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IP OVER ATM

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Abstract

Integrated Service Internet running real-time and multimedia applications is rapidly becoming a reality. Meanwhile, ATM technology is appearing in the marketplace. It is an important problem to integrate ATM networks into this Integrated Service Internet. One of the approaches, classical IP over ATM, is now widely deployed, effectively solving the problem of internetworking and interoperability. A key remaining issue is to provide the QoS guarantees for Internet traffic running through ATM subnets. This thesis describes a priority scheme; named User Priority, for providing IP integrated Service with quality-of-service over ATM switched virtual circuits (SVCs) to get better performance of packet delivery.

Define the User Priority field as a three-bit field which uses Type of Service (TOS) field in the IP datagram header. This yields 8 different service classes with value 7=highest priority and 0=lowest priority. Class 6 and Class 7 services are for real-time traffics and have their own VCs. Packets with Class 0 through 5 are sent on aggregate VC. This method allows packets to be treated differently according to their priorities such that they can take advantage of the various QoS guarantees provided by ATM networks. This method is backward compatible with existing IP implementation. These newer options need only be implemented on the end systems that want to take advantage of them.

Keywords: IP over ATM, LAN Emulation, Resource Reservation, OoS (Quality of Service).

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Chapter I: Introduction

1. Introduction

Asynchronous Transfer Mode, commonly known as the acronym of ATM, is the mostly widely studied and implemented form of cell networking. ATM began as a technology designed specifically to address the needs of the international telecommunications carrier community. It has evolved over the past few years and various protocols and interfaces are defined in a set of standards created by the International Telecommunications Union (ITU). This gives network designers a solid base on which to build ATM networks. ATM is the underlying transmission system for ITU's next-generation ISDN, Broadband ISDN. B-ISDN is designed to provide subscriber communications services over a wide range of bit rates from a few megabits to several gigabits. The current ATM standards are designed to allow subscribers access to the telephone networks at speeds of up to 622 megabits/s and it is expected that eventually, gigabit speeds will also be supported, as the underlying ATM transmission system is clearly capable of gigabit speeds.

To make it suitable for use hi LANs as well as WANs, the ATM Forum, an international non-profit organization, continues to add functionalities to the ITU's work. The result is a networking scheme that lets users integrate their telecommunications and data networks. ATM technology provides a way of linking a wide range of devices from telephones to computers using one seamless network. It removes the distinction between local area and wide area networks, integrating them into one network, hi the local environment, such as an office or department, ATM can be used to replace or augment LAN technologies such as Ethernet, Token Ring, or FDDI. hi the wide area environment, ATM can be used as an alternative to frame relay, X.25, or statistical multiplexers.

It is clear that Asynchronous Transfer Mode (ATM) technology will play a central role in the evolution of current workgroup, campus and enterprise networks. ATM provides some important advantages over existing LAN and WAN technologies, such as the flexibility of scalable bandwidth and guarantees of Quality of Service (QoS). These advantages can facilitate new classes of applications such as multimedia.

The major selling point of ATM is that it is the first technology that can deliver different types of traffic, such as voice, video and data, over a single digital transport mechanism. ATM can also handle scalable amounts of bandwidth, as a result of its switching architecture, which can support multimedia applications and network growth for years to come. Especially as the Internet Integrated Service (DS) is becoming important, ATM will play an important role as a backbone network technology for Internet.

However, in the very competitive market, ATM can not be the sole technology used. It is going to cooperate with existing network technologies in Internet environment. The combined networks are hoped to provide the guarantees of QoS, which is required by network users and for the performance of Internet. These benefits, however, come at a price. Contrary to common misconceptions, ATM is a very complex technology, perhaps the most complex ever developed by the networking industry. While the structure of ATM cells and cell switching do facilitate the development of hardwired and high performance ATM switches, the deployment of ATM networks requires an infrastructure which consists of layers of highly complex protocols and softwares. This infrastructure will allow individual ATM switches to be linked into a network, and such networks can internet work with the vast installed base of existing local and wide area networks.

ATM network is a popular topic in the field of computer communications recently. To harmonize this new technology with the existing Internet, some schemes for managing the transmission of IP datagrams over ATM networks are required. These schemes ideally will preserve the advantages of ATM and effectively bridge the gaps between the legacy IP datagrams networks (connectionless) and high speed ATM networks (connection-oriented).

Using IP and ATM together presents some interesting challenges because they differ in fundamental ways, from their respective models of data forwarding (connectionless vs. connection-oriented) to support for the preferential treatment of packets (no support vs. the potential for support guarantees). This paper will introduce some strategies and propose a priority scheme to support QoS for IP datagrams carried over the interconnected ATM and TCP/IP networks.

The implications of various IP-over-ATM strategies on network performance, particularly the aspects relating to quality of service, virtual circuit multiplexing, and virtual circuit management are also addressed.

In Section 2, the backgrounds and motivations for this work will be introduced. Some related works are illustrated in Section 3, and the protocol design principles are shown in Section 4. In Section 5, the protocol operations for using the ATM virtual circuits with guarantee of performance to carry IP datagrams are shown. Last, conclusions and further studies are stated.

Chapter II: Background and Motivation

Asynchronous Transfer Mode (ATM) is a new network technology which is designed for broadband "integrated services" networks. These networks are capable of carrying multimedia data as well as conventional computer data traffic. ATM is one kind of connection-oriented protocol and adopts small, fixed-sized packets called cells to carry data through a switching network. The protocols used for packaging various kinds of data are called ATM Adaptation Layers (AALs). They fragment larger, variable-sized packets into multiple cells for transmission in the ATM network. At destination, these cells will be reassembled into the packets. With the appropriate scheduling disciplines in the network switches and the signaling software, ATM networks will be able to deal with real-time traffic and provide some guarantees of performance, such as bounds on bandwidth, delay, and packet loss. These performance guarantees are necessary for many network applications. Especially in recent years, several new classes of distributed applications, which require performance guarantees, have been developed. For example, remote video, multimedia conferencing, virtual reality and long-distance learning. It is becoming clear that the Internet's primitive service model is inadequate for these new applications. This inadequacy stems from the Internet's point-to-point best effort service model, which is unable to address two application requirements, bandwidth and delay guarantees at transmission time.

As the switching technology progresses, ATM is gaining popularity. However, it is uncertain whether ATM will be ubiquitous or dominant. The main reason is the fast Ethernet is getting popular because it is simpler, cheaper and backward compatible with existing lOMbps Ethernet LAN implementations. The existing IP networks based on Ethernet or token ring need not change. Replacing these networks with ATM will be costly and unnecessary. For the foreseeable future, it appears that large-scale connectivity will continue to involve multiple and heterogeneous networks. In many circumstances, at least ATM networks will be used as backbones, connecting existing LANs.

2.1 ATM

Asynchronous Transfer Mode, or ATM is a network transfer technique capable of supporting a wide variety multimedia application with diverse service and performance requirements. It supports traffic bandwiths ranging from a few kilobites per second to several hundred megabits per second. And traffic types ranging from continuous, fixed-rate traffic to highly bursty traffic.

ATM is a form of packet-switching technology. That is, ATM networks transmit their information in small,fixed-length packets called "cells", each of which contains 48 octets (or bytes) of data and 5 octets of header information. The small , fixed cell size was chosen to facilitate the rapid processing of packets in hardware And to minimize the amount of the time required to fill a single packet.

ATM is also connection-oriented. In other words, a virtual connection must be established before a "call" can take place, where a call is defined as the transfer of information between two or more end points.

Another important characteristic of ATM is that its network functions are typically implemented in hardware. With the introduction of high speed fiber optic transmission lines, the communication bottleneck has shifted from the communication links to the processing at switching nodes and at terminal equipment. Hardware implementation is necessary to overcome this bottleneck, because it minimizes the cell processing overhead, thereby allowing the network to match link rates on the order of Gbit/s.

Finally, as its name indicates, ATM is asynchronous. Time is slotted into cell-sized intervals, and slots are assigned to calls in an asynchronous, demand-based manner. Because slots are allocated to calls on demand ATM can easily accommodate traffic whose bit rate fluctuates over time . moreover, in ATM also gains bandwidth efficiency by being able to statistically multiplex bursty traffic sources.

Since bursty traffic does not require continuous allocation of the bandwidth at its peak rate, statistical multiplexing allows a large number of bursty sources to share the network's bandwidth. Since its birth in the mid-1980s, ATM has been fortified bt a number of robust standards and realized by a significant number of network equipment manufacturers.

International standards-making bodies such as the ITU and independent consortia like the ATM forum have developed a significant body of standards and implementation agrements for ATM

2.1.1 ATM Standards

The telecommunication standardization sector of the ITU, the international standards agency commissioned by the United Nations for the global standardization of telecommunication, has developed standards for ATM networks. Other standards bodies and consortia have also contributed to the development of ATM.

2.1.2 Protocol Reference Model

The purpose of the protocol reference model is to clarify the funtions that ATM networks perform by grouping them into a set of interrelated, function-specific layers and planes. The reference model consist ofa user plane , a control plane and a management plane. within the user and control planes is a hierarchical set of layers.

The user plane definesa set of functions for the transfer of user information between communication end-points; The conrol plane defines the control functions such as call establishment, call maintanance, and call release; and the management plane defines the operations necessary to control information flow between planes and layers, and to maintain accurate and fault –tolerant network operation.

Within the user and control planes, there are three layers: the physical layer, the ATM layer, and the ATM adaptation layer (ALL).

The physical layer performs primarily bit level functions, the ATM layer is primarily responsible for the switching of ATM cells, and the ATM adaptation layer is responsible for the conversion of higher layer protocol frames into ATM cells. The functions that the physical, ATM, and adaptation layers perform are described in more detail in the following.

2.1.3 Physical Layer

The physical layer is divided into two sublayers: the physical medium sublayer and the transmission converge sublayer.

2.1.3.1 Physical Medium (PM) Sublayer : The physical medium sublayer performs medium-dependent functions. For example, it provides bit transmission capabilities including bit alignment, line coding and electrical/optical conversion. The PM sublayer is also responsible for bit itming, i.e.., the insertion and extraction of bit timing information. The PM sublayer currently supports two types of interface : optical and electrical

2.1.3.2 Transmission Convergence (TC) Sublayer : Above the physical medium sublayer is the transmission converge sublayer, which is primarily responsible for the framing of data transported over the physical medium. The ITU_T recomendation specifies two options for TC sublayer transmission frame structure cell-based and synchronous digital hierarchy (SDH). In the cell-based case, cells are transported continuously without any regular frame structure. Under SDH, cells are carried in a special frame structure based on the north American SONET (Synchronous Optical Network) protocol.

Regardless of which transmission frame structure is used, the TC sublayer is responsible for the following four funtions: Cell rate decoupling, header error control, celldelineation, and transmission frame adaptation. Cell rate decoupling is the insertion of idle cells at the sending side to adapt the ATM cell stream's rate to the rate of the transmission path .

Header error control is the insertion of an 8-bit CRC polynomial in the ATM cell header to protectthe contents of the ATM cell header. Cell delineation is the detection of cell boundaries. Transmission frame adaptation is the encapsulation of departing cells into an appropriate framing structure

2.1.4 ATM Layer

The ATM layer lies a top the physical layer and specifies the functions required for the switching and flow control of ATM cells,There are two interfaces in an ATM network: The user network interface (UNI) between the ATM end point and the ATM switch, and the network-network interface (NNI) between two ATM switches.

Although a 48 octet cell payload is used at both interfaces, the 5 octet cell header differs slightly at these interfaces. The VCI and VPI fieldsare identifier values for virtual channel (VC) and virtual path (VP), respectively. A virtual channel connects two ATM communication end-points. A virtual path connects two ATM devices, which can be switches or end-points, and several virtual channels may be multiplexed onto the same virtual path.

The 2-bit PT field identifies whether the cellpayload contains data or control information. The CLP bit is usedby the user for explicit indication of cell loss priority. İf the value of the CLP is 1 the the cell is subjected to discarding in case of congestion. The HEC field is an 8 bit CRC polynomial that protects the contents of the cell header.

The GFC field, which appears only at the UNI, is used to assist the customer premises network in controlling the traffic flow for different qualities of service .
At the time of writing the exact procedures for use of this field have not been agreed upon .

2.1.4.1 ATM Layer Functions

The primary function of the ATM layer is VPI/VCI translation. As ATM cells arrive at ATM switches, the VPI and VCI values contained in their headers are examined by the switch to determine which outport should be used to forward the cell. İn the process, the switch translates the cell's original VPI and VCI values into new outgoing VPI and VCI values, which are used in turn by the next ATM switch to send the cell toward its intended destination.

The table used to perform this translation is initialized during the establishment. An ATM switch may either be a VP switch, in which case it only translates the VPI values contained in cell headers, or it may be a VP/VC switch, in which case it translates the incoming VCI value into an outgoing VPI/VCI pair.

Since VPI and VCI values do not represent a unique end-to-end virtual connection. They can be reused at different switches through the network. This is important, because the VPI and VCI fields are limited in lenght and would be quickly exhausted if they were used simply as destination address.

The ATM layer supports two types of virtual connections: switched virtual connection (SVC) and permanent, or semipermanent , virtual connections(PVC). Switched virtual connections are established and torn down dynamically by an ATM signaling procedure. That is they only exist for the duration of a single call.

Permanent virtual connections, on the other hand, are established by networkadministrators and continue to exist as long as the administrator leavesthem up, even if they are not used to transmit data. Other important functions of the ATM layer include cell multiplexing and demultiplexing, cell header creation and extraction, and genericflow control.

Cell multiplexing is the merging of cells from several calls onto a single transmission path , cell header creation is the attachment of a 5- octet cell header to each 48 octet block of user payload, and generic flow control is used at the UNI to orevent short-term overload conditions from occurring within the network

2.1.4.2 ATM Layer Service Categories

The ATM Forum and ITU-T have defined several distinct service categories at ATM layer. The categories defined by the ATM forum include constant bit rate (CBR), real-time variable bit rate (VBR-rt), non real-time variable bit rate (VBR-nrt), available bit rate (ABR), and unsipecific bit rate (UBR). ITU-T defines four service categories, namely, deterministic bit rate(DBR), statistical bit rate (SBR), avalable bit rate (ABR) and ATM block transfer(ABT). The first of the three ITU-T service categories correspond roughly to the ATM Forum's CBR,VBR and ABR classifications, respectively .

