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CONGESTION AND ADMISSION CONTROL IN ATM NETWORKS

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ABSTRACT

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The advent of B-ISDN has been heralded as the next generation of communication systems. These networks will be based on ATM technologies. With the proleferance of multimedia traffic over the internet it seems natural to move over to the ATM technology which have been designed specifically to support integration of data, voice and video applications with guarantees of QoS. Workstations have been used to introduce multimedia applications to the desktop, including components of voice, video and image, besides growing amount of data. This development requires networks of greater bandwith than commonly present today with the capability of handling multiservice traffic on the same network.

So with the need of high bandwith and high quality of data, video traffic and quality of service (QoS) makes the ATM technology much more popular and attractive in this fields. The basic component of an ATM network is the switches. It is a switching network technology. The ATM technology is a bit expensive due to the prices of switches and the lack of API (application programming interface) standards, therefore designing cheaper ATM switches is the main target and a key point for the success of ATM technology.

Keywords: ATM, Quality of Service (QoS), Congestion and Admission Control.

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LIST OF ABBREVIATION

ABR	Available Bit Rate
ATM	Asynchronous Transfer Mode
B-ISDN	Broadband Integrated Services Digital Network
CAC	Connection Admission Control
CBR	Constant Bit Rate
CLR	Cell Lose Ratio
HB	High-Bursty
ISDN	Integrated Services Digital Network
LB	Low-Bursty
MPEG	Moving Picture Experts Group
nrtVBR	Non-real-time Variable Bit Rate
PCR	Peak Cell Rate
QoS	Quality of Service
RSVP	Resource Reservation Protocol
rtVBR	Real-time Vriable Bit Rate
UBR	Unspecified Bit Rate
VBR	Variable Bit Rate
VCI	Virtual Channel Identifier
VPI	Virtual Path Identifier
WAN	Wide Area Network
LAN	Local Area Network
UNI	User Network Interface
NNI	Network Network Interface
HEC	Header Error Control
FIFO	First In First First Out
SF	Switch Fabric
SONET	Synchronous Optical Network
BS	Buffer Size
DRAM	Dynamic Random Access Memory
ATD	Asynchronous Time Division

TDM	Time Division Multiplexing
CLP	Cell Lose Priority
UPC	Usage Parameter Control
TES	Transform-Expand-Sample
PCM	Pulse Code Modulation
ADPCM	Adaptive Differantials Pulse Code Modulation
SAD	Speech Activity Detector
DSI	Digital Speech Information

CHAPTER I

1. INTRODUCTION

In this chapter, the basic ATM models and feature of ATM networks are explaned. The basic characteristics of an ATM network, how it is working and the topology of such a network are explained in detail.

1.1. Introduction to ATM

Computer networks have a dramatic effect on the way they interact and communicate with one another. Fast, effective communication is essential in our information-oriented society and effects all aspects of our lives. Valuable information service, in both the public and private private sectors, are currently being offered over computer networks. Voice networks were integrated with data networks (computer and communication integration) during the 1980s where as during the 1990s we are experiencing the integration of computer, communication, and consumer electronics.

Although today's computer networks are capable of supportting many useful application (e-mail and file sharing, for example) , they often inadequately address the needs of emerging multimedia applications are becoming the norm rather than the expection, it is crucial that future computer networks are designed to support these application.

1.2. ATM Insight

Asynchronous transfer mode (ATM) is often described as the technology that will allow total flexibility and efficiency to be achieved in tomorrow's high-speed, multiservice, multimedia networks. ATM has received a great deal of attention in recent years, many "network experts" predict that ATM will be the technology that finally enables high bandwidth, and time-critical applications. It is clear that the ATM technology will play a central role in the evaluation of current workgroup, campus and enterprise networks.

ATM delivers important advantages over existing LAN and WAN technologies, including the promise of scalable bandwidths and Quality of Service (QoS) guarantees, which facilitate new classes of applications such as multimedia.

If the users are in the same building, and own the cable system that the network runs over, then it's likely that we're using a Local Area Network, or LAN. LAN's tend to operate at pretty high speeds simply because we can be sure that own cabling is of good quality, and we don't have to send signals very far.

If the users are spread over different buildings, or different sites, or different cities or different countries then we're using a Wide Area Network, or Wan. A key characteristic of WANs is that we never "own" them, we can only ever rent them because the cables we're using for the WAN connections are actually owned by Telecommunications providers or PTT's.

