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Abstract

ATM technology has become inextricably linked to promises of a global communications infrastructure. The expectation is that seamless integration of voice data and video across high-speed ATM links will provide universal access to multimedia services.

ATM equipment developers, service providers, and end users are all faced with these same challenges when testing ATM products. ATM technology is now in a period of dynamic growth. Standards groups (i.e., the International Telecommunications Union [ITU]) are challenged to keep up with this fast-paced development and their publications do not always correspond to specifications issued by other organizations

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Chapter I: INTRODUCTION TO ATM NETWROKS

1.1 Overview of ATM

Unlike traditional data networks, future broadband-ISDN (BISDN) wide-area networks will be required to carry a broad range of traffic classes ranging from bursty, variable-rate sources, such as voice and variable-rate coded video, to smooth, constant bit rate sources. Moreover, these networks will have to do so while providing a guaranteed performance or quality-of-service (QOS) to these traffic classes. The problem of characterizing performance in such networks is thus particularly important since this must be done not only for the traditional off-line tasks of dimensioning and design (e.g., determining link bandwidths, buffer capacities and processing capacities at network switchpoints) but also for on-line, *performance-driven* traffic control purposes such as session-level admission control. In this section, we outline our research aimed at providing the analytical tools and techniques for analyzing such networks and their diverse workloads and for designing and characterizing the properties of scheduling policies in these networks.

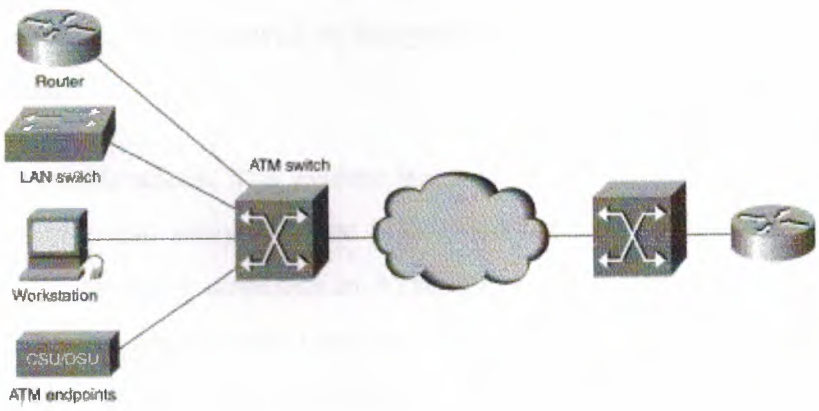

Figure 1.1 ATM simple network

Traditional approaches towards performance evaluation of communication networks are generally not applicable in a high-speed network environment for a number of reasons. First, the traffic in future high-speed networks is projected to be ``bursty," often processing complex correlations in the cell arrival processes.

Simple Markovian assumptions (such as memoryless inter-arrival times) are thus likely to be invalid in such networks. The gigabit rates of future high-speed networks also render simulation ineffective for systems of any realistic size. The very performance metric(s) of interest in a high-speed network will also be different average delay will no longer be the primary performance measure, and, instead, *secondary performance measures* such as probability of buffer overflow, maximum packet delay, and the tail of delay distributions must be considered. In this respect, there is evidence that even relatively sophisticated performance models which work well for predicting average delay in the presence of correlated arrivals may be less well-suited for computing these new performance measures of interest.

In many previous performance evaluation techniques have been confined to studying the performance of the *aggregate traffic* generated by a set of *identical* sources at a *single* multiplexer in isolation; in connection-oriented high-speed networks with QoS guarantees, it is clear that performance must be examined on a *per-session* basis and in a *network* setting. Finally, we note that given the potentially complex nature of network traffic, it is becoming increasingly valuable to be able to broadly characterize properties (such as optimality or near optimality) of network control mechanisms which hold over a wide range of traffic models and assumptions. However, little work has been reported on this problem for traditional networks, much less for high-speed networks.

Given the above considerations, it is evident that new performance evaluation techniques are required for the design and analysis future high-speed networks. Furthermore, given the QoS requirements of such emerging standards as ATM, these techniques will also be needed for *on-line* performance driven, traffic control purposes such as session-level admission control. Our research in these areas is aimed at addressing this need. Our research divides broadly into four areas:

1. We are developing a methodology for obtaining tight pessimistic bounds on the per-session performance of a collection of heterogeneous sessions within a high-speed network.

The metrics of interest are *the distribution of the packet delay and the probability of buffer overflow.* Our research is unique in that we are interested in computing provable performance *bounds,* and in doing so on a *per-session* basis in a very general *network* setting in which sessions may traverse a number of hops. This methodology is useful, not only for the purpose of analyzing session- and network-level performance, but also as a mechanism to be used for call admission where performance (QoS) guarantees are to be provided.

2. We conjecture that the bounds computed using the above methodology may sometimes be too loose to be of practical interest. In these cases, it will be of interest to compute performance. We have thus also examined a number of approximate approaches towards evaluating the performance of heterogeneous sessions within a high-speed network. The primary performance metric of interest is probability of buffer overflow. Our research here is noteworthy in considering heterogeneous sources and in doing so in a network setting.
3. We have developed a methodology for designing and analyzing the qualitative behavior of scheduling policies. This is based on sample path analysis and the theory of majorization. Our focus is on a wide class of performance metrics, including the average and maximum packet delay, the probability of buffer overflow, and the length of a buffer overflow burst. As discussed above, this work is of particular interest and value in that it can characterize properties of scheduling mechanisms (such as their optimality, or near optimality) over a wide range of assumptions, thus obviating the need for a potentially time-consuming and/or difficult case-by-case performance analysis.
4. We are also developing *routing policies* for high-speed networks. Given the need to provide QoS guarantees to admitted sessions (and thus implicitly ``reserve" resources for on-going calls), the routing problem in high-speed networks shares much more in common with circuit-switched routing algorithms than with routing in traditional data communication networks is aimed at exploiting the similarities with routing in traditional circuit-switched networks and adapting these policies for the case of high-speed ATM networks. Of particular interest is the fact that ATM routing will be more processing intensive, and will be required to route traffic with heterogeneous bandwidth requirements.