The fourth service category, ABT, is solely defined by ITU-T ans is intended for bursty data application.The UBR category defined bt the ATM Forum is for calls that request no quality of service guarantees at all. The constant bit rate CBR (or deterministic bit rate DBR) service category provides a very strict QoS guararentee. It is targeted at real-time applications, such as voice and raw video, which mandate severe restrictions on delay, delay variance(jitter) and cell loss rate.

The only traffic description required by the CBR service are the peak cell rate and the cell delay variation tolerance. A fixed amount of bandwidth, determined primarily by the call's peak cell rate, is reservedfor each CBR connection. The real-time variable bit rate VBR-rt (or statistical bit rate SBR) service category is intended for real time bursty application, which also require strict QoS guarantees.

The primary difference CBR and VBR-rt is in the traffic descriptions they use. The VBR-rt service requires the specification of the sustained cell rate and bursty tolerance in addition to the peak cell rate and the cell delay variation tolerance. The ATM Forum also defines a non-real-time VBR-nrt service category, in which cell delay variance is not guaranteed.The available bit rate(ABR) service category is defined to exploit the network's unutilized bandwidth. It is intended for non-real time data application in which the source is amenableto enforced adjustment of its transmission rate.

A minimum cell rate is reserved for the ABR connection and therefore guaranteed by the network. When the network has unutilized bandwidth, ABR sources are allowed to increase their cell rates up to an allowed cell rate(ACR), a value which is periodically updated by the ABR flow control mechanism. The value of ACR always falls between the minimum and the peak cell rate for the connection and is determined by the network

The ATM forum defines another service category for non-real-time application called the unspecified bit rate (UBR) service caregory. UBR service is entirelybest efford; the call is provided with no QoS guarantees. .The ITU-T also defines an additional sewrvice category for non-real-time data applications. The ATM block transfer (ABT) service category is intended for the transmission option (ABT/IT), the block of data is sent at the same time as the reservation request.

If bandwidth is not available for transporting block, then it is simply discarded; and the source must retransmit it. In the ABT service with delayed transmission (ABT/DT); the source waits for a confirmation from the network that enough bandwitdh is available before transmitting the block of tha data. In both cases, the network temprorarily resource reserves bandwitdh according to the peak. Cell rate for each block.Imediately after transporting the block, the network releases the reserved bandwidh

2.1.4.3 ATM Adaptation Layer

The ATM adaptation layer (AAL), which resides a top ATM layer, is responsible for mapping the requirements of higher layer protocols onto the ATM network. It operates in ATM devices at the edge of the ATM network and is totally up sent in ATM switches. The adaptation layer is divided into two sublayers: The convergence sublayer (CS), which performs error detection and handling, timing and clock recovery and the segmentation and reassembly (SAR) sublayer, which performs segmentation of convergence sublayer protocol data units (PDUs) into ATM cell-sized SAR sublayer service data units data units (SDUs) and vice versa in order to support different service requirements, the ITU-T proposed for AAL-specific services classes.

Note that while these ALL service classes are similar in many ways to the ATM layer service catagories defined in the perivious section, they are not the same; each exists at a different layer of the protocol reference model, and each requires a different set of functions. ALL service class A corresponds to constant bit rate (CBR), services with a timing the relation required between source and destination. The connection mode is connection – oriented . CBR audio and video blong to this class. Class B corresponds to variable bit rate (VBR) services. This class also requires timin between sources and destination , and its mode is connection-oriented. VBR audio and video are examples of class B services. Class C also corresponds to VBR connection –oriented services but the timing between source and destination needs not be related. Class C includes connection-oriented data transfer such as X.25, signaling and future high speed data services. Class D corresponds to connectionless services. Connectionless data services such as those supported by LANs and WANs are examples of class D services .

Four AAL types , each with a unique SAR supplier and CS sublayer, are defined to support the four service classes. ALL type 1 supports constant bit rate services (Class A), and AAL type 2 supports available bit rate services with a timing relation between source and destination (Class B). ALL type 3 /4 was orginally specified as two different AAL type (Type 3 and Type 4), but due to their inherient similarities, they were eventually merget to support both Class C and Class D services. AAL Type 5 also supports class C and Class D services

2.1.4.4 ALL Type5

Currently the most widely used adaptation layer is AAL type 5. AAL type 5 supports connection-oriented and connectionless services in which there is no timing relation between source and destinamtion(class C and class D). Its functionality was intentionaly made simple in order to support high speed data transfer. AAL type 5 assumes that the layres above the ATM adaptation layer can perform error recovery. Retransmition and sequence numbering when require and those it does not provide this functions. Therefore, only none assured operation is provided; lost or corrupted AAL type 5 packet will not be corrected by retransmition.

Figure2 depicts the SAR-SDU format for AAL type 5 . the SAR supplier of AAL type 5 performs segmentation of CS-PDU into a size suitable for the SAR-SDU pay load. Unlike other AAL types, Type 5 devotes the entire 48-octet payload of the ATM cell to the SAR-SDU ; there is no overhead. An AAL specific flag in the ATM Payload Type (PT)

Field of the cell header is set when the last cell of a CS-PDU is sent . the assembly of the CS-PDU frames at the destionation is controlled by using this flag.

Figure 1 depicts the CS-PDU format for AAL type 5. it conteins the user data payload , along with any recessary padding bits (PAD) and a CS-PDU trailer, which are added by the CS supplier when it recveives the uder information from the higher layer . the CS – PDU is padded using 0 + 47 bytes of PAD field to make the land of the CS_PDU and integral muliple of 48 bytes (the size of the SAR -SDU)at the receiving end, reassembled PDU is passed to the CS sublayer from the SAR sublayer, CRC values are then calculared and compared.

If there is no error ,the PAD field is removed by using the value of lenght field(LF) in the CS -PDU trailer, and user data is passed to the higher layer. İf an error is detected,the erroneous information is either deliverded to the use or discardedaccording to user 's choice . the use of the CF field is for further study

2.2 ATM Signaling

ATM follows the principle of out-of-band signaling that was establihed for N-ISDN . in other words, signaling and data channels are separate. The main purposes of signaling are:

1) To establish ,maintain and release ATM virtual connections.
2) To negotiate the traffic parameters of new connections

The ATM signaling standards support the creation of point to point as well as multicast connections . Typically certain VCI and VPI values are reserved by ATM networks for signaling messages. If additional signaling VCs are required,they may be establish through the process of meta-signaling.

2.3 The Advantages and Drawbacks in IP Networks

Legacy IP network has some advantages that ATM network can not provide. It's the de facto standard of the networking world and allows the construction a very large network with less central management. Also its robust routing of packets can accommodate failures in channels. Besides, it uses a connectionless model for simple heterogeneous internetworking. However, IP packets are based on the datagram model and are not really suitable for real-time applications.

It is routed independently via a connectionless model and out of order packet delivery is possible. Also, the delivery of IP packets is best effort only. Hence, QoS is not supported.

The Internet Engineering Task Force is currently designing a successor to IP, known as IP version 6 or IPv6. In the context of IPv6, the original Internet Protocol is referred to as IP version 4, or IPv4. IPv6 is a network-layer protocol, which addresses the primary limitations of IPv4, while retaining much of the same basic protocol architecture. Among the new features of IPv6 are an expanded address space (128-bit addresses vs. 32-bit IPv4 addresses), ease of route aggregation for scalability, a redesigned packet header (see Figure 1) for efficient packet processing, and explicit support for security and authentication.

Version 4-bit	Priority 4-bit	Flow Label 24-bit
Payload Length 16-bit	Next Header 8-bit	Hop Limit 8-bit
Source Address 128-bit		
Destination Address 128-bit		

Figure 1: IPv6 Header Format.

The 4-bit Priority field in the IPv6 header enables a source to identify the desired delivery priority of its packets, relative to other packets from the same source. This field provides a simple priority scheme for transmitting packets.

2.4 The Advantages and Drawbacks in ATM Networks

There are some advantages in ATM technology. Its scalable bandwidth can satisfy the widely different traffic requirements. Guarantees of QoS facilitate emerging classes of new applications (e.g. multimedia). Also, in-order delivery of packets via connection oriented virtual circuits can be provided. Besides, multiple traffic streams (voice, data, video, etc.) are able to share the same physical paths. Although ATM has so many advantages, it also has some of its own problems. ATM's connection-oriented model makes it the most complex networking technology ever designed by industry. There will be a difficulty in rerouting virtual circuits when the connection failures occur. The most common ATM Adaptation Layer (AAL) protocol, AAL5, cannot be used to transmit multicast packets on a unidirectional multipoint-to-multipoint connection. The last and most important problem is the internetworking between the ATM networks and the existing network infrastructure (predominantly TCP/IP) is an important problem—because it needs special signaling, addressing and routing protocols.

2.5 The Concepts of IP Over ATM

In the current Internet, the solutions to forward data through a heterogeneous internet work is provided by the Internet Protocol (IP). IP is almost entirely independent of the subnet technology used. It just makes a few assumptions about the nature of individual subnets. IP packets can traverse many different types of subnets (including ATM networks) without either the senders or receivers being aware of the details of the networks encountered along the path. Unlike ATM, IP is a datagram protocol and does not require the establishment of connections before data are sent.

As ATM and the Internet will likely coexist in the future, it is desirable that hosts attaching to-these two types of networks can exchange data. One approach is to use an ATM network (with an appropriate adaptation layer) as a datalink layer, similar to Ethernet and FDDI, as shown in Figure 2.

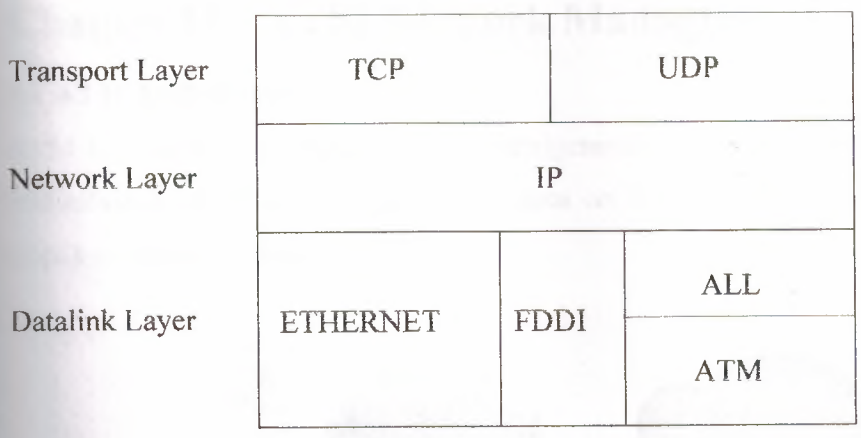

Figure 2: The Internet Protocol Suite and Datalink Layer

This method is commonly referred to as IP-over- ATM (BPOA). An interesting aspect in this approach is how to preserve the quality of service under the IP conversation. Besides, the issue of ATM QoS will impact the multiplexing and virtual circuit management. In fact, the performance of individual IP conversations and the resource reservation for a given virtual circuit will be a trade off among different multiplexing policies. Different virtual circuit management strategies will impact the resource reservations and delays.

Chapter III: ATM Network Management

3.1 ATM Architecture

ATM is a method for providing a a heterogeneous mix of network protocols to support transmission of voice, data and video data on a single network, using cell-relay and circuit switching techniques.

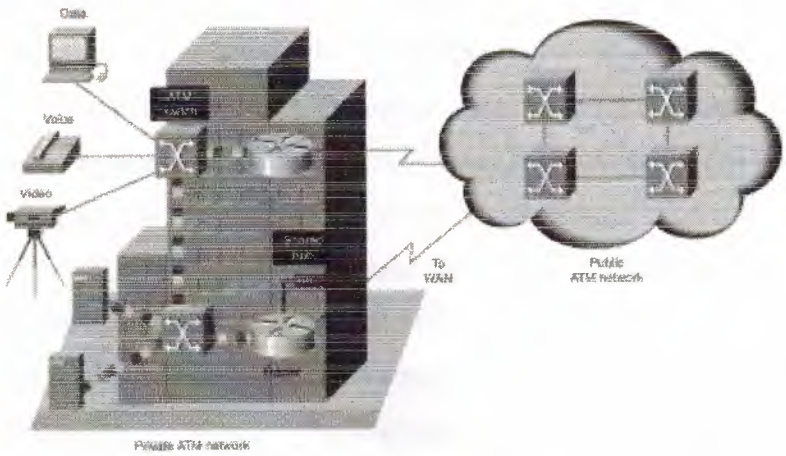

Figure 3: ATM Network

ATM carries all traffic in a stream of fixed-size packets (cells), each containing 5 bytes of header information and a 48-byte information field (payload). The reason for choosing a fixed-size packet is to ensure that the switching and multiplexing function can be carried out as quickly as possible. ATM is a connection-oriented service in that before two systems on the network can communicate, they need to inform all intermediate switches about their service requirements and traffic parameters. This is similar to the telephone networks where a fixed path is set up from the calling party to the receiving party.

In ATM networks, each connection is called a virtual circuit (VC), and it allows the capacity of each link to be shared by all connections on a demand basis rather than by fixed allocations. The connections allow the network to guarantee the quality of service (QoS) by limiting the number of VCs. Typically, a user declares its key service requirements at the time the connection is established, along with other traffic parameters and may agree to control these parameters dynamically as demanded by the network.

3.2 ATM Protocol

The protocol is divided into three layers:-

1. ATM adaptation layer (AAL).
2. ATM layer.
3. Physical layer .

3.2.1 ATM Adaptation Layer

The ATM Adaptation Layer (AAL) interfaces the higher layer protocols to the ATM Layer. It relays ATM cells both from the upper layers to the ATM Layer and vice versa. When relaying information received from the higher layers to the ATM Layer, the AAL segments the data into ATM cells. When relaying information received from the ATM Layer to the higher layers, the AAL must take the cells and reassemble the payloads into a format the higher layers can understand. This is called segmentation and reassembly (SAR). Four types of AALs are used, each supporting a different type of traffic or service expected to be used on ATM networks. The service classes and the corresponding types of AALs are as follows:

3.3 Class A - Constant Bit Rate (CBR) Service

AAL1 supports a connection-oriented service in which the bit rate is constant. Examples of this service include 64 Kbit/sec voice, fixed-rate uncompressed video and leased lines for private data networks.