In the past, specific communications technologies were developed for users over a LAN, like Ethernet or Token Ring, while totally different technologies were developed for the WAN, like Frame Relay or X.25. In other words, if we want to communicate from one Ethernet to another that's located in a different town, we can't use Ethernet as the WAN link. We'd have have to convert the connection to a WAN technology for the journey between towns, and then back to Ethernet at the other end.

ATM is different because it can be used in the LAN or the WAN with no conversion steps between the two. ATM is the first truly integrated network technology to emerge and it is likely to become its ultimate benefit.

ATM bases it's sharing ability on the use of very small information unit called cells. An ATM a cell is 53 bytes long, which means that in a single cell we can't even fit one line of text from a book. However by transmitting lots of cell at very high speeds, we can move a whole encyclopedia around in a few seconds.

The reason we can move these cells so quickly is that each cell is always exactly the same length, and has exactly the same format of address. The address is the part of the cell that tells us where the information is going, and the more regular this address format, the easier it is for us to perform this decision in hardware, rather than software.

Connectionless Communication

- Each Packet Fully Addressed.
- All Addresses Long and Complex.
- Each Packet Routed Individually.
- Routing in Software.

Connection-Oriented Communication

- Each Call Fully Addressed.
- Call Addresses Long and Complex, Cell Addresses Short and Simple.
- Each Cell Routed Over Existing Virtual Circuit.
- Routing in Hardware.

Figure 1.1 Connectionless and Connection-Oriented Communication

To keep address really simple, ATM is based on connection-oriented technology. In telephone system, we lift the receiver, dial a number and wait for a connection. When we are finished with the cell, we clear the connection by putting the phone down. ATM works in exactly the same way. When an ATM system wants to send information, it dials the other system, makes a connection and sends the information. This figure 1.1 compares two systems.

By operation this way, the complicated task of routing the cell is performed only once for whole call. In LAN systems, routing has to be performed on every single frame and this adds complexity and cost to the devices, called routers that are used to direct and share information streams. When we make an ATM connection, we have the chance to ask the network for a specific service quality. We could request the bandwidth we think we need, or the maximum delay on the we can tolerate, and many other service parameters.

So the combination of very efficient sharing, and the chance to ask for a connection quality means that ATM is also the first network technology that truly supports all forms of digital information.

1.3. Quality of Service (QoS) in ATM

When an ATM end station connects to the ATM network, it is essentially making a *contract* with the network based on quality of service (QoS) parameters. This constract specifies an envelope that describes the intended traffic flow. This envelope specifies values for peak bandwidth, average sustained bandwidth, and burst size.

It is the responsibility of the ATM device to adhere to the contract by means of *traffic shaping*. Traffic shaping is the use of queues to constrain data burst, limit peak data rate, and smooth jitter so that the traffic will fit within the promised envelope.

ATM swithes have the option of using *traffic policing* to enforce the contract. The switch can measure the actual traffic flow and compare it against the agreed upon traffic envelope. If it finds that traffic is outside of the agreed upon parameters, the switch can set the CLP bit of the offending cells. Setting the CLP bit makes the cell *discard eligible*, which means that the switch, or any other switch handling the cell, is allowed to drop the cell during periods of congestion.

To deliver QoS guarantees, ATM switches implement a function known as Connection Admission Control (CAC) (Figure 1.2). Whenever a connection request is received, the switch performs the CAC function. That is, the switch determines whether setting up the connection violets the QoS parameters of the requested connection. The switch accepts the connection only if it can commit the resources necessary to support that traffic level while at the same time maintaining the agreed QoS of existing connections. By accepting the connection, the network forms a traffic contract with the user. Once the connection is accepted, the network continues to provide the agreed QoS as long as the user complies with the traffic contract. CAC is a local switch function and is dependent on the architecture of the switch and local decisions on the strictnes of QoS guarantees.

