed based on supporting current and future applications with diverse requirements.

For all other applications, namely, either nonreal-time applications, or applications sending nontime-based information, the bandwidth requirement is a function of the response time requirement (the total delay allowed before all the information are transferred) and the amount of information being communicated.

Obviously, to support any application satisfactorily, the bandwidth provided by the network should always exceed the bandwidth requirement of the applications. However,the bandwidth can be insufficient temporarily in the packet switched network, which is possible because the network bandwidth is shared with others using statistical multiplexing (a random process). Hence, network buffering is required to avoid dropping of information, which in turn introduces buffering delay. Moreover, if buffering is insufficient, then it may be necessary to drop information, a process which introduces error. Such delay and error introduced due to insufficient instantaneous bandwidth must be within the delay and error constraint of the applications to maintain the required quality of service. (This is discussed in more detail later.) Therefore, bandwidth must be guaranteed for an application not only to satisfy its bandwidth requirement, but also to limit the delay and error introduced that may exceed its delay and error constraints.

Hence, to maintain the quality of an application in a packet switched network, the network protocol should be designed to first require each new application to negotiate with the network for available bandwidth before a connection be setup by the network (to guarantee that the new application will not need more bandwidth than is available for the existing applications being supported). Second, the protocol can guarantee the bandwidth for the new application once it is accepted by the network. This implies two criteria for the protocol: it is connection-oriented and is reservation-based.

It is also important to note that traditional data networks (such as ethernets, the Internet, token rings), though based on packet switched methodology, have no notion of bandwidth. Applications cannot request a particular bandwidth, because no bandwidth reservation schemes were implemented. Therefore, these networks cannot guarantee any quality of service and are not suitable for supporting many interesting multimedia applications that involves real-time applications carrying time-based information.

Bandwidth requirements in Wireless Networks

In the (single hop) wireless network (with or without a base station), all the users on the network share access to that medium, and each packet sent by any user is automatically broadcast to everybody. This has two implications: an advantage and a disadvantage. First, the bandwidth resources consumed, in multicast or broadcast connection is the same as that for point-to-point connection, because every user automatically receives all packets sent out by any user (and discards those packets not addressed to itself). Second, the bandwidth cost of a two way connection is the sum of the two one-way connections, because each party of the two-way connection needs to consume the bandwidth from the same available medium. Therefore, the total bandwidth requirement on a wireless network depends on, in addition to the delivery requirement and information types of the application, the number of users on the network, the number of parties involved in each application (point-to-point vs multipoint) as well as the connection types of each application (i.e., whether it is symmetric or asymmetric).

To determine the network bandwidth required in each scenario, it is useful to find the bandwidth requirements of a set of basic point-to-point one-way applications that comprise the various information types and delivery requirements. The total network bandwidth requirement is equal to the sum of various combination of these basic application depending on the number of users as well as the different attributes of the applications. Note that, however, the bandwidth requirement derived from response time is a peak bandwidth requirement, and the duration of such requirement is a function of the total amount of information to be transferred. The bandwidth requirements of each of these applications are shown in Table II.

The total bandwidth requirement of each scenario can be determined by summing all the bandwidth required for each connection supported by the network. Table III shows the number of connections of various applications that can be supported by a 10 Mbps medium (assuming 100% network utilization is possible, i.e., these are upper bound estimates). For response-time-driven bandwidth requirements, the table shows the number of transactions (movies, images delivered) that can be supported by a network over a period of time, using one hour for video delivery and one minute for image delivery.

(2) Delay

The issue of network delay in a packet switched network arises from insufficient instantaneous network bandwidth. If bandwidth is insufficient in any parts of the network, buffering is required (unless information is allowed to be dropped), a process which then introduced a random amount of the delay to the information being delivered.

For real-time delivery of time-based information, the delay requirements are absolute delay and delay jitter constraints. The absolute delay is important for real-time communication like video conferencing or conventional telephone conversation because feedback is expected within a certain time period for natural conversation to take place. The delay jitter is the variance of absolute delay incurred from packet to packet for the same information stream. If the bandwidth requirement is always satisfied by the network, then no jitter would occur in the delivery. The delay jitter must be constrained for time-based information because they must be presented at a certain rate for natural consumption by the user. The jitter constraint can be determined by the interval between consecutive samples of the time-based information when it is generated, because a sample is supposed to be displayed at the receiver after each of such interval. For example, 30 frames/sec video can allow a delay jitter of 33 msec, while tradition 8 kHz voice telephony can allow a 125 µsec delay jitter. However, if the absolute delay constraint is large compared to delay jitter constraint, for example, if 10 sec absolute delay can be allowed for video (such as video on demand), video frames can be buffered for a period of 10 sec to provide more flexibility for network delay jitter incurred (jitter allowed is increased to 10 sec from 33msec), assuming there is sufficient buffer at the receiver for removing the jitter.

For nontime-based information or nonreal-time delivery of time-based information, the major delay requirement is the absolute delay, which must be less than the response time required by the application.

Again, in traditional data networks, there is no guarantee in both absolute delay or delay jitter constraints, because the protocol in these networks had not been designed to guarantee delay. Furthermore, data integrity has been of much higher priority than delay requirements in these networks. Hence, in order to ensure no error in data transfer, their protocols have been designed to retransmit data detected with error, thus introducing more delay. Therefore, applications with either delay or response time constraints cannot be supported satisfactorily in these networks.