3.3.1 Class B - Variable Bit Rate (VBR) Service

AAL2 supports a connection-oriented service in which the bit rate is variable but cell delay is controlled (or bounded). Examples of this service include compressed voice or video data. The requirement on bounded delay for delivery is necessary to allow the receiver time to reconstruct the original uncompressed voice or video. At the moment, AAL2 has not been fully developed.

3.3.2 Class C - Connection-Oriented Data Service

Examples of AAL3/4 services include connection oriented file transfer and general data network applications where a connection is set up before data is transferred. This service has variable bit rate and does not require bounded delay for delivery.

3.3.3 Class D - Connectionless Data Service

AAL5 is similar to AAL3/4, but has a simplified information header scheme that requires only one header per data unit. Examples of this service include datagram traffic and data network applications where no connection is set up before data is transferred.

Although each AAL is optimized for a specific type of traffic, there is no requirement that AALs designed for one class of traffic cannot be used for another.

3.4 ATM Layer

The ATM layer provides an interface between the AAL and the physical layer. This layer is responsible for relaying cells from the AAL to the physical layer for transmission and from the physical layer to the AAL for use at the end systems. When it is inside an end system, the ATM layer receives a stream of cells from the physical layer and transmits either cells with new data or empty cells if there is no data to send. When it is inside a switch, the ATM layer determines where the incoming cells should be forwarded to, resets the corresponding connection identifiers and forwards the cells to the next link. In addition, it buffers incoming and outgoing cells, and handles various traffic management functions such as cell loss priority marking, congestion indication, and generic flow control access.

The fields in the ATM header define the functionality of the ATM layer. The format of the header for ATM cells has two different forms, one for use at the user-to-network interface (UNI) and the other for use internal to the network, the network-to-node interface (NNI). At the UNI, the header dedicates four bits to a function called generic flow control (GFC), which was originally designed to control the amount of traffic entering the network. This allows the UNI to limit the amount of data entering the network during periods of congestion. At the NNI, these four bits are allocated to the virtual path identifier (VPI).

The VPI and the virtual channel identifier (VCI) together form the routing field, which associates each cell with a particular channel or circuit. The VCI is a single-channel identifier; the VPI allows grouping of VCs with different VCIs and allows the group to be switched together as an entity. However, the VPIs and VCIs have significance only on the local link; the contents of the routing field will generally change as the cell traverses from link to link. For the UNI, the routing field contains 24 bits and the interface can support over 16 million sessions. At the NNI, the field contains 28 bits, allowing over 268 million sessions to share a link within a subnet.

Values VCI	**Function**
5	Signaling from an edge device to its switch (ingress switch)
16	ILMI for link parameter exchanges
18	PNNI for ATM routing

Table 1: Commonly Used VCI

The payload type indicator (PTI) field is used to distinguish between cells carrying user data and cells containing control information. This allows control and signaling data to be transmitted on a different subchannel from user data and hence separation of user and control data. A particular combination is used by the AAL if the cell is a part of an AAL5 connection. Another combination is used to indicate that the cell has experienced congestion.

The cell loss priority (CLP) bit provides the network with a selective discard capability. This bit can be set by a user to flag lower-priority cells that can be discarded by the network during periods of congestion. For example, data applications generally cannot suffer any cell loss without the need for retransmission, while voice and video traffic can usually tolerate minor cell loss.

The header error check (HEC) field is used to reduce errors in the header that cause a misrouting of the cell for one user into another user's data stream. This field contains the result of an 8-bit CRC checksum on the ATM header (but not on the data). In addition, single-bit errors commonly produced by fibre optic links can be corrected.

3.4.1 The Physical Layer

The physical layer defines the bit timing and other characteristics for encoding and decoding the data into suitable electrical/optical waveforms for transmission and reception on the specific physical media used. In addition, it also provides cell delineation function, header error check (HEC) generation and processing, performance monitoring, and payload rate matching of the different transport formats used at this layer.

3.4.2 LAN Emulation

In order for ATM is be *useful* as a general network backbone, it must be able to support local area networks for computers. One approach is to provide an ATM protocol to emulate existing LAN services, allowing network layer protocols to operate as if they are still connected to a conventional LAN. The LAN emulation specification defines how an ATM network can emulate a sufficient set of the medium access control (MAC) services of existing LAN technology (eg. Ethernet), so that higher layer protocols can be used without modification. Look at figure 4.

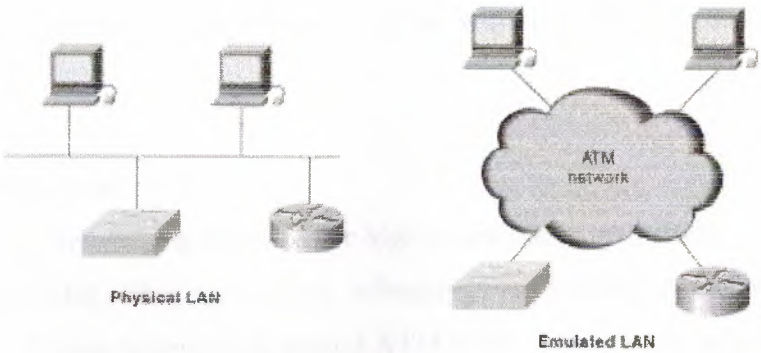

Figure 4 : ATM Emulate the Physical LAN

Another scheme is to implement a LAN emulation service as device drivers below the network layer in ATM-to-legacy LAN bridges and other ATM end systems. In an ATM end system adapter, LAN emulation device drivers would interface with existing driver specifications, such as Network Driver Interface Specification (NDIS) and Open Datalink Interface (ODI) used by TCP/IP and IPX.

A major difference between existing LANs and ATM networks is that LANs are connectionless, whereas ATM natively supports only connection-oriented services. An important function for a LAN emulation service is to be able to support a connectionless service over ATM.

In existing LAN services, a source system sends a data frame to a destination by adding to the frame the destination address and sending it to the network. A receiving system will then accept the frame when the destination address included in the frame matches its own address. In a network with multiple LAN segments, bridges and routers are used to handle the forwarding of the frame to the segment to which the destination system is attached.

In a connection-oriented network such as ATM, however, a source system needs to first set up a connection to the destination before it can transfer data frames. This requires the source system to exchange control information with the network using a signaling protocol.

3.5 ATM Performance

ATM is a nascent networking standard for high-speed packet-switching. It was designed for modern low-noise, fiber networking infrastructures utilizing digital signaling. The major standards-setting organization behind ATM is the ATM Forum, a large consortium of computer networking and telecommunications industry corporations. ATM and Gigabit Ethernet are emerging as the two main high-speed network alternatives going into the next millennium. See figure 5

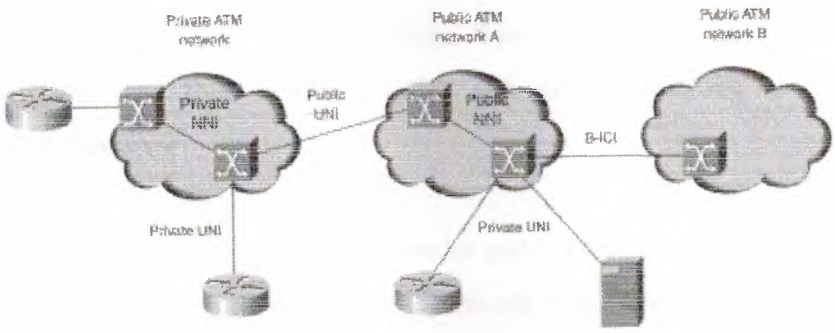

Figure 5 : ATM Interface Specifications

ATM provides a number of improvements over older packet-switching networks. The main ones are:

1. Very high data rates (up to 622 Mbps or more).
2. A variety of Quality of Service specifications, supporting real-time delivery of multimedia streams like audio and video.
3. Scalability from LANs to WANs (i.e. possible seamless networking from desktops around the globe).

There are three layers associated with ATM switching .

1. The physical layer specifies the transmission medium and an encoding scheme. ATM has a native format, but it can also be encapsulated in other protocols, like SONET (a form of TDM)
2. The *ATM layer* occupies a slot just above the physical layer. (See figure5). Since ATM is designed to run over high-speed, low-noise connections, it has only minimal error control (and then only for the packet headers) at this level.
3. The *ATM Adaptation Layer* (AAL) handles translations between the higher layers and the ATM layer. This is where error and flow control are implemented for specific higher-level networking protocols and applications. For example, TCP/IP over ATM could be implemented at this level.

3.5.1 ATM Packet Format

ATM uses fixed-size *cells* instead of packets. Small fixed size cells minimize packet transmission time (and thus queuing delay) and allow switching to be efficiently implemented in hardware.

Cells are 53 bytes long. Each has a 5-byte header and a 48-byte data field. Two variations are supported: user-to-network packets and network-to-network packets. The latter are for internal network control communications.

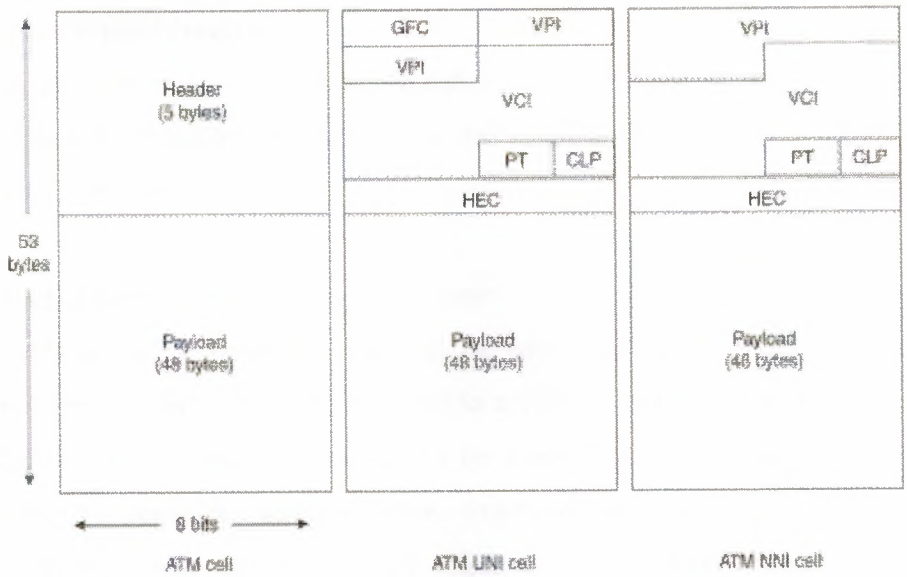

Figure 6 :ATM Packet Format

Addressing is accomplished via a Virtual Path Identifier (8 or 12 bits) and a Virtual Channel Identifier (16 bits) to be discussed later.

The header also includes a short payload type field, which identifies the contents of the data field and an 8-bit CRC header error-control field (recall CRCs from our discussion of data link protocols). In addition to catching a variety of errors, this CRC is designed to allow simple one-bit errors to be corrected under certain conditions.

3.5.2 ATM Virtual Circuits

ATM provides virtual circuits, which are called *Virtual Channel Connections* (VCCs) in ATM parlance. Furthermore, several VCCs can be collected into a group, called a *Virtual Path Connection* (VPC).

There are advantages to this two-level structure:

1. VPCs allow a bunch of VCCs to be configured as a group, minimizing the number of communications required to configure individual VCCs. It also allows certain administrative functions to be controlled by end user rather than the network, reducing administrative overhead and increasing flexibility. For example, a long distance carrier might allocate a VPC to a company with a specific data rate; the company can subdivide the VPC into individual VCCs as it likes for flexibility.
2. It reduces setup and connection time. Once a VCC has been established within a VPC, some of the routing and connection costs have been determined and future VCCs within that VPC can be created with little overhead.

VPCs and VCCs can be set up between two users, between a user and the network (for control signaling) or internally between two network entities.

VPCs and VCCs have the following characteristics:

1. Quality of Service (QoS see next section)
2. Dynamic and semi-permanent connections; this allows long-distance carriers to allocate permanent VPCs for customers, for example.
3. Cell sequence integrity (cells are guaranteed to arrive in the order they were sent, just like a virtual circuit).
4. Traffic negotiation and usage monitoring. Each VCC and VPC have QoS parameters that are set up by "negotiation" with the network (more on this later); the traffic over the VCCs and VPCs are monitored to ensure that they do not exceed these parameters.

3.5.3 ATM QoS Categories

ATM was designed to accommodate a variety of network traffic types that are difficult to handle on traditional packet-switched networks, such as video and audio streams. These kinds of streams cannot suffer large packet delays or data rates that fall below a minimum threshold level without distorting the video or audio stream noticeably at the receiving end (the audio or video breaks up annoyingly).

Let examine the categories of service that have been identified for use under ATM. Later we will examine the methods that ATM uses to provide these capabilities.

ATM allows two classes of quality of service: *real-time* and *non real-time* service.

1. **Real-time service:**Real-time service is for applications like audio or video distribution that require a minimum packet delay and possibly a minimum threshold data rate. There are two subcategories within real-time: CBR and rt-VBR.
2. **Constant bit rate (CBR):** Supplies a constant data rate and tight upper bound on packet delay. Used for videoconferencing, telephony, uncompressed audio/video distribution.
3. **Real-time variable bit rate (rt-VBR):** For bursty sources, such as compressed video, that still have tight time constraints. Allows the network a little more flexibility; can be used with statistical TDM, for example.
4. **Non real-time service:**Non real-time service is for applications that have bursty traffic characteristics and do not have tight constraints on packet delay or delay variation.
5. **Non real-time variable bit rate (nrt-VBR):** This category provides "best-effort" service. The network application specifies the peak cell rate, average cell rate and a measure of how bursty traffic is likely to be and the system will attempt to optimize packet delivery in the non-real time category. Examples: airline reservations, banking transactions, process monitoring
6. **Available bit rate (ABR):** Application specifies peak cell rate and minimum cell rate. The network allocates capacity to service the minimum and then distributes the excess evenly over the rest of the channels.

7. **Unspecified bit rate (UBR):** The lowest class of service; uses excess (whatever is left over) capacity. For file transfer, http, e-mail: typical Internet type uses.

3.5.4 Maintaining QoS

In most packet-switched networks (e.g. the current Internet) there are no controls on usage. End users are free to impose whatever traffic the network will bear on their end. As anyone who has used the Internet knows, this can place a heavy load on the network and introduce congestion and delays for everyone.

In order to be able to provide the above classes of QoS, an ATM network must be able to control congestion and usage tightly. To do this it adopts a different usage model: in order to get on the network you need to negotiate what kind of QoS you will expect. If that requested QoS cannot be supported while still supporting the QoS for existing connections, you will be denied access.