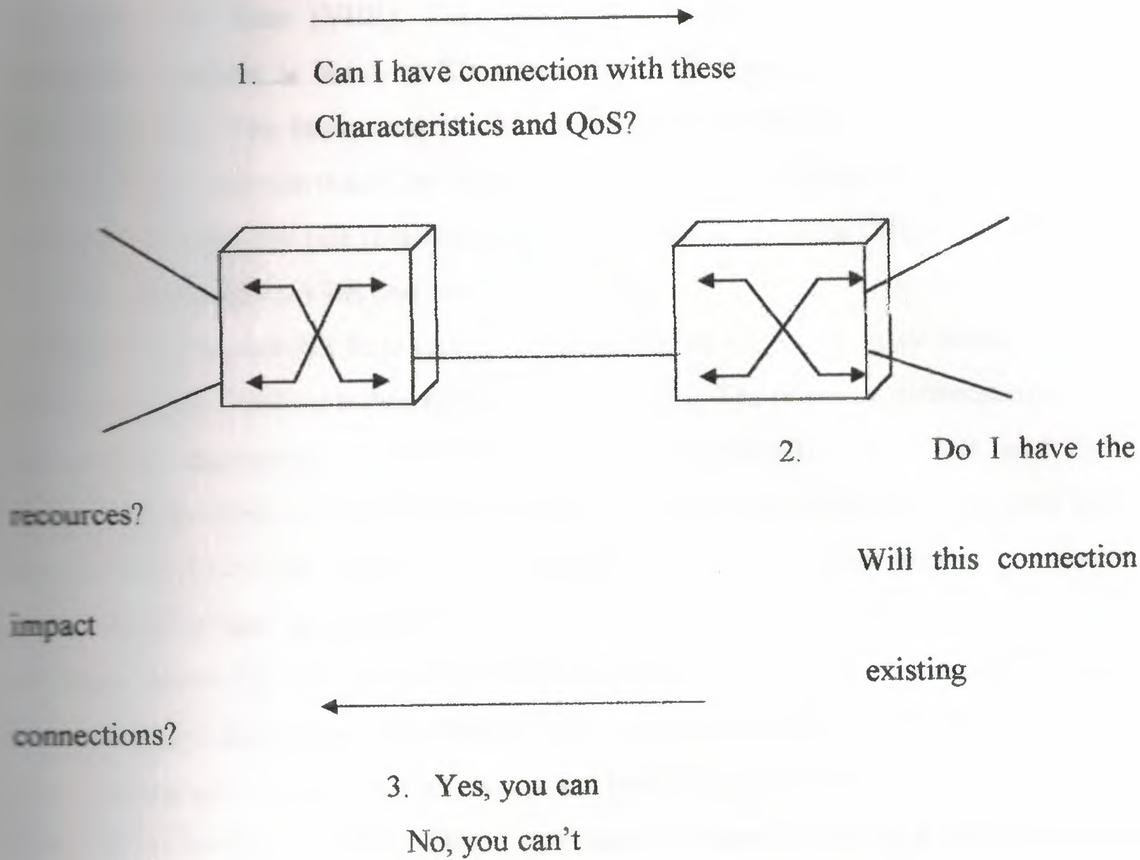

Figure 1.2. Connection Admission Control

For a given connection the major traffic contract parameters consist of the following:

1- Connection traffic descriptor, consisting of:

a-)Peak Cell Rate (PCR), which is the maximum cell rate a source is allowed to maintain.

b-)Sustainable Cell Rate (SCR), which is the average cell rate source is allowed to maintain.

c-)Maximum Burst Size (MBS), which is the maximum number of cells that a source is allowed to send consecutively at the PCR.

2- Service category, consisting of:

a-)Constant Bit Rate(CBR). This is the highest-priority category, designed for traffic that must meet strict throughput and delay requirements. such traffic includes voice and interactive video. Because of the strict delay and bandwidth requirements of this kind of service, the network must be able to maintain the PCR for every connection throughout the connection life.

b-)Variable Bit Rate (VBR). This category is designed for applications whose information transfer is bursty. VBR connections are characterized in terms of PCR, SCR and MBS. The basic idea is that bursty transmissions at rates higher than the sustainable cell rate can occur, but must be offset by periods of lower transmission rates so that hte average cell rate remains no higher than the sustainable cell rate. VBR comes in two types, real-time VBR and non-real-time VBR.

c-)Real-time Variable Bit Rate (rtVBR). This category can carry delay-sensitive traffic while using less bandwidth than CBR services require. The principle difference between applications appropriate for rt-VBR and those appropriate for rtVBR and those appropriate for CBR is that rtVBR application transmit at a rate that varies with time. Equivalently, a rtVBR source can be characterized as somewhat bursty (e.g., voice applications that use compression).

d-)Not-real-time Variable Bit Rate (nrtVBR). This service category is intended for non-real-time applications that have busty traffic and do not have strict delay guarantees (e.g., applications like airline reservations and banking transaction).

e-)Available Bit Rate (ABR): This category is intended for data traffic, such as Internet traffic. ABR traffic management defines flow control mechnaisms to allocate background bandwidth fairly among applications that do not have rigorous cell transfer delay tolerances but do have low cell loss requirements.

f-)Unspecified Bit Rate (UBR): At any given time, a certain amount of the capacity of an ATM network is consumed in carrying CBR and two types of VBR traffic. All of this unused capacity could be made available fot the UBR. This category is intended for non-real-time applications, e.i., those not requiring tightly constrained delay and delay variation. Examples are electronic mail and file transfer. UBR service does not specify traffic related service guarantees.