1.2 What is ATM

ATM was developed because of developing trends in the networking field. The most important parameter is the emergence of a large number of communication services with different, sometimes yet unknown requirements. In this information age, customers are requesting an ever increasing number of new services. The most famous communication services to appear in the future are HDTV(High Definition TV), video conferencing, high speed data transfer, videophony, video library, home education and video on demand.

This large span of requirements introduces the need for one universal network which is flexible enough to provide all of these services in the same way. Two other parameters are the fast evolution of the semi - conductor and optical technology and the evolution in system concept ideas - the shift of superfluous transport functions to the edge of the network.Both the need for a flexible network and the progress in technology and system concepts led to the definition of the Asynchronous Transfer Mode (ATM) principle.

Today's telecommunication networks are characterized by specialization. This means that for every individual telecommunication service at least one network exists that transports this service. Each of these networks was specially designed for that specific service and is often not at all applicable to transporting another service. When designing the network of the future, one must take into account all possible existing and future services.

The networks of today are very specialized and suffer from a large number of disadvantages:

1. Service Dependence: Each network is only capable of transporting one specific service .
2. Inflexibility: Advances in audio, video and speech coding and compression algorithms and progress in VLSI technology influence the bit rate generated by a certain service and thus change the service requirements for the network. In the future new services with unknown requirements will appear. A specialized network has great difficulties in adapting to new services requirements.
3. Inefficiency: The internal available resources are used inefficiently. Resources which are available in one network cannot be made available to other networks.

It is very important that in the future only a single network will exist and that this network is service independent. This implies a single network capable of transporting all services, sharing all its available resources between the different services. It will have the following advantages: Flexible and future safe. A network capable of transporting all types of services that will be able to adapt itself to new needs.

4. Efficient in the use of its available resources All available resources can be shared between all services, such that an optimal statistical sharing of the resources can be obtained.
5. Less expensive: Since only one network needs to be designed, manufactured and maintained the overall costs of the design, manufacturing, operations and maintenance will be lower.

1.3 ATM Technology

The definition of a service independent network has been influenced by an evolution in technology and system concepts. The ideal network in the future must be flexible. The most flexible network in terms of bandwidth requirements and the most efficient in terms of resource usage, is a network based on the concept of packet switching. Any bandwidth can be transported over a packet switching network and the resources are only used when useful information has to be transported.

The basic idea behind the concept changes is the fact that functions must not be repeated in the network several times if the required service can still be guaranteed when these functions are only implemented at the boundary of the network. Progress In Technology In recent years large progress has occurred both in field of electronics
and in the field of optics.

Broadband communication systems can be developed based on different technologies, the most promising being CMOS. (Complementary Metal Oxide Semiconductor) CMOS allows high complexity and reasonably high speed (up to 200 to 300 Mbits/s). The low power dissipation of CMOS is particularly important, and allows the realization of high complexity, high speed systems on a very small chip surface. With the increased complexity per chip, the system cost can easily be reduced since the large integration will continuously allow the volume of the system to shrink or to increase the functionality at a constant cost.

Optical technology is also evolving quite rapidly. Optical fiber has been installed for transmission services for several years.

1.4 ATM Requirements Performance

In the future broadband network a large number of services have to be supported. These services are:

1. Low speed like telemetry, low speed data ,telefax,..etc.
2. Medium speed like hifi sounds, video telephony, high speed data,..etc.
3. Very high speed like high quality video, video library ...etc.

A single typical service description does not exist. All services have different characteristics both for their average bit rate and burstiness. To anticipate future unknown services we must try to characterize as general a service as possible.

The optimal transfer mode should support the communication of various types of information via an integrated access. Ideally the transfer mode must provide the capability to transport information, whatever type of information is given at the network, very much like the electricity network, which provides power to it's customers without regarding the way the customer uses his electricity.

Two other important factors are:

1. Semantic transparency determines the possibility of network to transport the information error free. The number of end to end errors introduced by the network is acceptable for the service. No system is perfect. Most of the imperfections of telecommunication systems are caused by noise.

2. Other factors contribute to a reduced quality: limited resources causing blocking; any system errors. One of the most important parameters used to characterize imperfections is the BER (bit error rate) - the ratio between erroneous bits and transmitted bits.
3. Time transparency determines the capability of the network to transport the information through the network from source to destination in a minimal time acceptable for the service. Time transparency can be defined as the absence of delay and delay jitter(different part of the information arrive at the destination with different delay). The value of end to end delay is an important parameter for real time services, such as voice and video. If the delay becomes too large echo problems may arise in a voice connection.