The negotiation/reservation procedure (*called Connection Admission Control*) . The resources are controlled on a VCC/VPC basis. An end-user requests a VCC; that VCC must belong to an existing VPC. If there is no existing VPC then one is created. If there is enough capacity in the VPC to accommodate the QoS requested by the new VCC then the VCC is added to the VPC, otherwise the request is blocked temporarily or denied. By this we can infer that the QoS for the VPC must be able to accommodate the QoS for all the VCCs inside it. The QoS parameters that are negotiated when setting up the VCC or a VPC include:

1. **Peak Cell Rate** the maximum rate at which cells are generated by a source on the connection.
2. **Cell Delay Variation** an upper bound on the variation in cell arrivals relative to the peak rate.
3. **Sustainable Cell Rate** the average rate at which cells are generated by a source on the connection.

4. **Burst Tolerance** an upper bound on the variation in cell arrivals relative to the average rate.

The PCR and CDV are negotiated for all categories of service. The non-real time services may negotiate the SCR and BT as well.

Once a VCC has been allocated, a special algorithm (the Usage Parameter Control) monitors the packet traffic on the connection to ensure that the PCR and CDV do not fall outside the negotiated parameters. For example, if an application attempts to exceed its negotiated data rate by exceeding the peak cell rate, the ATM interface can simply discard packets until the link usage falls into the expected parameters. UPC can be performed on a per-VCC or per-VPC basis.

3.5.5 The ATM Adaptation Layer

The ATM cell format does not have provision for error or flow control of the payload (data), only for the header information. Basically, the ATM layer is designed to switch cells at high rates and that means little processing at the ATM layer. The AAL layer is designed to handle:

1. Transmission errors (single-bit and burst errors are the most frequent types of errors encountered on fiber optic links).
2. Segmentation and reassembly (breaking a data stream into ATM cells at the sender and then reassembling the data stream from the cells at the receiver).
3. Lost cells.
4. Flow and timing control.

Error control is handled in the AAL layer by appending a 32-bit CRC code to the data source. The result is segmented into 48-byte blocks for transmission in ATM cells. At the receiving end, after the data has been reassembled, the CRC is examined for errors.

3.5.6 The ATM Physical Layer

Two compatible physical layer formats are defined for ATM:

1. A cell-based physical layer (native format) is simply a stream of 53-byte ATM format cells.
2. A SDH-based physical layer embeds ATM cells in a synchronous TDM frame. SDH/SONET is a standard for TDM used by many long-distance telephone carriers. This format allows them to carry ATM traffic over their existing SDH/SONET lines.

Chapter IV: IP Over ATM

4. 1 Related Work

Many works of IP over ATM concern various paradigms for these services, such as IETF Classical IP over ATM, ATM Forum LAN Emulation and Multi-Protocol Over ATM, and how they affect the issues of addressing and routing. Multiplexing and virtual circuit management in IP over ATM have been studied in the best-effort service, but quality of service issues are still not addressed yet. Although various solutions for supporting quality of service or performance guarantees in internetwork have been proposed, such as Resource ReSerVation Protocol (RSVP), they did not deal with the specific characteristics of ATM subnets. This report will focus on the support of quality of service in IETF Classical IP over ATM.

4.2 The Paradigms of IP over ATM

The paradigm for IETF Classical IP over ATM is shown in Figure 7. This protocol is commonly used in IP over ATM by multiple ATM switch vendors nowadays.

In Figure 7, several IP members called LIS (Logical IP Subnet) have the same IP network/subnet number and address mask. Members in an LIS directly connect to the ATM network, and resolve IP addresses to ATM addresses via ATMARP and vice versa via InATMARP when using Switched Virtual Circuits (SVCs). If Permanent Virtual Circuit (PVC) is used, members in an LIS will need InATMARP to resolve Virtual Circuits (VCs) to IP addresses. Members in an LIS would be able to communicate to each other via ATM. Two hosts belong to different subnets, but attached to the same ATM network, can only communicate via router that is a member of both subnets.

While classical IP over ATM is potentially inefficient in that a path between ATM-attached hosts may require forwarding through a router, it has the advantage of preserving the original semantics of IP subnets. Another approach, taken by the Routing Over Large Clouds (ROLC) working group of the IETF, seeks to remove the potential inefficiency of the classical model, hi the ROLC model, hosts attached to the same ATM network can communicate directly, even if they do not belong to the same LIS.

Since part of the original IP routing model dictates that hosts on different subnets must communicate via a router (rather than directly), this method forces changes to the way that IP routing and forwarding is performed. A Next-Hop Routing Protocol (NHRP) is used to send data between subnets directly across the ATM network.

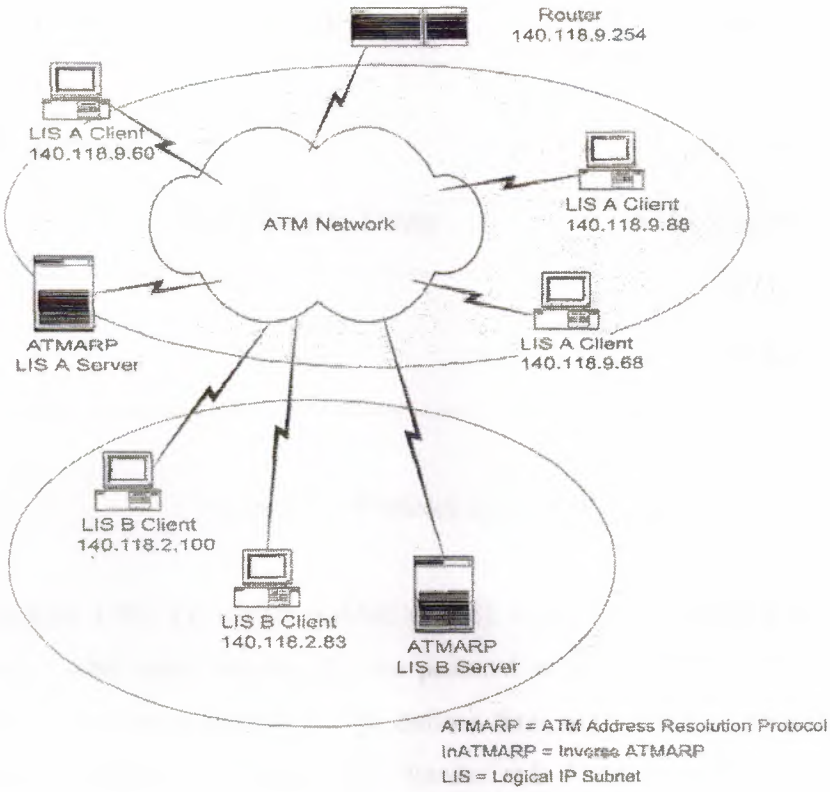

Figure 7: The Model of IETF Classical IP Over ATM

The Protocol hierarchy of IP over ATM is shown in Figure 7. It is the encapsulation and transmission of IP network or link layer packets across an ATM Adaptation Layer (AAL) 5 connection. We know that the audio and video applications generally ran over HDP, which does not provide reliable data transport. These applications are time-sensitive and need not retransmit packets when they are lost.

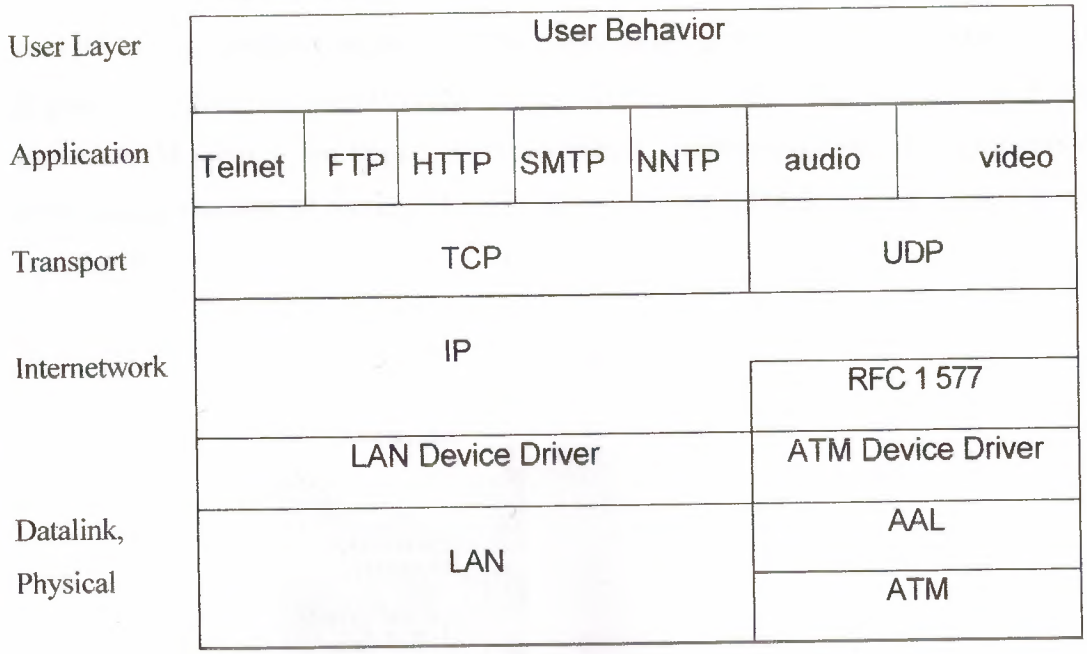

Figure 8 The Protocol Stack of IP over ATM

A third paradigm, LAN Emulation (LANE or LE), as shown in Figure 8 is proposed by ATM Forum. LANE does not modify the protocol used in the routers, but provides a complementary MAC-level service. LANE defines three main areas to emulate 802 LANs (connectionless, broadcast/multicast, 802 hardwired MAC addresses) over ATM networks (connection-oriented, point-to-point, network-defined telephone-like addresses). Also, it defines how an ATM adapter in a host acts as an Ethernet or Token Ring logical interface to the protocol stack. This enables applications and LAN protocols, which were implemented to run above Ethernet or Token Ring LANs to operate without changing anything. This paradigm is quite similar to Classical IP, but supports multiple network protocols (such as IPX or AppleTalk) in addition to IP.

For simplicity's sake, this report chooses the Classical IP approach as its own model of IP over ATM. The similarity among these three approaches is that ATM-attached hosts transmit IP packets to other hosts across ATM virtual circuits. When the cells making up an IP packet exit the ATM network, the IP packet is reassembled, and forwarding of the packet continues according to standard IP routing (if necessary). Thus, an ATM network is treated as a link layer by IP.

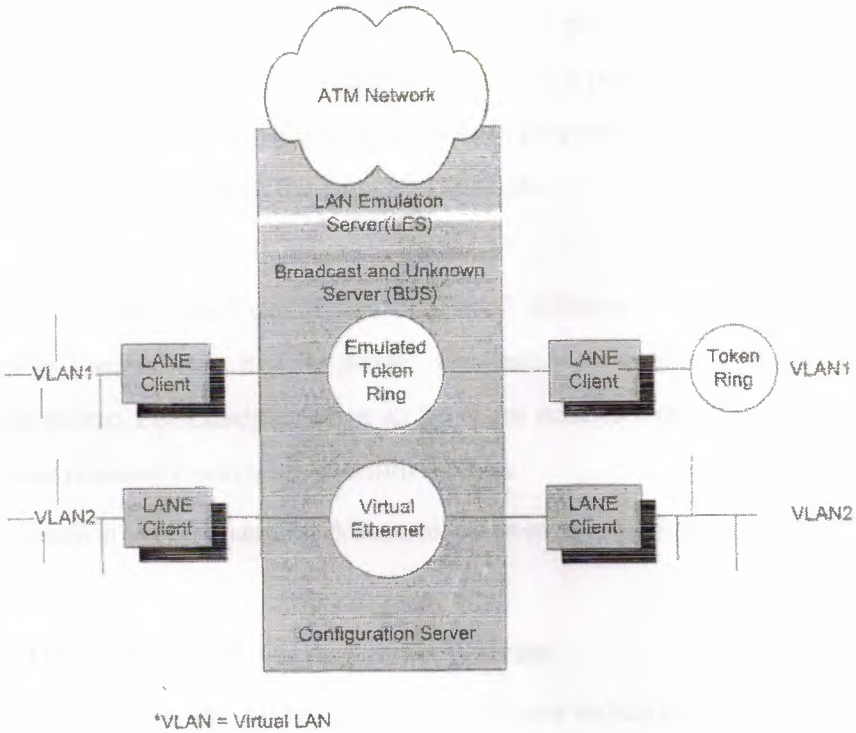

Figure 9 Components of LAN Emulation

4.3. Multiplexing Strategies

Three different multiplexing policies can be considered to support QoS in IP over ATM.

1. Virtual circuit per pair of routers, carrying all traffic passing through the pair of routers, regardless of source or destination host.
2. Virtual circuit per IP conversation (e.g. TCP connection or UDP flow)
3. Virtual circuit per application type (e.g. one virtual circuit for all Telnets passing through a pair of hosts)

When performing router multiplexing, it is difficult to preserve a meaningful QoS to each virtual circuit, because the nature of the aggregate traffic between the routers is unknown. Therefore, all of the policies that use a multiplexing policy of use virtual circuit per router pair do not use any of ATM's QoS features. Virtual circuit per IP conversation is limited by VPI/VCI {(Virtual Path Identifier) / (Virtual Channel Identifier)} quota, and may reserve too many and useless resources on a given virtual circuit. But multimedia applications, such as digital audio and video, will likely have their own VCs because they don't want to be disturbed by other packets. And virtual circuit per application aggregates the same types of applications into one VC. This policy solves the problem as described in Virtual circuit per IP conversation. Thus, a priority scheme is purposed, which will be described in Section 4.1 to take advantage from the latter two methods.

Besides, source and destination may need different resource reservations. This situation is called "asymmetric transfer mode". Destination often needs more resource reservation than the source. For example, when an ftp client sends a request to get files, ftp server will need more resource reservation to return the files.

So does in HTTP protocols. Almost all client-server application have this fact.

4.4Virtual Circuit Management Policies

In this section, we will examine three different virtual circuit management policies

They are PVC, SVC and SVC with cache.