1.4. Switching in ATM

ATM is the transfer mode select by CCITT (ITU) for integrated broadband communication. The services carried by ATM all have their different requirements on cell delay, cell jitter and cell loss. The most important part of an ATM Network is the ATM switch. No single architecture for ATM switches has so far emerged as the "right" way to build a switch that fulfils these sometimes confilicting requirements. Rather, the different architectures proposed in the literature each have their strengths and weaknesses.

CHAPTER 2

2. ATM Architecture

B-ISDN Protocol Reference Model

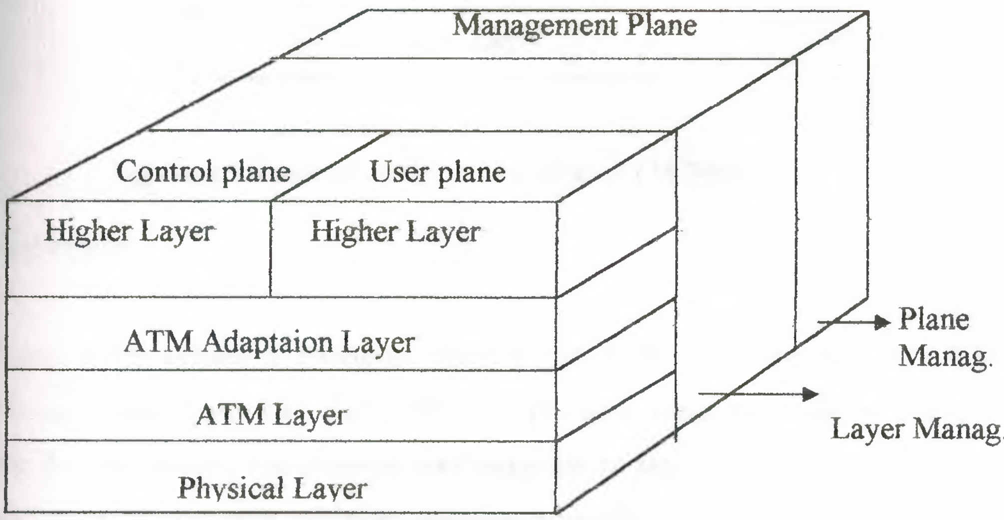

Figure 2.1 B-ISDN Protocol Reference Model

The ATM protocol reference model is based on standards developed by the ITU. The protocol reference model for ATM is divided into three planes: a **user plane** to transport user information, a **control plane** to manage signaling information, and a **management plane** to maintain the network and carry out operational function. The management plane is further subdivided into layer management and plane management to manage the different layers and planes (Figure 2.1). Protocols of the control plane and the user plane include following layers: the **physical layer**, **the ATM layer**, and the **ATM adaptation layer**. The ATM layered network architecture is shown in Figure 2.2.

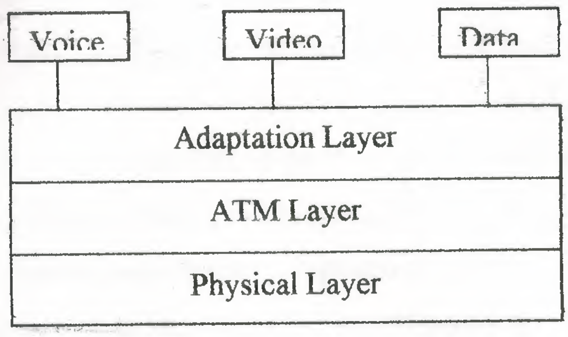

Figure 2.2 Layered Architecture of an ATM Network

2.1. Phsical Layer

The physical layer defines a transport method for ATM cells between two ATM entities. It is divided into two sublayers: the physical medium sublayer which is responsible for the correct transmission and reception of bits on the physical medium, and transmission convergence sublayer which is primarily responsible for the framing of data transported over the physical medium.