(3) Error

Packet switching, because of its statistical nature in multiplexing and switching, can introduce a random amount of the delay when the instantaneous bandwidth is not available at parts of the network and the information needs to the stored temporarily in buffers. Two types of error may occur from this process. First, buffer may be insufficient and information needs to be discarded. Retransmission of such information is useful only when application is not a real-time application carrying time-based informations. Second, in the case of real-time application carrying time-based information, the delay introduced by buffering this information may exceed the delay jitter constraint, which will make that piece of information useless even when it finally arrives at the receiver (which makes retransmission futile); this is equivalent to the piece of information being dropped because of buffer overflow, i.e., leading to additional error.

In the wireless environment, an additional source of error arises from the unreliable communication channels, due to noise and unfavorable propagation conditions.

V. WIRELESS PROTOCOL REQUIREMENTS

Many multimedia applications involve real-time delivery of either time-based or nontime-based information. The traffic characteristics of these applications include both periodic and bursty types. To guarantee the quality of service (QOS) of such applications, bandwidth must be sufficient to either support the data rates or meet the response time constraints requirements. In other words, bandwidth needs to be reserved for such applications during the connection setup phase, which also means a connection-oriented protocol is required to manage connection setup and bandwidth consumption of each connection. Contention-based MAC protocol, such as ALOHA and CSMA types, may not of themselves be inherently capable of supporting those particular applications since they do not have built-in mechanisms for bandwidth reservation. (However, conceivably, in the case of the bandwidth requirement of the multimedia application is small compared to the aggregate network capacity, and the network is lightly loaded so that required instantaneous bandwidth is usually available, the QOS may still be acceptable.) On the other hand, reservation based protocols can provide mechanism of bandwidth reservation, which allows gurarantees in QOS for multimedia applications. One purpose of this paper is to facilitate the task of developing enhancements of contention-based protocols to support a wider variety of multimedia applications, as well as to provide a framework for consideration of reservation-based MAC protocols.

VI. CONCLUSION AND FURTHER STUDIES

A framework of applications characterization is presented, with emphasis on multimedia applications. Many of these multimedia applications require guaranteed bandwidth or response time to satisfy their QOS. Contention-based MAC protocols may not be universally applicable for supporting of multimedia applications, but both contention-based and reservation-based MACs are fertile ground for further research. Ultimately, a complete mathematical description of how the traffic characteristics and associated communications requirements (bandwidth, delay and error) translate into network requirements (bandwidth and buffer) is required. This description includes the design of a good traffic descriptor set for negotiation between the terminal and network, as well as the monitoring of traffic once the application is supported. Only those studies that reflect such a more complete mathematical model will provide convincing evidence of the potential success of any specific MAC.

Finally, what have presented is only a framework, and many issues in the above discussion needs to be explored in more detail.

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DELIVERY REQUIREMENTS	INFORMATION TYPES	
	Time-based	**Nontime-based**
Real-time	Traffic generation rate	Response Time & Information Volume
Nonreal-time	Response Time & Information Volume	Response Time & Information Volume

TABLE I Factors determining bandwidth requirements in point-to-point application

TIME-BASED INFORMATION WITH REAL-TIME DELIVERY

Video	Uncompressed	Compressed
videoconference	9/36 Mbps	p x 64 kbps (H.261)[10]
NTSC	~ 200 Mbps	1.5 Mbps (MPEG)[9]
HDTV	~ 1 Gbps	20 Mbps [11,12]
Audio		
voice telephony	64 kbps	16 kbps
CD Quality Stereo	1.4 Mbps (2x706 kbps)	256kbps (MPEG) [13] (2x128kbps)

TIME-BASED INFORMATION WITH NONREAL-TIME DELIVERY

Video (2 hours)	Compressed movie size	Peak Bandwidth	
Delivery Time		15 min [1]	8 hrs [2]
NTSC	1.35 Gbytes	12 Mbps	0.375 Mbps
HDTV	18 Gbytes	160 Mbps	5 Mbps

REAL-TIME AND NONREAL-TIME DELIVERY OF NONTIME-BASED APPLICATIONS

Images	Uncompressed Mbytes	Compressed (JPEG) Mbytes	Peak Bandwidth	
Response time			0.1 sec	10 mins
Photo:[3] 1k x 1k x24 bit	3	0.06 - 0.3 (Lossy)	4.8 - 24 Mbps	8 - 40 kbps
X-ray: 2kx2kx12 bit	6	3 (Lossless)	240Mbps	0.4 Mbps

TABLE II Bandwidth requirements in point-to-point one way application.

Applications (All compressed)	Number of concurrent connections
Voice conversation	3125
CD Quality Stereo	39
Videotelephony	
64 kbps	781
384 kbps	130
MPEG video one-way real-time delivery (point-to-point/multicast/broadcast)	6
Nonreal-time video delivery	Number of transactions per hour
MPEG video	3
HDTV	0.25
Image delivery	Number of transactions per minute
1kx1kx24 photos	250 - 1250
2kx2kx12bit X-rays	25

TABLE III Maximum number of concurrent connections or transaction rates supported on a 10 Mbps channel.

1 Less than the typical time required to go to the local video store.
2 Overnight delivery
3 Photo size, resolution and color depth varies.