1.5 Basic Principles of ATM

ATM is considered a packet oriented transfer mode based on:

1. Asynchronous time division multiplexing
2. The use of fixed length cells

An ATM cell structure is displayed in the following figure:

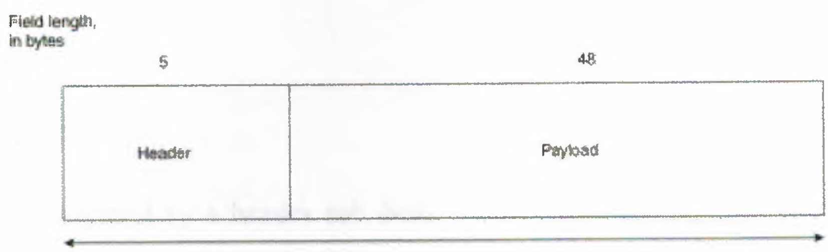

Figure 1.2 ATM cell structure

Each cell consist of an information field and a header. The header is used to identify cells belonging to the same virtual channel and to perform the appropriate routing. To guarantee a fast processing in the network, the ATM header has very limited function. Its main function is the identification of the virtual connection by an identifier which is selected at call set up and guarantees a proper routing of each packet. In addition it allows an easy multiplexing of different virtual connections over a single link.

The information field length is relatively small, in order to reduce the internal buffers in the switching node, and to limit the queuing delays in those buffers - small buffers guarantee a small delay and a small delay jitter as required in real time systems.The information field of ATM cells is carried transparently through the network. No processing is performed on it inside the network. All services (voice, video, data) can be transported via ATM , including connectionless services.

1.5.1 Routing

ATM is connection oriented. Before information is transferred from the terminal to the network, a logical/virtual connection is set. The header values are assigned to each section of a connection for the complete duration of the connection, and translated when switched from one section to another. Signalling and user information are carried on separate virtual channels Two sorts of connections are possible:

1. Virtual Channel Connections VCC
2. Virtual Path Connections VPC

When switching or multiplexing on cells is to be performed, it must first be done on VPC, then on the VCC.

1. Virtual Channels

This function is performed by a header sub field - VCI. Since the ATM network is connection oriented each connection is characterized by a VCI which is assigned at call set up. A VCI has only a local significance on the link between ATM nodes and will be translated in the ATM nodes.

When the connection is released, the VCI values on the involved links will be released and can be reused by other connections.An advantage of this VCI principle is the use of multiple VCI values for multicomponent services. For instance video telephony can be composed of 3 components: voice, video and data each of which will be transported over a separate VCI. This allows the network to add or remove components during the connection.

For instance, the video telephony service can start with voice only and the video can be added later as shown in figure 1.3.

Figure 1.3 VCs concentrate to create VPs

The network has to support semi-permanent connections, which have to transport a large number of simultaneous connections. This concept is known as virtual path.

All ATM switches can be schematically described as follows. A number of incoming links transport ATM information to the switch, where depending on the value of the header this information is switched to outgoing link. The incoming header and the incoming link number are used to access a translation table. The result of the access to the table is an outgoing link and a new header value.

1.5.2 Resources

As ATM is connection oriented, connections are established either semi-permanently, or for the duration of a call, in case of switched services.This establishment includes the allocation of a VCI (Virtual Channel Identifier)and/or VPI (Virtual Path Identifier), and also the allocation of the required resources on the user access and inside the network. These resources are expressed in terms of throughput and Quality of Service.

They may be negotiated between user and network for switched that previous connection during the call set up phase.

1.5.3 ATM Cell Identifiers

ATM cell identifiers are:

1. Virtual Path Identifier
2. Virtual Channel Identifiers
3. Payload Type Identifiers

They support recognition of an ATM cell on a physical transmission medium. Recognition of a cell is a basis for all further operations.VPI and VCI are unique for cells belonging to the same virtual connection on a shared transmission medium. As such they are limited resources. Within a particular virtual circuit, cells may be further distinguished by their PTI, which cannot be allocated freely but depends on the type of payload carried by the cell.

This field indicates whether the cell is carrying user information to be delivered transparently through the network or special network information.In case the field indicates network information, part of the information field indicates the type of network control whereas the remaining part of information field may be processed inside the network.

1. Throughput

   Bandwidth has to be reserved in the network for each virtual connection. ATM offers the possibility to realize resources saving in the total bandwidth needed when multiplexing traffic of many variable Bit Rate connections.The amount which can be saved depends heavily on the number of multiplexed connections, on the burstiness of the traffic they carry, on the correlation between them and on the quality of service they require.

2. Quality of Service

   The quality of service of a connection relates to the cell loss, the delay and the delay variation incurred by the cells belonging to that connection in an ATM network. For ATM, the quality of service of a connection is closely linked to the bandwidth it uses. When providing limited physical resources using more bandwidth increases the cell loss, the delay, and the delay variation incurred, decreases the QoS for cells of all connections which share those resources.

3. Usage Parameter Control

In ATM there is no physical limitation on the user access rate to the physical transmission medium, apart from the physical cell rate on the medium itself. Multiplexing equipment will do its utmost to avoid cell loss to offer the highest possible throughput whatever the user chooses to send.As virtual connections share physical resources, transmission media and buffer space, unforeseen excessive occupation of resources by one user may impair traffic for other users. Throughput must be monitored at the user - network interface by a Usage Parameter Control function in the network to ensure that a negotiated contract per VCC or VPC between network and subscriber is respected.

It is very important that the traffic parameters which are selected for this purpose can be monitored in real time at the arrival of each cell.

4. Flow Control

In principle, no flow control will be applied to information streams at the ATM layer of the network. In some cases it will be necessary to be able to control the flow of traffic on ATM connections from a terminal to the network. In order to cope with this a GFC (general flow control) mechanism may be used. This function is supported by a specific field in the ATM cell header. Two sets of procedure are used: Uncontrolled Transmission - for the use of point to point configuration. Controlled Transmission - can be used in both point to point and shared medium configuration. Another principle is no error protection on link by link basis.