2.1.1 Physical Medium (PM) Sublayer

The physical medium sublayer performs medium dependent functions. For example, it provides bit transmission capabilities including bit alignment, line coding and electrical/optical conversion. The PM sublayer is also responsible for bit timing, that is the insertion and extraction of bit timing information. The PM sublayer currently supports two types of interface: optical and electrical.

2.1.2 Transmission Convergence (TC) Sublayer

Above the physical medium sublayer is the transmission convergence sublayer. The ITU-T recommendation specifies two options for for TC sublayer transmission frame structure: cell based and Synchronous Digital Hierarchy (SDH). In the cell based case, cells are transported continuously without any regular frame structure. Under SDH, cells are carried in a special frame structure based on the North American SONET

(Synchronous Optical Network) protocol. The TC sublayer is responsible for the following four functions: cell rate decoupling, header error control, cell delineation, and transmission frame adaptation. **Cell rate decoupling** is the insertion of idle cells at the sending side to adapt the ATM cell stream's rate to the rate of the transmission path. **Header error control** is the insertion of an 8 bit CRC polinomial in the ATM cell header to protect the contents of the ATM cell header. **Cell delineation** is the detection of cell boundaries. **Transmission frame adaptation** is the encapsulation of departing cells into an appropriate framing structure (either cell-based or SDH-based).

2.2 ATM Layer

The ATM layer is a unique layer that carries all the different classes of services supported by B-ISDN within a 53-byte cell. The ATM layer is responsible for cell relaying between ATM-layer entities, cell multiplexing of individual connections into composite flows of cells, cell demultiplexing of composite flows into individual connections, cell rate decoupling or unassigned cell insertion an deletion, priority processing and scheduling of cells, cell loss priority marking and reduction, and generic flow control access.

The fields present in the ATM cell header, shown in Table 2.1 define the functionality of the ATM layer. The cell header contains a generic flow control (GFC) field, the VPI/VCI fields, a payload type indicator (PTI) field, a cell loss priority (CLP) field, and a header checksum field.

Cell (53 Bytes)					
Header					Information
Generic Flow Control	VCI/VPI Field	Payload Type Indicator	Cell Loss Priority	Header Checksum	Payload
4 bits	24 bits	2 bits	2 bits	8 bits	48 bytes

Table 2.2.1 ATM cell structure

The GFC field is used by the user network interface (UNI) 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. The VCI/VPI fields are used for channel identification and simplication of the multiplexing process. The PTI field is used to distinguish between cells and control cells. This allows control and signalling data to be transmitted on a different sub-channel from user data. The CLP field is used to indicate whether a cell may be discarded during periods of network congestion. The header checksum field is used to protect the header field from transmission errors.

It is important to realize that the functions performed by the ATM layer are designed to be carried out in hardware at very high data rates.

2.2.1 ATM Layer Functions

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

An ATM switch may either be a VP switch, in which case it only translates the VPI values contained in cell headers, or it may be a VP/VC switch, in which case it translates the incoming VCI value into an outgoing VPI/VCI pair. Since VPI and VCI values don't represent a unique end-to-end virtual connection, they can be reused at different switches through network. This is important, because the VPI and VCI fields are limited in length and would be quickly exhausted if they were used simply as destination addresses.

The ATM layer supports two types of virtual connection: switch virtual connections (SVC) and permanent or semipermanent virtual connections (PVC). SVC are established and torn down dinamically by an ATM signaling procedure. That is, they only exist for the duration of a single call. PVC on the other hand, are established by network administrators and continue to exist as long as the administrator leaves them up, even if they are not used to transmit data.

Other important functions of the ATM layer include cell multiplexing and demultiplexing, cell header creation and extraction, and generic flow control. Cell multiplexing is the merging of cells from several calls onto a single transmission path, cell header creation is the attachment of a 5-octet cell header to each 48-octet block of user payload, and generic flow control is used at the UNI to prevent short-term overload conditions from occuring within the network.