1. **PVC:** A set of permanent virtual circuit is established to carry IP packets. These connections are never torn down.
2. **SVC:** Switched virtual circuits, with some time out policy to be determined, are used to carry IP packets.
3. **SVC with Cache:** Similar to SVC, but with the additional feature that virtual circuit used by other IP conversations can be cached and reused to carry the packets.

We observe that these three policies are not entirely independent. In an IP over ATM service using PVCs, it is impractical to set QoS parameter for an unknown workload traversing a fixed set of virtual circuits. Moreover, the sheer number of virtual circuits required for a complete PVC set may force a multiplexing policy of one virtual circuit per router pair. Using PVC to carry IP traffic will be wasteful and unnecessary. Thus, the only PVC policy that can be considered is QoS-oblivious (no QoS). SVC with cache saves the connection setup and tear-down time. Applying SVC with cache sounds better than using SVC only, but a problem is that the resource requirement of the latter IP conversation may not match the previous IP conversation', such as telnet and ftp. Hence, both SVC and SVC with cache have their own advantage and drawback respectively.

Chapter V: Congestion Control and Protocol Architecture

5. Introduction

A congestion control strategy has been developed to improve the performance of the network under heavy load. It takes two actions; one is a preventive measure and the other is a reactive measure.

The preventive measure tracks the link and buffer space utilization. This measure restricts the call's entry into the network beyond the preset level of link utilization. Whenever the buffer gets filled beyond a threshold point, data cells coming in are dropped based on their priority. The reactive measure monitors the buffer space and its utilization whenever the buffers get filled to a peak value; control cells are generated and are sent to the host computers responsible for buffer overflow. The control cells have a field set to the rate that host computer should use.

The new rate set in the control cells is based on the current rate of the call. After reducing the rate, the host will not reduce it further for a period of time called the recovery period that is also based on the current call rate. After the recovery period is over, the host computer will again transmit at the previous call rate.

5.1 ATM Communication

ATM is a specific packet-oriented transfer mode based on fixed length packets called cells. ATM networks are connection-oriented, which means before any two nodes can communicate, the path of communication has to be fixed between both nodes. Each user declares the traffic parameters and quality of service requirements at the time of connection establishment. ATM supports statistical multiplexing for bandwidth allocation and supports variable bit-rate for multimedia type applications. ATM networks handle multiple data types with varying bit-rates.

The traffic can be of continuous bit-rate type, bursty type (traffic that generates cells only in a certain period in the form of a long burst and generates zero cells in the other period) and variable bit-rate type- The continuous bit-rate type traffic includes real time services such as voice and video.

The bursty type of traffic is similar to a large file or an image transfer type, which is non-real time service. The encoded video data are of the type of variable bit-rate traffic. Different types of calls require different types of quality of services. For real time services, delay is critical compared to cell losses. For a file transfer type of service, which is a non-real time service, cell loss is very critical, but delay is not very stringent.

High-speed networks using B-ISDN lines are expected to use the Asynchronous Transfer Mode. These networks will support applications such as video and audio data. All these services transmit at different rates. The requirement to fulfill the desired quality of services for various traffic types is very complex. For the efficient and fair operation of the network, proper traffic management schemes are important and necessary. Thus, the traffic management schemes are concerned with ensuring that the users who have constantly varying demand will get their desired quality of service. The traffic management problem gets worse for heavy load which may result in the congestion of cells at some intermediate network location. The congestion results in cell loss and cell delay.

ATM networks do not implement any type of cell loss recovery and retransmission mechanism. Thus, congestion control is important to avoid cell loss. Hence, the congestion-control strategy that monitors the network parameters and takes appropriate actions in case of congestion needs to be developed. The development can be done to this strategy that will prevent the network from getting congested and also take reactive actions when the network gets congested.

The aim of the strategy was to minimize the cell loss and cell delay. Also, while reducing the congestion, care should be taken to keep the generation of control cells to a minimum, so that they do not increase the network load when the network is congested. Also, the strategy should respond to congestion as fast as possible. The control cells should reach the sources responsible for congestion in the minimum possible time.

In any network, when the total demand of any resource is greater than the available resource, in any time interval, the resource is said to be congested for that time interval. The different types of resources include link bandwidths, buffers etc. If the sum of all call's bandwidth utilization is greater than the available capacity, then the links are said to be congested. If the total capacity of buffers in the switches is less than the incoming traffic then the switches are said to be congested. Numerous types of strategies have been developed for congestion control.

5.2 Congestion Problem in ATM Networks and its Solutions

The traffic management and congestion control is very critical in any type of network. The congestion control is specifically difficult in ATM networks due to varying types of loads, different service requirements and very high link speeds. Due to high link speed, ATM cells arrive at the switches very fast. In order to avoid congestion at switching nodes, processing time at the switching nodes should be minimum. A lot of work has been done on the congestion control in ATM network. The two main types of the congestion schemes in an ATM network are as in.

5.2.1 Credit-Based Approach

This approach is based on per Mk, per VC and window flow control. Each node has a separate queue for each VC. Each link at one end has a sender and has a receiver at the other end. The receiver monitors the queue length of each VC and determines the number of the cells that the sender can transmit on that VC. This number is called credit.

The sender transmits the cells allowed by this credit. This scheme is called guarantees zero cell loss. The main drawback of this scheme is that it is required to maintain separate queue for each VC. Since for a large switch the number of VCs is very large, the complexity of the switch becomes very high.

5.2.2 Rate-Based Approach

Loop approach, in which case sources can change their rates based on the network's status. This scheme controls the congestion using the current network information. The control cells are sent in the reverse direction, which has the information about the rate at which the sources should emit the cells. Following type of the rate-based schemes have been suggested which are described at length in first part of this chapter.

5.2.2.1 Explicit Rate Control :

In this scheme the network periodically checks the network load and determines the rate at which the sources should transmit. Network sends the control cells to each source, which contains the new rate at which the sources should transmit

5.2.3 Forward Explicit Congestion Notification (FECN)

This is an end-to-end scheme. When a switch gets congested, the switch marks Explicit Forward Congestion Indication bit "EFCI" bit in all the data cells passing through the switch on that path in the forward direction which indicates the congested switch status. When these cells reach the destination, the destination sends congestion notification cells in the reverse direction to notify the source regarding congestion. The sources use this cell information to adjust their rate appropriately.

The main disadvantage of this type of scheme is that congestion control cell's effect is not fast because of round-trip delays, which may be very large in case of Wide Area Networks (WANs).

5.2.4 Backward Explicit Congestion Notification

In this scheme, congestion notification cells are generated by the switches, when they get congested. Thus congestion control cells are generated from the point where congestion occurs. The sources adjust their rate based on the control cell information. This scheme reacts to congestion immediately. Its response to congestion is much faster than the FECN scheme because it avoids the round-trip delay.

This chapter builds a congestion control scheme with congestion control in ATM network using backward propagation of control cells and dropping off of ATM cells based on priorities in case of buffer overflow.

5.3 Congestion Control Strategy

In any network, when the rate at which the cells arrive at the immediate network node exceeds the rate at which the cells can be transmitted, the queue size grows without any bound and delay experienced by the cells is very high. When a point of severe congestion is reached, queuing response results in dramatic growth in delays and cell losses. The same is true for ATM networks. These catastrophic events can be avoided using congestion control strategy.

The various causes for congestion in ATM network are described in the following sections. Also, the strategy implemented in this paper is described at length. ATM networks support multimedia type traffic having variable bit rate. Thus ATM networks support both low speed data and high-speed data transmission. The traffic on ATM network may be of continuous bit rate type or variable bit rate type or bursty type in nature. Each of this type has different quality of service requirements. ATM network should support each call's various qualities of service requirements.

QoSs requirement for different types of calls are different for real-time traffic like voice and video channel delay is critical parameter, while for non-real time bursty traffic like large file or image file cell loss is very important criteria. ATM networks should be able to balance between all these requirements and maintain high performance level.

The performance level of any network degrades in case of congestion. The congestion control is thus concerned with a strategy, which will make the network operate at an acceptable level. Without proper congestion control, performance level of the network decreases resulting in very high cell loss, cell delay and cell jitter. The congestion control strategy developed in this paper has two levels.

5.3.1 Connection-Level Control

This control deals with admission control of calls. At this level, the connection is either accepted or rejected depending upon the call requirements. The scheme developed here supports two types of calls described below.

5.3.2 Continuous Bit Rate (CBRI)

These are real-time services such as voice and video channel. The decision to accept or deny a connection is based on call's bandwidth requirements and available bandwidth on the route from source to destination.

5.3.3 Long Burst

These calls are non-real time services like file or image transfer type which will transfer periodically. The bandwidth is reserved only at the time of transmission and will be freed once transmission period is over. Whenever the source wants to transmit, the source will first try to establish the connection. If all the links on the path have sufficient bandwidth to service this call, then the permission is granted to establish the connection and bandwidth is reserved else service is blocked. If the service is blocked, the source may try after sometime.

5.3.4 Peak Bandwidth Control Module

This module keeps track of the usage of bandwidth by each call. When a call is entered in the network, its service requirement parameters are stored into the system. The violation of peak bandwidth agreement results in cell dropping.

5.3.5 Cell Level Control

In this level of control, the switch monitors the queue length of each output buffer. Two levels of control are implemented here. Peak level control: Each output buffer has the peak buffer value set by the switch. If any of the output buffers reaches its peak value (in terms of cells), the ATM switch will generate control cells for each source sending cells to this buffer.

The control cells have a special field called Backward Explicit Congestion Notification (BECN) to specify congestion notification from the congested switch. These control cells called reverse mode (RM) cells, are sent in the reverse direction, i.e. towards the source with BECN field set to 1. The sources receiving control cells will reduce their speed based on explicit cell rate of control cell.

5.4 Merits of the Proposed Strategy

In the proposed strategy, the round trip delay is avoided since the control cells are sent from the switch itself.

All the buffers are partitioned for control cells and data cells. The control cells are given preference over all the other type of the data. Due to this transmission, delay is avoided. The control cells reach the source very fast.

The sources will reduce their rate as calculated from equation A and B which proportional to the current call rate of its call. Thus, the scheme is fair, i.e. the sources, which are sending at higher rates, are more responsible for buffer overflows; hence they are reduced by higher factor.

The recovery period is also proportional to the peak call cell. The recovery periods of different sources are different. The lower rate sources will resume its normal speed at a faster rate than the sources having higher cell rate. Hence, all the sources will not increase their rates at the same time avoiding congestion chances.

The sources will decrease their rate only once. All the control cells received in the recovery period are ignored. Thus, the calls having long route, i.e. having many hopes will not get reduced by many switches avoiding unfairness for the calls of long route.

The scheme is very useful for the Wide Area Networks (WANs) because the propagation delay in WANs are very high. By using this scheme round-trip delays are avoided, thus sending control cells to the sources as fast as possible.

The scheme avoids unnecessary generation of control cells, avoiding overloading the network in case of congestion. The number of control cells generated is very low.

5.4.1 Simulation Model

Simulation was done using object-oriented programming language C++. An ATM switch consists of 4 input buffers connected to incoming links and 4 output buffers connected to the outgoing links. Each link is simulated as half duplex link of capacity 155 Mbits/sec.

5.4.2 Simulation Module

This module is responsible for generation of an ATM network and different calls. It reads the parameter from the global file for total number of switches in the network, total number of hosts, total number of links and the connectivity between all of them. Once all the data are read, it generates the network. This module is responsible for interpreting the simulation data and generating the results in accordance with the goals of the simulation.

The simulation module accepts the user-defined data for generation of ATM network. Thus simulation module establishes the ATM network using switches, links, input buffer, output buffers and hosts. After establishing the ATM network, the simulation module will call generation module for the generation of calls. Finally, the global clock is started. The simulation will run till it surpasses the simulation time. During the run, various statistics are calculated.

5.4.3 Generation Module

This module is responsible for the generation of calls. It also initializes It also initializes the calls with different characteristics and service requirements. This module keeps track of link usage. If the call is accepted, then this module updates the entire translation table on the route for this call. The hosts are responsible for producing calls and calls will generate the cells based on the rate of the call. The module is responsible for generating initial call data, which is then processed by simulation engine for analysis and study of various results. After initialization, simulation module has all the data regarding total number of calls entered in the network, service categories of each call, generation time of cells for each call, QoSs required by each call. After this, simulation engine is ready to start simulation with all the required data.

5.4.4 Output Buffer Management Module

This module is responsible for the output buffer management. The output buffer capacity is measured in terms of ATM cells. The output buffers of each ATM switch are monitored for its space utilization. The congestion control module keeps track of the output buffer. If the number of cells in the output buffer exceeds the preset peak value, then this module generates congestion control cells and sends them in the reverse direction to the hosts which are responsible for the buffer space utilization. The control cells are sent in the reverse direction of the links from which cells are coming to the output buffer. These cells have the value of the call rate, which the hosts should use to reduce its value.

Also if the number of the cells in the output buffer exceeds the threshold value then the incoming cells having less priority is dropped. If the number of the cells in the buffer is equal to the capacity of the buffer, then all incoming cells are dropped.

5.4.5 Statistics Collector

The statistics collection is vital to any simulation as the primary purpose of performing a simulation is to collect meaningful statistics about the system. Thus, this module is designed very carefully so that required statistics are collected for the system under study. The responsibility to collect the data lies under different modules. The final results are printed by the statistics collector. The statistics collector is a module, which is responsible for printing the final results generated during the simulation run.

5.4.6 Global Clock

The global clock in the simulation controls the local time management of all the elements of simulation model. The clock tick is set to the minimum possible value such that all the events in the model are synchronized properly.

5.5 Simulation Details

Using above ATM model, simulation is carried out. In our model four ATM switches are connected in a cyclic manner. A host on each switch is generating traffic on the ATM network. In the first step, an ATM network and the calls are initialized. The next step is to check all the input and output buffers of each switch. If the RM cells are not present then, the cells from the regular buffers are transferred provided their transmission delay is less than or equal to current time. If the output buffers are filled to their capacity then all the incoming cells are dropped If the output buffer is filled till the threshold value, then all the incoming cells having lower priority are dropped.

If the output buffer is filled to peak value, then control cells are generated for those hosts who are responsible for buffer utilization (those hosts which are sending cells to this output buffer).

5.6 The ATM Service Architecture

The ATM Service Architecture makes use of procedures and parameters for traffic control and congestion control whose primary role is to protect the network and the end-system in order to achieve network performance objectives. An additional role is to optimize the use of network resources. The design of these functions is also aimed at reducing network and end-system complexity while maximizing network utilization. To meet these objectives, the set of functions forming the framework for managing and controlling traffic and congestion can be used in appropriate combinations.