If a link in the connection, either the user to network link or the internal links between the network nodes, introduces an error during the transmission or is temporarily overloaded thereby causing the loss of packets, no special action will be taken on that link to correct this error (= no requesting for retransmission).This error protection can be omitted since the links in the network have a very high quality

5. Signalling

   The negotiation between the user and the network with respect to the resources is performed over a separate signalling virtual channel. The signalling protocol to be used over the signalling virtual channel is an enhancement of those used in ISDN signalling.

1.6 ATM: The Layered Model

The OSI model is very famous and used to model all sorts of communication systems. We can model the ATM with the same hierarchical architecture, however only the lower layers are used. The following relations can be found:

The Physical layer is more or less equivalent to Layer 1 of OSI model, and mainly perform functions on the bit level. The ATM layer can be located mainly at the lower edge of the layer 2 of the OSI model. The adaptation layer performs the adaptation of higher layer protocols, be it signalling or user information, to the fixed ATM cells.

These layers can then further be divided into sublayers. Each sublayer performs a number of functions, to be explained in the following sections.

1. Layer Sublayer AAL
2. Adaptation layer CS
3. SAR ATM layer and Physical Layer TC
4. PM - Physical Medium Sublayer

This sublayer is responsible for the correct transmission and reception of bits on the appropriate physical medium. At the lowest level the functions that are performed are medium dependent. In addition this sublayer must guarantee a proper bit timing reconstruction at the receiver. Therefore the transmitting peer will be responsible for the insertion of the required bit timing information and line coding.

Transmission Convergence Sublayer.

In this sublayer bits are already recognized, as they come from the PM sublayer. This sublayer performs the following functions dependently

1. Adaptation to the transmission system used
2. Generation of the HEC (Header Error Check) of each cell at the transmitter, and its verification at the receiver
3. Cell delineation: The mechanism to perform cell delineation is based on the HEC. If a correct HEC is recognized for a number of consecutive cells it is assumed that the correct cell boundary is found. To avoid erroneous cell delineation on user information, the information field of each cell is scrambled at the transmitting side and descrambled at the receiving side. This ensures that the probability of finding a correct HEC in the information field is very low
4. Once the cell delineation has been found an adaptive mechanism uses the HEC for correction or detection of cell header errors. Isolated single bit errors are corrected.
5. Cell uncoupling: The sublayer ensures insertion and suppression of unassigned cells to adapt the useful rate to the available payload of the transmission system ATM Layer.

The following main functions are performed by the layer:

1) The multiplexing and demultiplexing of cells of different connections into a single cell stream
2) A translation of cell identifiers, which is required in most cases when switching a cell from one physical link to another in an ATM switch or cross connect. This translation can be performed either on the VCI or VPI separately, or on both simultaneously.
3) Providing the user of a VCC or VPC with one QOS class out of a number of Classes supported by the network.

4) MANAGEMENT FUNCTIONS: the header of user information cells provides for a congestion indication and an ATM user to ATM user indication.
5) Extraction (addition) of the cell header before (after) the cell is being delivered to (from) the adaptation layer
6) Implementation of flow control mechanism on the user network interface.

1.6.1 ATM Adaptation Layer

This layer enhances the service provided by the ATM layer to a level required by the next higher layer. It performs the functions for the user, control and management planes and supports the mapping between the ATM layer and the next higher layer. The functions performed in the AAL depend on the higher layer requirements.

The AAL layer is divided into two sublayers:

1. SAR the segmentation and reassembly sublayer: The main purpose of the SAR sublayer is segmentation of higher layer information into a size suitable for the payload of the consecutive ATM cells of a virtual connection, and the inverse operation, reassembly of contents of the cells of a virtual connection into data units to be delivered to the higher layer.

2. CS - the convergence sublayer: This sublayer performs functions like message identification, time/clock recovery etc. AAL Service Data Units (SDU) are transported from one AAL Service Access Point to one or more access points through the ATM network. The AAL users will have the capability to select a given AAL - SAP associated with the QOS required to transport the SDU.

1.6.2 The Classes of ATM Services

The services which will be transported over the ATM layer are classified in four classes, each of which has its own specific requirements towards the AAL. The services are classified according to three basic parameters:

1. Time relation between source and destination: For real time applications like phone conversation, a time relation is required. Information transfer between computers does not require a time relation.
2. Bit Rate: Some services have a constant bit rate, others have a variable bit rate.
3. Connection mode: connectionless or connection oriented Four types of AAL protocols have been recommended up to now : AAL 1,AAL 2, AAL 3/4, AAL