<table>
<tr><td rowspan="6">Layer Management</td><td>Higher Layer Functions</td><td colspan="2">Higher Layers</td></tr>
<tr><td>Convergence</td><td>CS</td><td rowspan="2">AAL</td></tr>
<tr><td>Segmentation and Reassembly</td><td>SAR</td></tr>
<tr><td>Generic Flow control
Cell header generation/extraction
Cell VPI/VCI translation
Cell multiplex and demultiplex</td><td colspan="2">ATM Layer</td></tr>
<tr><td>Cell Rate Decoupling
Header Error Control (HEC)
Cell Delination
Transmission Frame adaptation
Transmission frame generation/recovery</td><td>TC</td><td rowspan="2">Physical Layer</td></tr>
<tr><td>Bit Timing
Physical medium</td><td>PM</td></tr>
</table>

Table 2.2.1.1 Functions of each layer in the protocol reference model

2.2.2 ATM Layer Service Categories

The ATM forum and ITU-T have defined several distinc service categories at the ATM layer. The categories defined by the ATM Forum include constant bit rate (CBR), real-time variable bit rate (VBR-rt), non-real time variable bit rate (VBR-nrt), avsliable bit rate (ABR), unspecifie bit rate (UBR). ITU-T defined four service categories, namely, deterministic (DBR), statistical bit rate (SBR), avaliable bit rate (ABR) and ATM block transfer (ABT). The first of the three ITU-T service categories correspond roughly to the ATM forum's CBR, VBR and ABR classification, respectively. The fourth service category, ABT is solely defined by ITU-T and is intended for bursty data aplications. The UBR category defined by the ATM Forum is for calls that request no quality of service guarantees at all. Figure 2.5 lists the ATM service categories, their quality of service (QoS) parameters, and the traffic descriptors required by the service category during call establishment.

The constant bit rate CBR (or deterministic bit rate) service category provides a very strict QoS guarantee. It is targeted at real-time application, such as voice and raw video, which mandatesevere destrictictions on delay, delay variance (jitter) and cell lose rate. The only traffic descriptors required by the CBR service are the peak cell rate and the cell delay variation tolerance. A fixed amount of bandwidt, determined primarily by the call's peak cell rate, is reserved for CBR connection.

The rea-time variable bit rate VBR-rt (or statistical bit rate SBR) service category is intended for real time bursty applications (e.g., compressed video), which also require strict QoS guarantees. The primary difference between CBR and VBR-rt is in the traffic descriptors they use. The VBR-rt service requires the specification of the sustained (or avarage) cell rate and burst tolerence (i.e., burst length) in addition to the peak cell rate and the cell delay variation tolerence. The ATM Forum also defines a non-real time VBR-nrt service category, in which cell delay variance is not guaranteed.

The avaliable bit rate (ABR) service category is defined to exploit the network's unutilized bandwidth. It is intended for non-real time data applications in which the source is amenable to enforced adjustment of its transmission rate. A minimum cell rate is reserved for the ABR connection and therefore guaranteed by the network. When the network has utilized bandwidth, ABR sources are allowed to increase their cel rates up to an allowed cell rate (ACR), a value which is periodically updated by the ABR flow control mechanism. The value of ACR always falls between the minimum and the peak cell rate for the connection and is determined by the network.

The ATM Forum defines another service category for non-real-time applications called the unspecified bit rate (UBR) service category. UBR service is entirely best effort; the call is provided with no QoS guarantees. The UTI-t also defines an additional service category for non-real-time data applications. The ATM block transfer (ABT) service category is intended for transmission of short bursts, or blocks, of data. Before transmitting a block, the source request a reservation of bandwidth from the network. If the ABT service is being used with immediate transmission option (ABT/IT); the block of data is sent at the same time as the reservation request. If bandwidth is not avaliable for transporting the block, then it is simply discarded, and source must retransmit it. In the ABT service with delayed transmission (ABT/DT), the source waits for confirmation from the network that enough bandwidth is avaliable before transmitting the block of data. In both case, the network temporarily reserves bandwidth according to the peak cell rate for each block. Immediately after transporting the block, the network releases the reserved bandwidth.

ITU-T Service Categories →	DBR	SBR	ABT	ABR	
ATM Forum Service Categories →	CBR	VBR-rt	VBR-nrt	ABR	UBR
Cell Loss Rate	specified				unspecified
Cell Transfer Delay	specified			unspecified	
Cell Delay Variation	specified		unspecified		
Traffic Descriptors	PCR/CDVT	PCR/CDVT SCR/BT		PCR/CDVT MCR/ACR	PCR/CDVT

PCR = Peak Cell Rate — SCR = Sustained Cell Rate

CDVT = Cell Delay Variation Tolerance — BT = Burst Tolerance

MCR = Minimum Cell Rate — ACR = Allowed Cell Rate

Table 2.2.2.1 ATM layer service categories

2.3 ATM Adaptatiton Layer

The ATM adaptation layer (AAL), which resides atop the ATM layer, is responsible for maping the requirements of higher layer protocols onto the ATM network. The purpose of the AAL is to provide a link between the services required by higher network layers and the generic ATM cells used by the ATM layer. It operates in ATM devices at the edge of the ATM network and is totally absent in ATM switches. The adaptation layer is divided into two sublayers: the convergence sublayer (CS), which performs error detection and handling, timing and clock recovery; and the segmentation and reassembly (SAR) sublayer, which performs segmentation of convergence sublayer protocol data units (PDUs) into ATM cell-sized SAR sublayer service data units (SDUs) and vice versa.