ATM Service Category (or Transfer Capability) relates quality requirements and traffic characteristics to network behavior (procedures and parameters). It is intended to specify a combination of Quality of and traffic parameters that is suitable for a given set of applications (user interpretation) and that allows for specific multiplexing schemes at the ATM layer (network interpretation).

A Service Category used on a given ATM connection, among those that are made available by the network, has to be implicitly or explicitly declared at connection set-up. All service categories apply to both Virtual Channel Connections (VCCs) and Virtual Path Connections (VPCs).

5.7 Generic Network Functions

Connection Admission Control (CAC) is defined as the set of actions taken by the network during the call (virtual connection) set-up phase, or during call re-negotiation phase, to determine whether a connection request can be accepted or rejected. Network resources (port bandwidth and buffer space) are reserved to the incoming connection at each switching element traversed, if so required, by the service category.

Usage Parameter Control (UPC) or Policing is defined as the set of actions taken by the network to monitor and control the traffic offered and the validity of the ATM connection at the User to Network Interface (UNI). It is an essential requirement for any network supporting multiple services. The main purpose of UPC is to protect network resources from malicious and unintentional misbehavior, which can affect the QoS of other already established connections. Procedures based on a Generic Cell Rate Algorithm (GCRA) may be applied to each cell arrival to assess conformance with respect to the traffic contract for the connection. Violations of negotiated parameters are detected and appropriate actions can be taken (eg. cell tagging, discard).

Feedback Controls are defined as the set of actions taken by the network and by the end-systems (possibly cooperating) to regulate the traffic submitted on ATM connections according to the state of network elements. Specific Feedback Control procedures may be associated with a service category.

5.8 Traffic Parameters

A source traffic parameter describes an inherent characteristic of a source. A set of these parameters constitute a Source Traffic Descriptor which, along with Cell Delay Variation Tolerance (CDVT) and a Conformance Definition, characterize an ATM Connection. The following parameters are considered for the purpose of defining the Service Categories:

1) Peak Cell Rate(PCR)
2) Sustainable Cell Rate (SCR)
3) Maximum Burst Size (MBS)
4) Minimum Cell Rate (MCR)
5) QoS Parameters.

The QoS parameters selected to correspond to a network performance objective may be negotiated between the end-systems and the network, e.g., via signaling procedures, or can be taken as default. One or more values of the QoS parameters may be offered on a per connection basis.

5.9 QoS Parameters

1) Cell Delay Variation (CDV).
2) Maximum Cell Transfer Delay (Max CTD).
3) Cell Loss Ratio (CLR).

A number of additional QoS parameters have been identified, but their negotiation is not foreseen, e.g., Cell Error Ratio (CER), Severely Errored Cell Block Ratio (SECBR), Cell Misinsertion Rate (CMR).

5.10 Traffic Contract and Negotiation

A traffic contract specifies the negotiated characteristics of a VP/VC connection at an ATM User Network Interface (either Private or Public UNI). The traffic contract at the Public UNI shall consist of a connection traffic descriptor and a set of QoS parameters for each direction of the ATM layer connection and shall include the definition of a compliant connection. The values of the traffic contract parameters can be specified either explicitly or implicitly. A parameter value is explicitly specified in the initial call establishment message. This can be accomplished via signalling for SVCs (Switched Virtual Connections) or via the Network Management System (NMS) for PVCs (Permanent Virtual Connections) or at subscription time. A parameter value is implicitly specified when its value is assigned by the network using default rules.

5.11 Some Typical Applications

This section identifies some sample applications, which can be seen as appropriate targets for one or more of the defined service categories. These applications are provided to convey teh original intention and to focus on the possible use of service categories, which broadly relate application aspects to network functionality.

However, an application is not constrained by this mappint, and may select any service category consistent with its needs, among those made available by a network.

5.11.1 Typical Applications for CBR

Any data/text/image transfer application which contains smooth enough traffic or for which the end-system's response time requirements justify occupying a fully reserved CBR channel. Examples are:

1) Videoconferencing
2) Interactive Audio (e.g., telephony)
3) Audio/Video Distribution (e.g., television, distance learning, pay-per-view)
4) Audio/Video Retrieval (e.g., video-on-demand, audio library)

For telephony and voiceband services over ATM, e.g., 64 kbit/s N-ISDN-compatible services, the access solution based on AAL1 requires CBR support for taking advantage of delay and variance bounds.

In the multimedia area, a near-term solution for residential services foresees VoD based on MPEG2 (Transport Stream, CBR mode) over AAL5, with transportation being provided by the ATM-layer with CBR service.

5.11.2 Typical Applications for VBR

VBR is suitable for any application for which the end-system can benefit from statistical multiplexing, by sending information at a variable rate, and can tolerate or recover from a potentially small random loss ratio. It is the case for any constant bit rate source, for which variable rate transmission allows more efficient use of network resources without sensible performance impairment.

Real-time VBR, in particular, can be used by native ATM voice with bandwidth compression and silence suppression. For some classes of multimedia communications real-time VBR may be very appropriate. Non-real time VBR can be used for data transfer, e.g., for response-time critical transaction processing applications (e.g., airline reservations, banking transactions, process monitoring) an frame relay interworking.

5.11.3 Typical Applications for ABR

Any non-time critical application running over an end-system capable of varying its emission rate can exploit the ABR service. Examples include LAN interconnection/internetworking services, which are driving the business service market for ATM. These are typically run over router-based protocol stacks like TCP/IP, which can easily vary their emission rate as required by the ABR rate control policy. The support through ABR will likely result in an increased end-to-end performance (goodput). Another application environment suitable for ABR is LAN Emulation.

Other application examples are critical data transfer (e.g., defense information, banking services) super computer applications, and data communications, such as remote procedure call, distributed file services, and computer process swapping/paging.

5.11.4 Typical Applications for UBR

UBR can provide a suitable solution for less demanding applications. Most data applications, e.g., file transfer submitted in the background of a workstation with minimal service requirements, are very tolerant to delay and cell loss (store and forward networks are in fact widely used for these applications). Examples may include:

1) Text/Data/Image Transfer, Messaging, Distribution, Retrieval
2) Remote Terminal (e.g., telecommuting)

The above services can take advantage of any spare bandwidth and will profit from the resultant reduced tariffs ("cheap" services).

5.12 Multimedia Banking

The banking and insurance industries are heavy users of information technology and telecommunication services, with almost every business transaction leading immediately to a computer-assisted process. Major Banks operating world-wide would be unable to conduct business without efficient telecommunication networks.

Banks also increasingly use information technology to serve their customers. An international consortium looked at how self-service banking and advisory support applications for up-to-the-minute analyses of bank products can be used to better serve customers in the future.

The integration of telecommunication and information processing, as well as recent progress in the field of audio and video presentation on computer workstations, is opening up new horizons for marketing and distribution applications. Multimedia product Information consisting of still and moving images, sound or voice sequences, charts, and text can help customers obtain information on the services offered by a bank.

In addition to product presentation, networked multimedia provides an innovative opportunity to hold small video-conferences regardless of distance. High-performance broadband networks are required to provide a high-quality connection; interest here is focused on ATM-based networks, with their capacity for flexible use of bandwidths. Broadband networks are being discussed as an important national infrastructure for the country's economy.

Almost everywhere in Europe, broadband networks are set up and are already interconnected or will be in future. An international consortium has investigated multimedia technology in the banking sector, establishing the BANK (Banking applications using image and broadband communications network) application project.

The high level of customer acceptance of self-service facilities and the increasing efficiency of modern information technology are the driving forces behind an extension of customer self-service, both quantitatively and qualitatively. In the future self-service machines will not only assist in processing transactions, but they will also increasingly serve as a marketing tool, supplying information on more complex products, such as loans, mortgages and investment and portfolio management.

The common feature of these products is that the customer has a number of options regarding the product structure and is therefore often reliant upon advice. In order to preserve the close customer-bank relationship the customer must always have access to the advice of his personal advisor at the bank. Thus the outstanding features of future self-service terminals must be simple operating instructions, high-quality presentation of product information and the opportunity, if required, to obtain advice from bank personnel.

There is a noticeable trend in the banking industry toward differentiation within the sales network. However, the cost of employing a specialist in every branch for each product in the growing product range would be prohibitive, so experts from larger branches offer specialized support to personnel in smaller branches.

Such an improvement would also enable customers at smaller branches to base their investment decisions on expert knowledge available only at the bank's larger branches.

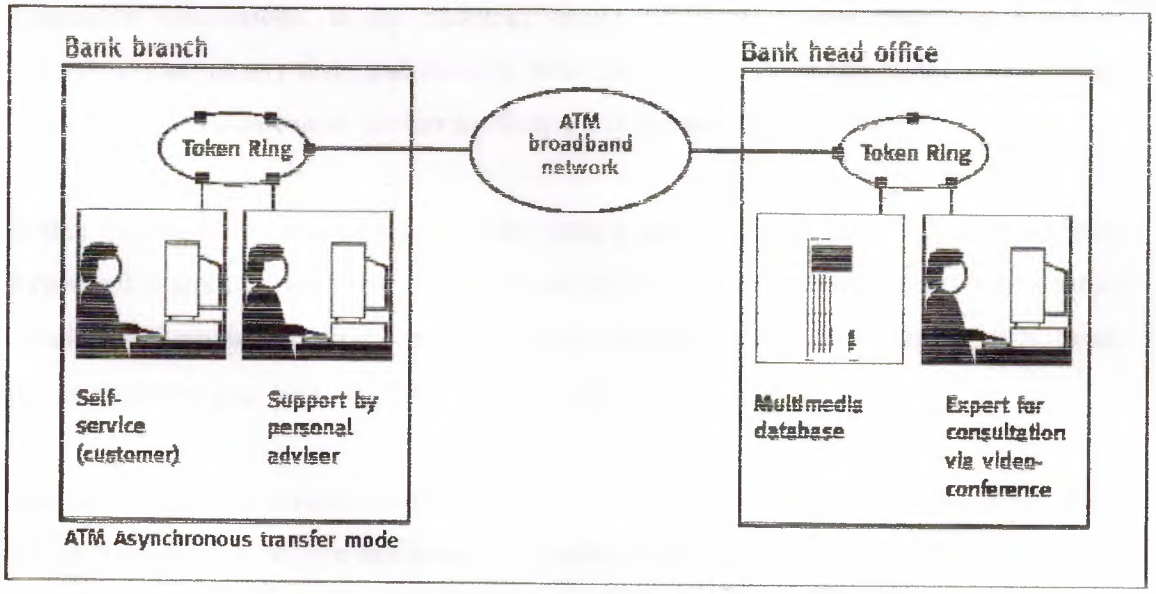

Figure 10 : Network Configurations for Connections Between Bank Branches and the Head Office.

In the BANK project, the two commercial scenarios outlined above were implemented on a prototype basis with specific applications.

In the first scenario a multimedia information terminal was created as a supplement to the automatic cash dispenser and statement printer. It consisted essentially of a workstation with a touch screen and equipment for audio and video recording and reproduction. The interest of a passing customer is attracted by a video film about the bank and its products.

By touching the screen the customer is given an overview of the institution's product spectrum and with the aid of the buttons shown on the screen he can select a subject area of interest. Information on the relevant subject is then conveyed by a varied multimedia product presentation, which contains videos, spoken explanations, texts, charts and still images.

Whatever the product, the customer is able to influence the course of the information sequence, for example by requesting more detailed information or carrying out illustrative calculations. If the customer needs more information than that which is offered he can, at any time and directly from the self-service mode, enter into a video-conference with a customer adviser to discuss any outstanding questions.

In this way customer and adviser can also jointly prepare documents such as application forms and charts for evaluating a portfolio or perform illustrative calculations. In a future development phase, facilities for concluding agreements (e.g. card reader, PIN input, printer, scanner, and fax) can also be integrated.

The two application scenarios can be installed in parallel at one bank. In this case there are multimedia self-service machines and workstations at the bank branch to support the investment consultant in his discussion with the customer; additional workstations are connected to a local network (token ring). An ATM broadband network links the branch to the bank's head office where there are experts who, by means of a desktop video-conference, can assist the customer with specialized advice.

A multimedia database contains up-to-date information on new products, which can be accessed if required.

Options were implemented for the interactive and non-interactive presentation of multimedia documents as well as shared working (joint pointing, joint viewing) for the purposes of a banking application. The functions of shared working were extended with an application-sharing mechanism for OS/2, DOS and Windows applications.

This creates a flexible basis for enabling the applications available at the bank employee's placeof work also to be used in consultations via video-conference if required. Multimedia and broadband technologies form the basis for innovative self-service equipment in the banking sector, offering new opportunities in the marketing of financial products and creating new sales channels for bank products.

In the BANK project a prototype multimedia system has been developed which can be used both as a self-service unit for bank products as well as an advisory support system for traditional customer service. Apart from pure product information, the system offers integrated desktop video-conference and the opportunity of processing documents jointly (application sharing). For data transfer between the workstations an ATM communications infrastructure was selected.

Multimedia documents with the option of video-conference and integrated application sharing can be used in other scenarios than those implemented. There are a number of other possible applications for such systems, e. g. in commercial education and in-service training. There are also potential applications in office support systems and co-ordination procedures within banks. These applications can overcome the restrictions of a particular location as well as providing enhanced communication with business partners. With sufficient availability of end-user equipment and inexpensive, suitable multimedia services within the public network a similar service can also be offered in the home banking sector.