1. **AAL 1 - Adaptation for constant bit rate services**

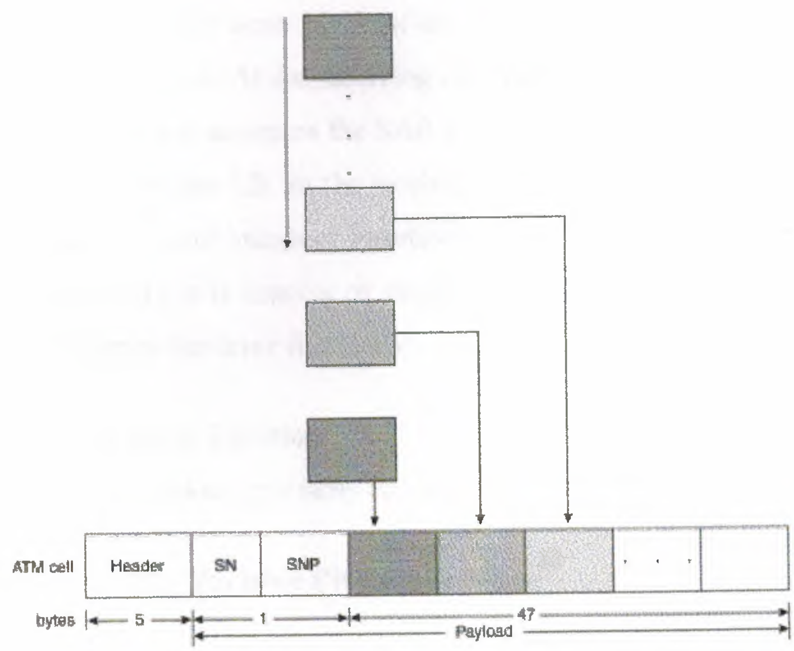

Figure 1.4 AAL1 Prepares a Cell for Transmission So That the Cells Retain Their Order

Recommended for services such as digital voice and digital video. It is used forapplications that are sensitive for both cell loss and delay. Constant Bit Rate (CBR) services require information to be transferred between source and destination at a constant bit rate after virtual connection has been set up. The Layer services provided by the AAL 1 to the user are: Transfer of Service Data Units with a constant bit rate and their delivery with the same bit rate

1) Transfer of data structure information
2) Transfer of timing information between source and destination
3) Indication of lost or corrupted information which is not recovered by the AAL itself when needed.

   CSI - CS indication - 1 bit.

   SN - sequence Number - 3 bits.

   SNP - sequence number protection 4 bits.

The SAR sublayer accepts a 47 octet block of data from the CS and then adds a one octet SAR-PDU header to each block. At the receiving end, the SAR sublayer gets a 48 byte block from the ATM layer, and then separates the SAR PDU header. The SAR sublayer receives a sequence number value from the CS. At the receiving end this number is passed to the CS. IT may be used to detect loss and incorrect insertions of SAR payloads. The SNP is used for protection against bit errors. It is capable of single bit error correction and multiple bit error detection. The convergence Sublayer functions:

1) Handling of cell delay Variation
2) Handling of cell payload assembly delay

2. AAL 2 - Adaptation for Variable Bit Rate Services

These type AAL offers a transfer of information with a variable bit rate. In addition, timing information is transferred between source and destination. Since the source is generating a variable bit rate, it is possible that cells are not completely filled and that the filling level varies from cell to cell. Therefore more functions are required from the SAR .

1) The SN field (Sequence Number) contains the sequence number to allow the recovery of lost or misrouted cells.

2) The IT (Information Type) indicates the beginning of a message (BOM), continuing of a message (COM),the end of a message(EOM) or that the cell transports timing or other information . BOM,COM or EOM indicate that the cell is the first, middle or last cell of a message, i.e. an information unit as defined in the CS layer with possibly a variable length.

3) The LI (length indicator) field indicates the number of useful bytes in partially filled cells.

4) The CRC field allows SAR to detect bit errors in the SAR SDU

In the CS sublayer the following functions have to be performed:

Clock recovery by means of insertion and extraction of time information.

1) Handling of lost or incorrectly delivered cells.
2) Forward error correction for video and audio services.

3. **AAL ¾:- Adaptation for Data Services**

This AAL is recommended for transfer of data which is sensitive to loss, but not to delay. The AAL may be used for connection oriented as well as for connectionless services, since functions like routing and network addressing are performed on the network layer.

Two modes of AAL 3/4 are defined:

1) Message Mode: The AAL SDU is passed across the AAL interface in exactly one AAL Interface Data Unit (IDU) . This service is provided for the transport of fixed or variable length AAL SDU.

2) Streaming mode: The AAL SDU is passed in one or more AAL IDU. ransfer of these IDUs may occur separate in time. The service provided for long variable length AAL SDUs. The <u>SAR</u> sublayer functions:

1) Segmentation and reassembly of variable length CS PDUs. The SAR PDU contains two fields for this purpose:

   1. ST Segment Type: Indicates which part of the CS PDU is carried by the SAR PDU.

   2. LI Length Indicator

   * Error Detection – using CRC field

   * Multiplexing of multiple CS PDUs on a common bearer in the ATM layer.

   Multiplexing is supported by a multiplexing identifier. The CS functions are

1) Delineation and transparency of SDUs.
2) Error detection and handling: Corrupted SDUs are either discarded or optionally delivered to the service specific convergence sublayer.
3) Buffer allocate size- each PDU carries up front an indication to the receiving entity of the maximum buffer required to receive the PDU.
4) Abort - a partially transmitted PDU can be aborted.

4. AAL 5 - Adaptation for Data Services

This AAL is recommended for high speed connection oriented data service. This AAL offers a service with less overhead and better error detection.

1) The SAR sublayer functions: The SAR sublayer accepts variable length SAR SDUs which are multiples of 48 octets from the CS sublayer, and generates SAR PDUs containing 48 octets of data.
2) The CS functions: The functions implemented by the AAL5 are the same as the ones offered by the AAL 3/4 except that the AAL 5 does not give a buffer allocation size indication to the receiving peer entity. Also error protection in the AAL 5 is fully handled at the CS layer itself, instead of being shared between SAR and CS.

Chapter II: ATM ARCHITECTURE

2.1 How ATM Works

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 witching techniques. As in figure 2.1.