In order to support different service requirements. the ITU-T has proposed four AAL-specific service classes. Table 3 depicts the four service classes defined in recommendation I,362 [1]. Note that while these AAL service classes are similar in many ways to the ATM layer service categories defined in the previous section, they are not the same; each exists at a different layer of the protocol reference model, and each requires a different set of functions.

AAL service class A corresponds to constant bit rate (CBR) services with a timing relation required between source and destination. The connection mode is connection-oriented. CBR audio and video belong to this class. Class B correspons to variable bit rate (VBR) services. This class also requires timing between source and destination, and its mode is connecting –oriented. VBR audio and video examples of class B services. Class C also corresponds to VBR connection-oriented services, but the timing between soure and destination needs not be related. Class includes connection-oriented data transfer such as X.25, signaling and future high speed data services. Class D corresponds to connectionless services. Connectionless data services such as those supported by LAN's and MAN's are examples of class D services.

Four AAL types (Types 1,2,3/4 and 5),each with a unique SAR sublayer and CS sublayer, are defined to support the four service classses. AAL Type 1 suports constant bit rate services (classA). and AAL Type 2 supports variable bit rate services with a timing relation between source and destination (clas B). AAL Type ¾ was origanally specified as two different AAL types (Type 3 and Type 4), but due to their inherent similarities they were eventuaaly merged to support both class C and class D services. AAL Type 5 also supports class C and class D services.

2.3.1 AAL Type 5

Currently the most widely used adaptation layer is AAL Type 5. AAL Type 5 supports connection-oriented and connectionless services in which there is no timing relation between source and destination (classes C and D). It's functionality was intentionally made simple in order to support high speed data transfer. AAL Type 5 assumes that the layers above to ATM adaption layer can perform error recovery, retransmision and sequence membering when required, and thus, it does not provide these funtions. Therefore only non-assumed operation is provided; lost or corrupted AAL Type 5 packets will not be corrected by retransmission.

Figure 2.3.1.1 depicts the SAR-SDU format for ALL Type 5 (5,6). The SAR sublayer of ALL Type 5 performs segmentation of a CS-PDU into a size suitable for the SAR-SDU payload. Unlike other AAL Types , type 5 devotes to entire 48-octed payload of the ATM cell to the SAR-SDU; there is no overhead. An AAL specific flag (End of frame) in the ATM Payload Type (PT) field of the cell header is set when yhe last cell of a CS-PDU is sent. The reasembly of CS-PDU frames at the destination is controlled by using this flag.

Cell Header	SAR-SDU Payload

Figure 2.3.1.1 SAR-SDU format for AAL Type 5

Figure 2.3.1.2 depicts the CS-PDU format for AAL Type 5 (5,6). It contains the user data payload, along with any necassary padding bits (PAD) and a CS-PDU trailer, which are added by the CS sublayer when it receives the user information from the higher layer. The CS-PDU is padded using 0 to 47 bytes of PAD field to make the length of the CS-PDU an integral multiple of 48 bytes (the size of the SAR-SDU payload). At the receiving end, a reassembled PDU is the passed to the CS sublayer from the SAR sublayer, and CRS values are then calculate and compared. If there is n error, the PAD field is removed by using the value of length field (LF) in the CS-PDU trailer, and user data is passed to the higher layer. If no error is detected, the erroneous information is either delivered to the user or discarded according to the user's choice.