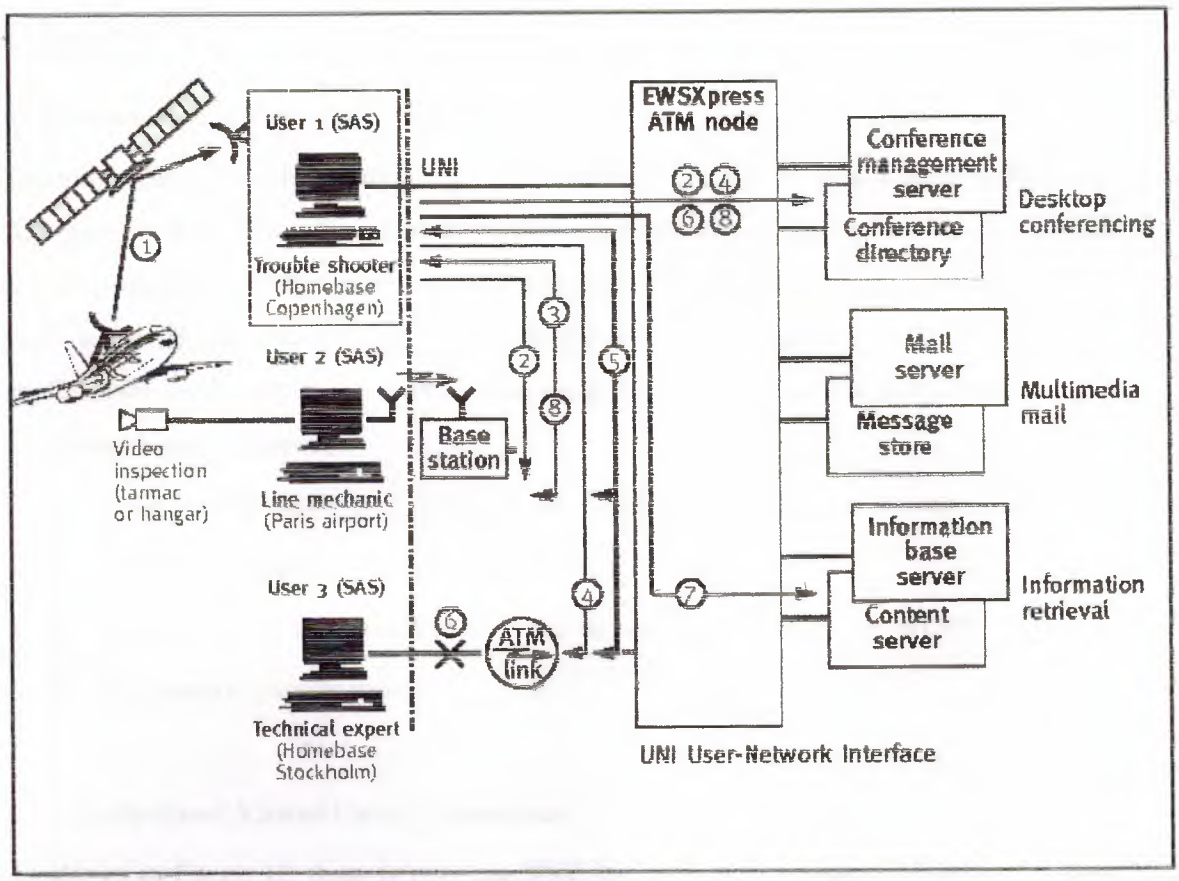

Figure 11 : CSCW Application Case.

5.13 CSCW Application Support By Future Multimedia Desktop Conferencing Teleservices.

The conference management server and the conference directory in figure 3.4 represent the essential functional groups, comprising some of the higher-layer teleservice functions such as association of partners, screening of access and distribution of output. Other advanced multimedia teleservices, such as multimedia mail, are also implemented in dedicated servers attached to network nodes. The lower-layer functions of the teleservice mainly refer to the user plane protocols of the ATM layer and AAL and the control plane signalling protocols up to Layer. The functions in the user plane deal with, for example, the support of multipoint communication configurations. The signaling protocols will enable the flexible handling of multiple parties, multiple information types and multiple connections, which are all perceived by the user as a single call.

Chapter VI

6. Protocol Design Principles

The aim of this report is to extend the QoS features of an ATM network to IP applications. Although, IP in its current form has no provision for QoS support, the underlying ATM subnet has the capability to offer performance guarantees. Therefore, for Internet applications to gain some of the benefits of ATM performance guarantees, without end-hosts or applications necessarily being aware of this capability. Thus, to achieve this parameter, there are several issues to consider.

1. What kind of traffic types? (real-time, connection-oriented, or connectionless.
2. Server-oriented resource reservation.
3. How to provide an environment of guaranteed services in IP over ATM?
4. Asymmetric transfer mode.

6.1 Priority-Based Virtual Circuit Connection

As shown in Figure 12, there is only one SVC between each host-pair in Classical IP over ATM. These SVCs provide "Best Effort" service. They did not guarantee any quality of service. To improve this, we provide a priority scheme to support QoS.

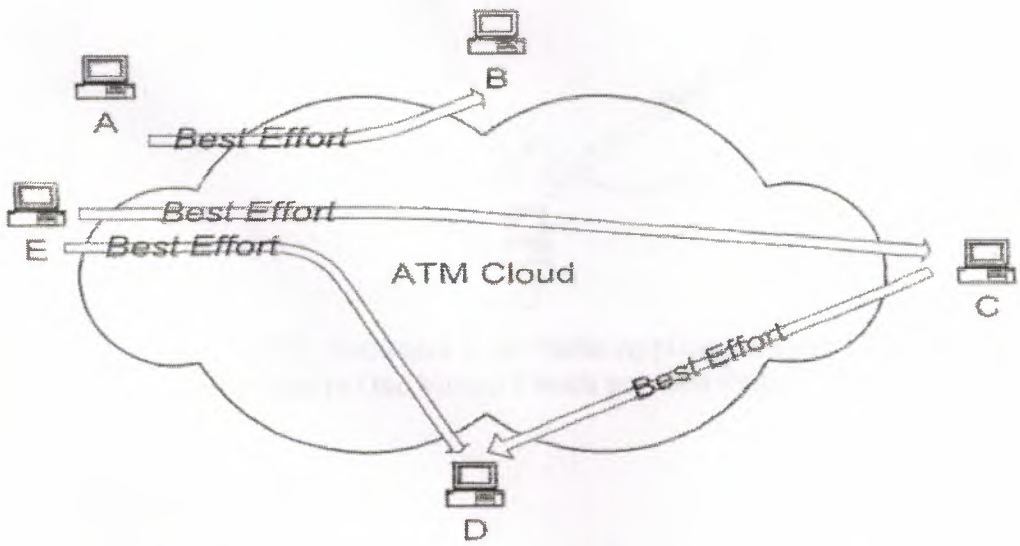

Figure 12 : Virtual Circuit Connection in Classical IP over ATM.

As we described in Section 6.2, different application types can be grouped into different virtual circuits. For example, different FTP sessions can be aggregated into one FTP virtual circuit and so does Telnets and HTTPs. Each virtual circuit is used by one application type. As shown in Figure 13, the same applications share the same virtual circuit because they have the same characteristic of resource requirements. For example, HTTPs care response time and Telnets focus on delay. All of these VCCs are Switched Virtual Circuits (SVC). These VCs are created on demand. On the contrary, real time traffic would likely have their own VC to transmit data, as shown in Figure 13, such that the other packets can't disturb them. Thus, we propose a priority scheme by using the precedence field in the Type of Service (TOS) field of IP header to determine whether a flow should initiate a new VC or join an existing one.

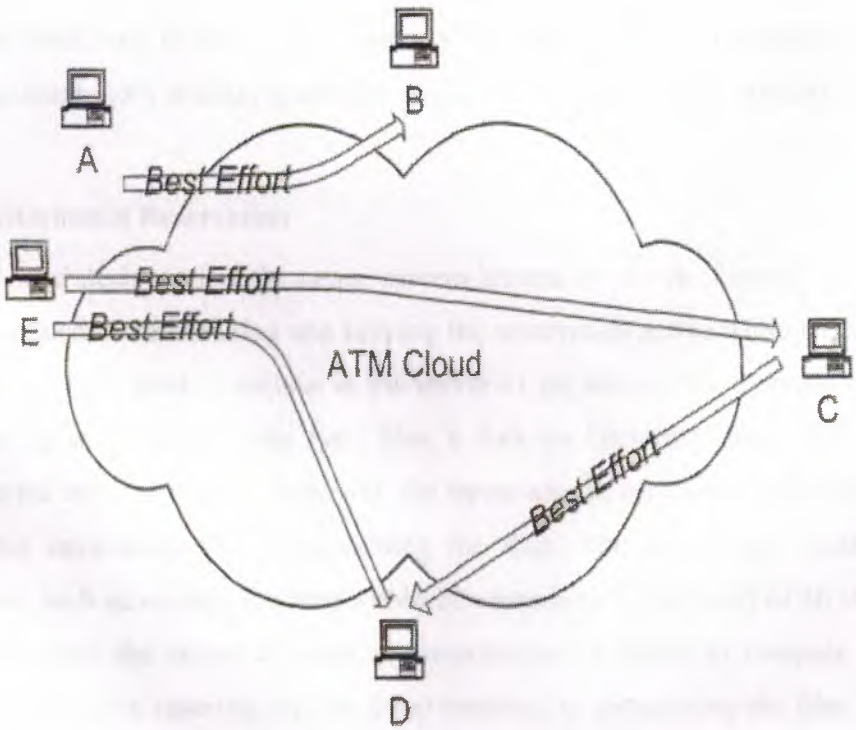

Figure 13 : Sessiones in the Same Application Type Aggregate to One Virtual Circuit per Host Pairs.

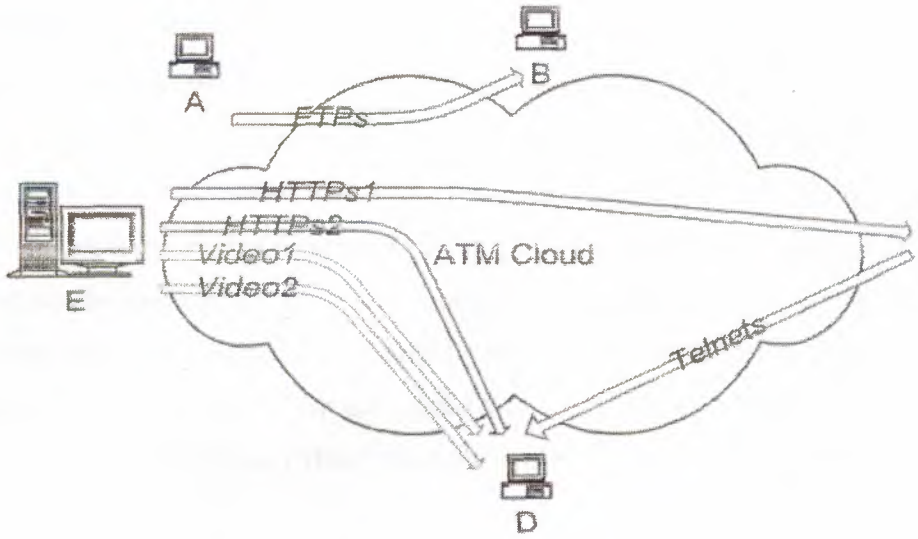

Figure 14 : Two Video Flows Have Their

In contrast, traditional IP over ATM networks uses only one SVC to transmit IP packets. There is no bandwidth or delay guarantee in classical IP over ATM networks.

6.2 Server-Oriented Reservation

Server-oriented design principle means servers choose the level of resources reservation and are responsible for initiating and keeping the reservation active while transmitting the data. When a client sends a request to the server to get service, for example, an ftp client requests an ftp server to transmits some files, it does not know how many data will arrive. Thus, after the server receives the request, the server has the responsibility for initiating and keeping the reservation while transmitting the data. The client only sends the QoS requirement, such as sending the files within 20 seconds or at the speed of 30 video frames per second. After the server receives the requirement, it needs to compute how much bandwidth need to be reserved and the delay variation in transmitting the files. This report recommends that server should have the responsibility to initialize and reserve the network resources.

6.3 Asymmetric Transfer Mode

On a client-server model the client often sends request to the server to get service. A request does not consume many resources, but the server will require many resources to transmit the data. Often servers need more resource reservations, such as ftp client sends a request to get files and ftp server returns the files it wants and so does HTTP. Almost all client-server applications have this similarity. According to the above discussion, different resource reservations are needed by the client and the server. That is so called "Asymmetric Transfer Mode" On the model that is used for this report, we recommend that requests be sent on "Best Effort" (BE) VC and data transmission use the QoS VC.

6.4 Traffic Types

ATM networks offer a specific set of service classes. At connection setup, the user must request the service class from the network for that connection. ATM networks use service classes to differentiate the types of connections, and each type of connections consists a particular mix of traffic and QoS parameter. Since these different traffic types may need to be differentiated within the network, priorities can be used to allow for the requested behavior. The set of QoS classes, defined by the ATM Forum UNI 3.x and UNI 4.0 is described in the following:

1. Continuous Bit Rate (CBR): End systems would use CBR connection types to carry constant bit rate traffic with a fixed timing relationship between data samples, typically for circuit emulation.
2. Variable Bit Rate—Real Time (VBR-RT): The VBR (RT) service class is used for connections that carry variable bit rate traffic, in which there is a fixed timing relationship between samples; for instance, video compression.
3. Variable Bit Rate—Non-Real Time (VBR-NRT): The VBR (NRT) service class is used for connections that carry variable bit rate traffic in which there is no timing relationship between data samples, but a guarantee of QoS (on bandwidth or latency) is still required. Such a service class might be used for Frame Relay internetworking, in which the Committed Information Rate (CIR) of the Frame Relay connection is mapped into a bandwidth guarantee within the ATM network.

4. Available Bit Rate (ABR): The ATM Forum is currently focusing on its work on the ABR service. As with the VBR (NRT) service, ABR supports variable rate data transmission and does not preserve any timing relationship between source and destination. Unlike the VBR (NRT) service, however, the ABR service does not provide any guaranteed bandwidth to the user. Rather, the network provides a "best effort" service, in which feedback (flow control mechanism) is used to increase the bandwidth available to the user—it is the Allowed Cell Rate (ACR)—if the network is not congested and to reduce the bandwidth when there is congestion.

ABR is designed to map to existing LAN protocols that opportunistically use as much bandwidth as is available from the network, but can either back off, or be buffered in the presence of congestion. The ABR service can optionally provides a guaranteed Minimum Cell Rate (MCR) for an ABR connection, but the exact nature of this guarantee is currently a matter of debate within the ATM Forum.

5. Unspecified Bit Rate (UBR): The UBR service does not offer any service guarantees. The user is free to send any amount of data up to a specified maximum while the network makes no guarantees at all on the cell loss rate, delay, or delay variation that might be experienced.

The following are the parameters for any type of connections:

1. Peak Cell Rate (PCR): determines how often data samples are sent.
2. Cell Delay Variation Tolerance (CDVT): determines how much jitter is tolerable.
3. Sustainable Cell Rate (SCR): determines the long-term average cell rate.
4. Burst Tolerance (BT): determines the size of the maximum burst of continuous cell rate.
5. Minimum Cell Rate (MCR), for ABR only.