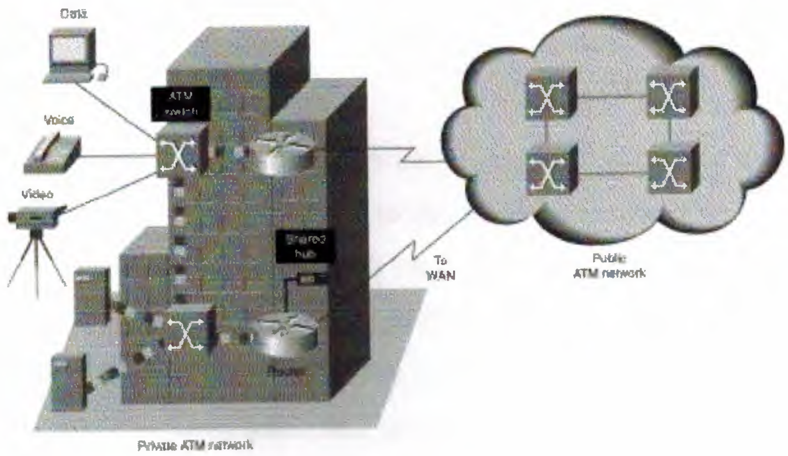

Figure 2.1 ATM network

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.

2.2 ATM Protocol

The protocol is divided into three layers:

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

2.3 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:

1. **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.

2. **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. **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.

4. **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.

2.4 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 layout of an ATM cell is shown below :

Table 2.1 Layout of ATM cell

Generic Flow Control 4 bits	Virtual Path Identifier 8 bits	Virtual Channel Identifier 16 bits	Payload Type 3 bits	Cell Loss Priority 1 bit	Header Error Control 8 bits
Cell Payload 48 bytes					

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 2.2 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.

2.4.1 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.

2.4.2 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 2.2.

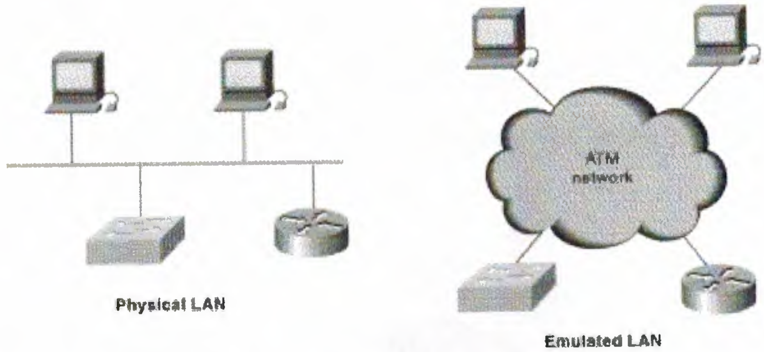

Figure 2.2 ATM Emulate the physical LAN

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.

2.5 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 2.3.

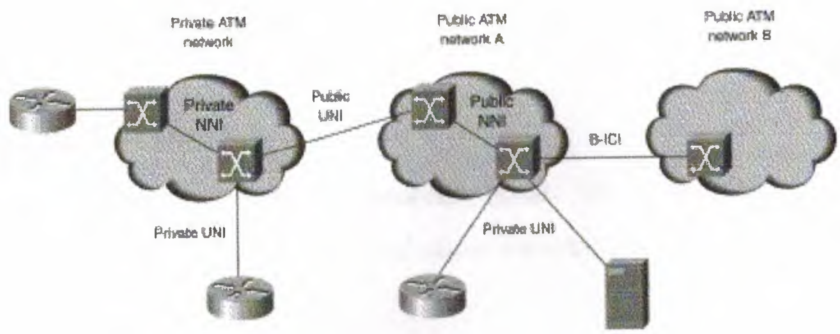

Figure 2.3 ATM interface specifications

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 figure 2.3). 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.

2.5.1 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.

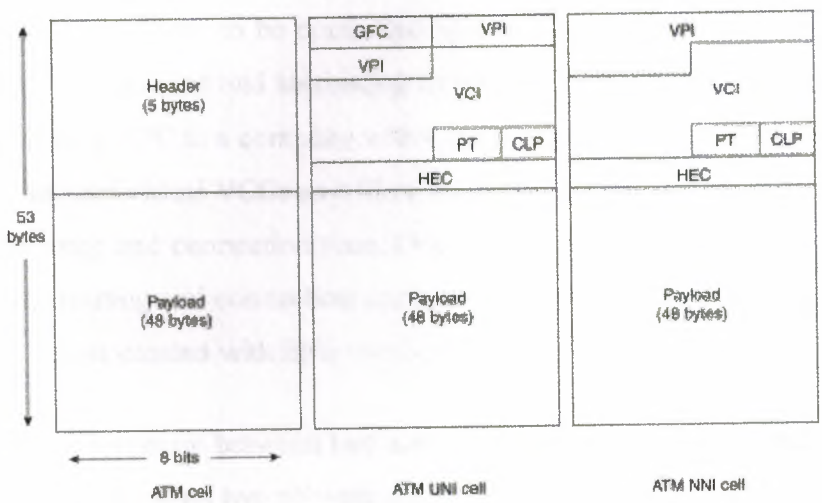

Figure 2.4 ATM packet format

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.

2.5.2 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.

2.5.3 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)

1. Supplies a constant data rate and tight upper bound on packet delay.
2. Used for videoconferencing, telephony, uncompressed audio/video distribution.

3. Real-time variable bit rate (rt-VBR)

1. For bursty sources, such as compressed video, that still have tight time constraints.
2. 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)

1) 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.
2) 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)

1) The lowest class of service; uses excess (whatever is left over) capacity.
2) For file transfer, http, e-mail: typical Internet type uses.

2.5.4 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.

2.5.5 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.

2.5.6 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 III:

3.1 Overview

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.

3.2 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.

3.3 Motivation and Goal

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.

3.4 CONGESTION PROBLEM

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.

3.4.1 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 [1-6]. 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 [1,3].

i) 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.

ii) Rate-based approach: This approach is a closed

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.

A. 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

B. 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).

C. Backward Explicit Congestion Notification (BECN)

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.

3.5 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.

3.5.1 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.

1. 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.

2. 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.

3. 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.

3.5.2 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.

The Explicit Call Rate for any call, is calculated by the switch as follows:

Load factor = 1 - bandwidth used by the source/total bandwidth of the link (3.1)

Explicit call rate = load factor* current call rate (3.2)

Thus, the sources will reduce their current call rate proportionately to their actual load. When a source receives control cells, it will note down the current time and reduce the current call rate as shown above. The recovery will resume to its original rate after a period called recovery period. The recovery period is also based on source's peak call rate.

Recovery period = peak call rate*100 (micro secs) (3.3)

The recovery period is based on the peak call rate. The higher the call rate, recovery period is higher. This is because, the sources whose call rate is low if recovers faster, will not increase the network load as compared to the sources with the higher call rate. If the recovery period of the sources having high call rate is high, then it gives enough time to the network to stabilize. Once the source reduces to its speed, all the control cells received by it in the recovery period are ignored. The switches once generated control cells, will not generate control cells till the minimum of recovery period of any of the sources plus time to reach source is over time for generation of next set of

$$\text{RM cells} = \text{current time} + \text{minimum (recovery period of the sources} + \text{time to reach the source)} \tag{3.4}$$

The control cells have the highest transfer priority so that these cells will not have any delay in reaching the source host in case of congestion.

Threshold level control: In the mean time, if the output buffer reaches its threshold value, then incoming cells having lower priority are discarded. If the buffer is full, then all incoming cells are dropped.

3.5.3 Merits of the Proposed Strategy

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

ii) 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.

iii) 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.

iv) 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.

v) 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.

vi) 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.

vii) 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.

3.5.4 Limitations of the Proposed Strategy:

i) In case of congestion, if any of the source is having very low call rate, the number of control cells generated are high. This is because the control cells are generated by the switch after certain fixed period which is explained below.

$$\text{Time for generation of next set of control cells} = \text{current time} + \text{minimum} \\ (\text{recovery period of the sources} + \text{time to reach the source}) \qquad (3.5)$$

If the call rate is low, then recovery period is short, and if the propagation delay is not very significant, then generation time for control cells is also low producing control cells at faster rate. If the propagation delay is high, then effect of small call rate is not significant.

ii) The sources, once reduced, will ignore all the control cells coming from all the switches. If some other switch gets congested, it may happen that the sources may not respond to the control cells sent by it (if they have already been reduced by some other switch and recovery period is not yet over) which may incur in the cell loss.

3.6 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.

3.6.1 Various Modules for Simulation

The various modules developed for the simulation are described here.

1. 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.

2. 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.

3. 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.

4. **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. **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.

3.6.2 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).

3.7 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).

Functions such as Connection Admission Control (CAC), Usage Parameter Control (UPC), Feedback Controls, Resource Allocation, etc., are made available within the ATM node equipment and are, in general, structured differently for each Service Category. The CAC and UPC procedures implementation is network specific.

3.7.1 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.

3.7.2 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:

Traffic Parameters

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.

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).

3.7.3 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.

3.7.3.1. 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.

3.7.3.2 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.

3.7.3.3 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.

3.7.3.4 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.

3.7.3.5 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).

3.8 ATM co-operative multimedia user business cases

This section describes ATM user business cases conducted in a joint effort between network operators, manufacturers of telecommunication and information technology equipment, software companies and end users from different industry sectors. The results are related to multimedia applications in the aircraft and banking industries. Common to both case studies is the concept of computer supported collaborative work (CSCW). Experiences are taken as a basis for deriving functional and technical requirements for supporting ATM networks. The focus lies on the future support of CSCW by teleservices and ATM bearer services in the network. The projects were partially funded by the European Union.

The BANK project (Banking applications using an image and broadband communications network) addresses possible multimedia enhancements of current banking services,

3.8.1 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.