These parameters define an "envelope" in a traffic stream, but not all of these parameters are valid for every kind of all service classes. For CBR connections, only the PCR and the CDVT are relevant. For VBR connections, the SCR and BT are used together to determine the long-term average cell rate and the size of the maximum burst of continuous cell rate that can be transmitted, hi the case of the ABR service, the PCR determines the maximum value of the Allowed Cell Rate (ACR), which is dynamically controlled by the network through congestion control mechanisms, to vary between the MCR and PCR.

When setting up a connection, the requesting node informs the network of the type of service required, the traffic parameters of the data flows in each direction of the connection, and the QoS requested for each direction. These descriptions form the traffic descriptors for the connection. In UNI 3.0/3.1, the QoS requested for each direction is not explicitly specified. Instead, the network offers a number of specified QoS classes that correspond to some or all of the QoS service types. The network administration has the responsibility of ensuring that the network is configured and each of the offered QoS classes provides levels of QoS appropriate for each QoS type. However, the ATM Forum decided that this method was too ambiguous and replaced it in UNI 4.0 with explicit signaling of QoS parameter. This parameter is the desired values requested at connection set-up time. After this modification, UNI 4.0 signaling messages will carry both the QoS service classes and the explicit parameters, and switches could operate on either, depending upon their own implementation, hi the proposed scheme, the server has to set the QoS parameters before initializing the VCs.

6.5 Resource Requirement

In classical IP networks, there are three types of traffic. They are connection-oriented, connection-less and real-time traffics. Connection-oriented traffic, such as HTTP, FTP, Telnet, NNTP and SMTP, currently dominate Internet traffic. Especially HTTP (HyperText Transfer Protocol), which is the basis for the World Wide Web (WWW). As shown in Table 2, statistics from the NSFNet backbone show the incredible growth in HTTP usage since January 1994. Connection-less (or UDP) traffic, such as SNMP and Name service, offers unreliable data transmission.

Therefore, it has no QoS problem. Real-time traffic, such as digital video and audio of multimedia applications, concerns the timing relationships between the data sampling. However, they generally run over UDP, which does not provide reliable data transport. The audio applications transmit the multimedia data at a constant bitrate, and the video applications send the variable bitrate compressed data. In this thesis, we focus on connection-oriented and real-time traffic. Connection-less traffic still uses the "best effort" service.

Table 2: Packets Count Percentages For Various Protocols on NSFnet Backbone.

Month	HTTP	NNTP	FTP (data)	Telnet	SMTP	DNS	#Packets*10^9
1994 Jan.	1.5%	8.8%	21.4%	15.4%	7.4%	5.8%	55
1994 Apr.	2.8	9.0	20.0	13.2	8.4	5.0	71
1994 Jul.	4.5	10.6	19.8	13.9	7.5	5.3	74
1994 Oct.	7.0	9.8	19.7	12.6	8.1	5.3	100
1995 Jan.	13.1	10.0	18.8	10.4	7.4	5.4	87
1995 Apr.	21.4	8.1	14.0	7.5	6.4	5.4	59

Different application types need different resource quota. In Table 3, we specify the traffic types for major Internet applications. Telnet /rlogin are typical interactive applications, and delay is their primary concern. Throughput on the data connection is likely concerned by ftp; HTTP is interested in response time, and video applications require sufficient bandwidth for their video streams. Different application types should be considered with different parameter sets.

Application	Description	Traffic Type
telnet	Remote login	Interactive
FTP	File transfer	Bulk transfer
HTTP	World Wide Web	Bulk transfer, somewhat interactive
audio	Digital audio	Continuous media (constant bit rate)
video	Digital video	Continuous media (variable bit rate)
SMTP	Electronic mail	Bulk transfer (background load only)
NNTP	Network news	Bulk transfer (background load only)

Table 3 : Internet Applications.

Chapter VII: Protocol Operations

7. Introduction

In this section, we introduce the proposed priority scheme. It uses the Type of Service (TOS) field in the IP datagram header and is backward compatible with existing IP implementation. These newer options need only be implemented on the end systems that want to take advantage of them.

7.1 Specification of the Type of Service Octet

As shown in Figure 15, the Precedence (named "User Priority" in this report) facility is one of the features of the Type of Service octet in the IP datagram header. The Type of Service octet consists of three fields.

0 1 2	3 4 5	6 7
Precedence	Type of Service (TOS)	MBZ
3-bits	4-bits	1-bits

MBZ: Must Be Zero.

Figure 15 : Type of Service in IP Datagram Header.

The first field, "Precedence", is intended to denote the importance or priority of the datagram. This field is not defined and used in current IP implementation. The applied scheme take this field as "User Priority" to determine the QoS types.

The 4 TOS bits are minimize delay, maximize throughput, maximize reliability, and minimize monetary cost, respectively. Table 4 shows the recommendation values of TOS .Only one of these 4 bits can be turned on. If all 4 bits are zero, it implies normal service, RFC 1340 specifies how these bits should be set by all the standard applications. RFC 1349 contains some corrections to RFC 1340, and a more detailed description of the TOS feature.

The last field, labeled "MBZ" (for "must be zero") above, is currently unused. The originator of a datagram sets this field to zero (unless participating in an Internet protocol experiment, which makes use of that bit). Routers and recipients of datagrams ignore the value of this field. This field is copied on fragmentation.

Table 4: Recommended Values for Type-of-Service Field

Aplication	Minimize delay	Maximize throughput	Maximize reliability	Minimize monetary cost	Hex value for TOS Octet
Telnet/Rlogin	1	0	0	0	0*10
FTP (control)	1	0	0	0	0*10
(data)	0	1	0	0	0*08
(any bulk data)	0	1	0	0	0*08
TFTP	1	0	0	0	0*10
SMTP (command phase)	1	0	0	0	0*10
(data phase)	0	1	0	0	0*08
DNS (UDP query)	1	0	0	0	0*10
(TCP query)	0	0	0	0	0*00
(zone transfer)	0	1	0	0	0*08
ICMP (error)	0	0	0	0	0*00
(query)	0	0	0	0	0*00
(any IGP)	0	0	1	0	0*04
SNMP	0	0	1	0	0*04
NNTP	0	0	0	1	0*02
BOOTP	0	0	0	0	0*00

7.2 Specification of the User Priority Octet

As stated above, this thesis defines the "User Priority" field as a three-bit field. This yields 8 different service classes with value 7 = highest priority and 0 = lowest priority. Table 5 defines the semantics of the User Priority field values. 0 is referred as the default User Priority.

Table 5: Recommendation Values of User Priority

Priority	Service	ATM QoS	Application
0	Best Effort (BE)	UBR	unspecified traffic
1	Bulk transfer (background)	ABR	NNTP, SMTP
2	Bulk transfer	ABR	FTP, HTTP
3	Interactive traffic	ABR	Telnet, HTTP
4	Internet control message	ABR	ICMP
5	Non-real-time(VBR)	nrtVBR	nrt digital video
6	Real-time (VBR)	rtVBR	digital video
7	Real-time (CBR)	CBR	digital video

7.3 Selecting User Priority Classes

One fundamental question is "who gets to decide what the classes mean and who gets access to them?" One approach would be for the meaning of the classes to be "well-known". The standardized set of classes is described in Table 5. It should be used in end stations implementing them. Class 6 and Class 7 services are for real-time traffics and have their own VC. Packets with Class 0 through 5 are sent on aggregate VC as described in Section 4.1.

The well-known port numbers distinguishes the separation between different applications. Most of the TCP or UDP applications have the property that they are assigned "well-known" ports. Assigning fixed port numbers to certain applications enables client processes to easily located server processes. For example, a telnet client application knows that it can locate telnet servers on remote hosts on TCP port 23. An ATM-attached router can check the source and destination port numbers of a TCP packet; if it sees a well-known port number in the TCP source port field, the packet is likely transmitted by a server process to a client process. Conversely, if a well-known port number appears in the TCP destination port field, the packet is likely transmitted by a client process to a server process.

7.4 Use of the User Priority Field in the Internet Protocol

For the User Priority facility to be useful, the User Priority field in IP packets must be filled in with reasonable values. When sending a datagram, the Internet Protocol uses the User Priority requested by the application. There is no requirement that both the client and server use the same User Priority. That is so called "Asymmetric Transfer Mode". For example, server sends packets on the QoS VC, but client uses the BE VC.

7.5 DataVC

When the end-station receives the request from the client, it needs to decide whether a flow should initiate a new VC or join an existing one. The end-station or router itself is responsible for local admission control and scheduling packets onto its link in accordance with the service agreed. Figure 16 shows the flow chart of these procedures.

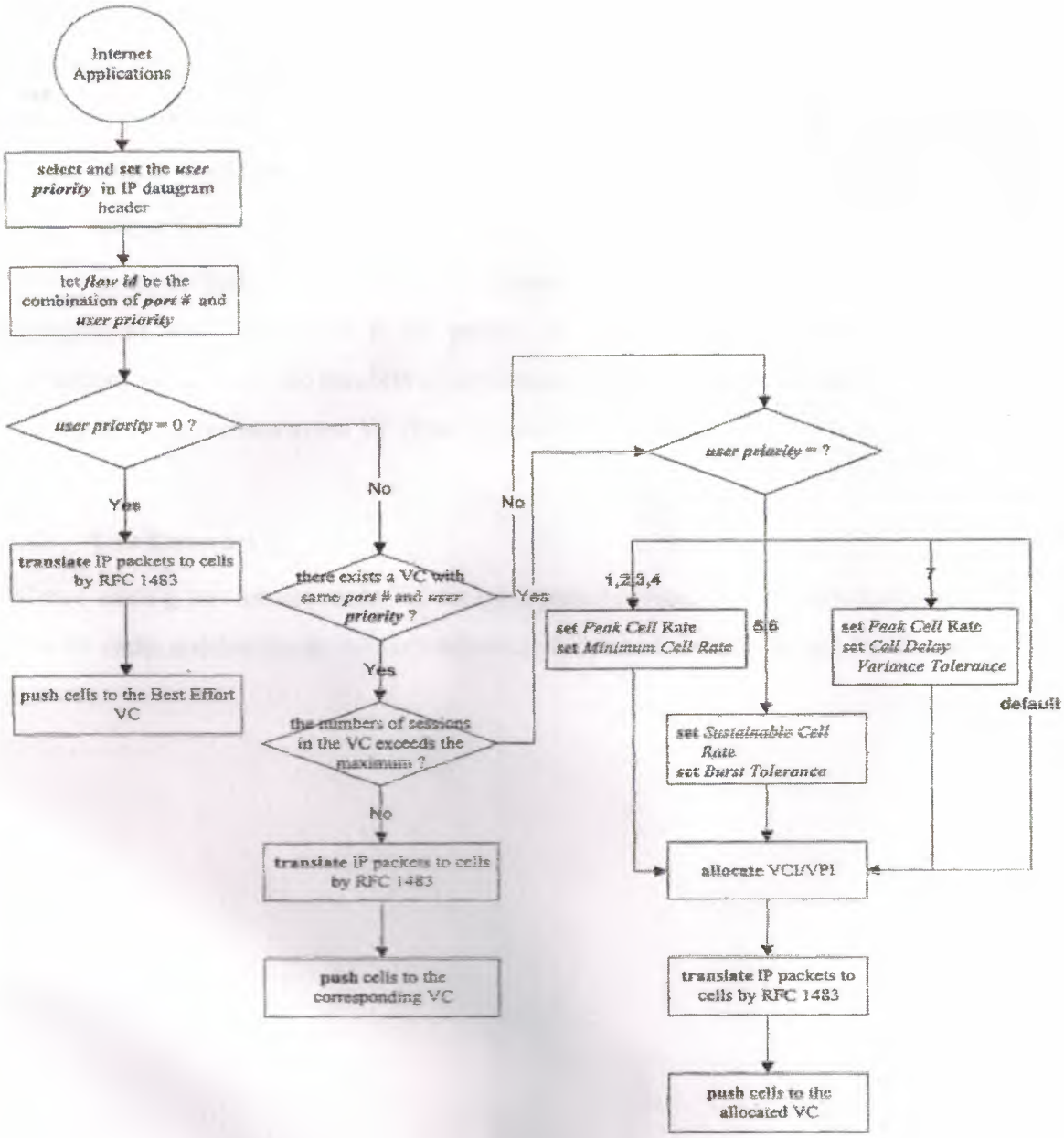

Figure 16 : How the QoS is Implemented in IP Over ATM

7.5.1 Establishing a VC

According to the "Server-oriented" principle, when the server finds that there is no VC connected to the client, it will create a new VC that corresponds to the appropriate User Priority value to send data.

7.5.2 Join a VC

When the server finds that there is a VC connected to the client, it will send data on the aggregate VC that corresponds to the priority and port number. The end station need a mechanism which limits the numbers of sessions on a given aggregate VC if the performance is going down and creates a new VC if the maximum is exceeded.

7.5.3 Tear Down a VC

When a sending host has not used a VC for some period of time, say t minutes, it clears the VC from the cache and notifies the network which can then tear down the VC and free the network resource.

Summary and Conclusions

In this report, a priority scheme (named "User Priority") is proposed by extending the IP datagram header. The aim is to enable the better packet delivery performance in traditional IP over ATM networks with QoS guarantees. Today's networks is mostly IP traffic. Then can be benefited by applying the method when passing through ATM networks.

Unlike RS VP, which provides dynamic quality of service in that the resources that are requested may change any time, the User Priority scheme does not support dynamic change QoS. There are several common reasons for a change of reservation QoS. First, an existing receiver can request a new larger QoS. Second, a sender may change it traffic specification, which can trigger a change in the reservation requests of the receivers. Finally, a new receiver can make a reservation that is larger than existing reservations. Since ATM service, as currently defined in UNI 3.x and UNI 4.0, does not allow renegotiating the QoS of a VC, dynamically changing the reservation means creating a new VC with the new QoS, and tearing down an established VC. Tearing down a VC and setting up a new VC in ATM are time-consuming.

Besides, we need an enhanced signaling protocol. Set up a connection is a hop-by-hop process in UNI 3.x and 4.0. A possible candidate is the Connection Request Protocol (CRP). It uses a parallel connection set up and resource management scheme. Its key feature is that it combines address resolution with connection set up to improve performance. Further, it eliminates the need for IP end points to support ATM signaling protocols, thereby significantly simplifying their configuration and management.

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