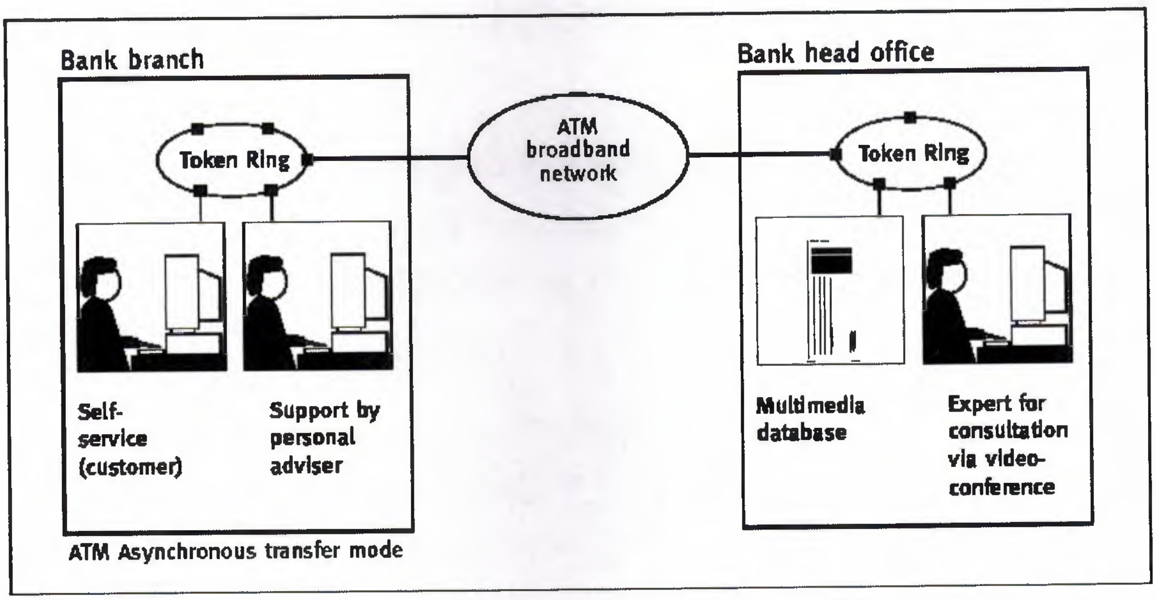

Figure 3.1 ATM network configurations for connections between bank branches and the head office.

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.

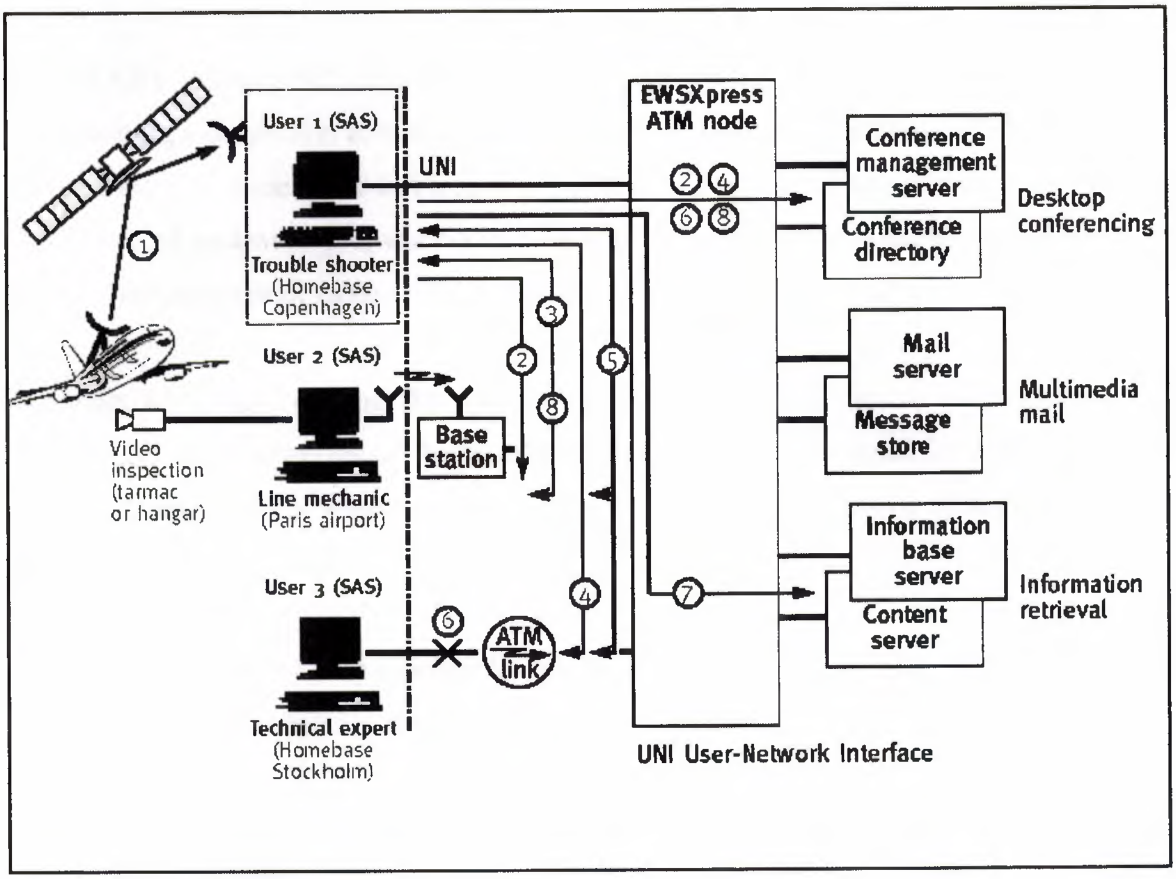

Figure 3.2 CSCW application case.

3.8.2 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.

SUMMARY and CONCLUSION

In this report, a congestion control strategy is developed for an ATM network. The results clearly shows drastic decrease in the cell loss with the proposed scheme. The congestion control strategy makes use of backward propagation of RM cells from the ATM switches, when the switches exceed their preset peak value. The approach used is simple and effective.

The main advantage of using this approach is, it responds very fast to congestion occurrence since the strategy minimizes all the possible delays. It can be concluded from the results that the proposed strategy gives good results when the link utilization is above 90% and buffer size is between 12 to 15 cell size.

More information to be added to conclusion, having at least 2 pages of information. You should generalize all the information summary of chapters 1-2 and 3.

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