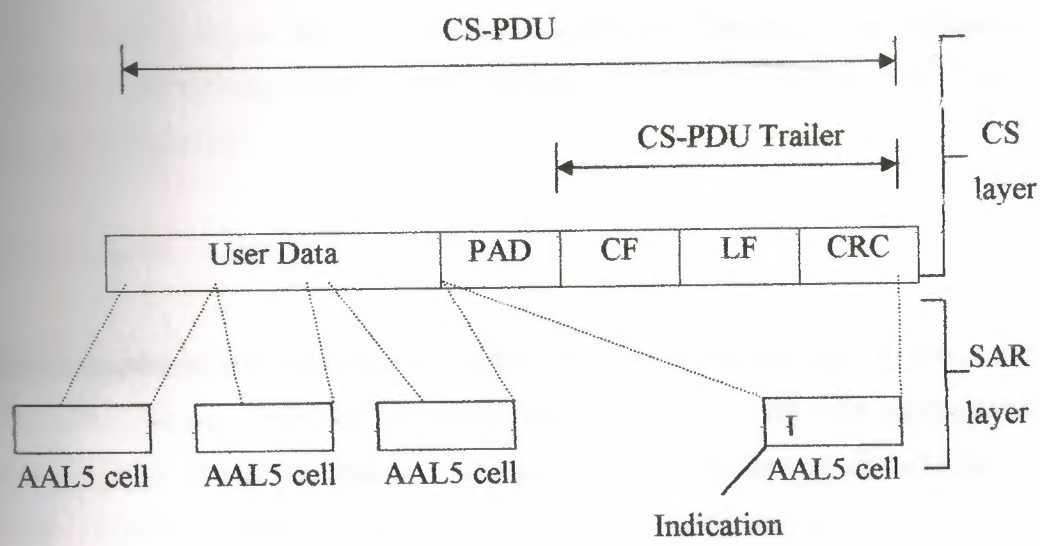

Figure 2.3.1.2 CS-PDU format, segmentation and reassembly of AAL Type 5

2.3.2 AAL Type 1

AAL Type 2 supports constants bit rate services with a fixed timing relation between source and destination users (class A). At the SAR sublayer, it defines a 48-octet service data unit (SDU), which contains 47 octets of user payload, 4 bits for a sequence number, and a 4-bit CRC value to detect errors in the sequece number field. AAL Type 1 performs the following services at the CS sublayer: forward error correction to ensure high quality of audio and video applications, clock recovery b monitoring the buffer filling, explicit time indication by inserting a time stamp in the CS-PDU, and handling of lost and misinserted cels which are recognized by the SAR. At the time of the writing, the CS-PDU format has not been decided.

2.3.3 AAL Type 2

AAL Type 2 supports variable bit rate services with a timing relation between source and destination (class B). AAL Type 2 is nearly identical to AAL Type 1, except that it transfers service data units at a variable bit rate, not a constant bit rate. Furthermore, AAL Type 2 accepts variable length CS-PDUs, and thus, there may exist some SAR-SDUs which are not completely filled with user data. The CS sublayer for AAL Type 2 perform the follow functions: forward error corecion for audio and video services, clock recovery by inserting a time stamp in the CS-PDU, and handling of lost andmisinserted cells. At the time of writing, both the SAR-SDU and CS-PDU formats for AAL Type 2 are still under discussion.

2.3.4 AAL Type ¾

Aal Type ¾ mainly supports services that require no timing relation between the source and destination (classes C and D). At the sublayer, it defines a 48-octet services data unit, with 44 octet of user payload, a 2-bit payload type field to indicate whether the SDU is the beginning, middle, or end of a CS-PDU, a 4-bit cell sequence number, a 10-bit multiplexing identifier that allows several CS-PDUs to be multiplexed over a single VC, a 6-bit cell payload lenght indicator, and a 10-bit CRC code that covers the payload. The CS-PDU format allows for up 65535 octets of user payload and contains a header and trailer to delineate the PDU.

The functions that AAL Type ¾ performs include segmentation and reassembly of variable-lenght user data error handling. It supports message mode (for framed data transfer) as well as streaming mode (for streamed data trnasfer). Since Type ¾ mainly intended for data services, it provdes a retransmission mechanism if necessary.

CHAPTER 3

3.1 Cell Structure of ATM

ATM provides a good bandwith flexibility and can be used efficiently from desktop computers to local area and wide area networks. As itis shown Figure 3.1. All packets are of fixed length 53 bytes (5 bytes for header and 48 bytes for information). No processing like error control is done on the information field of ATM cells inside the network and it is carried transparently in the network.

Header of Cell [5 Bytes]	Payload of Cell [48 Bytes]

Figure 3.1 A typical ATM Cell of 53 bytes

The main characteristics of ATM can summarized as follows:

1- ATM provides cell sequence integrity. That is cells arrive at the destination in the same order as they left the source. This may not be the case with other packet-switched networks.

2- Cells are much smaller than standard packet switched networks. This reduces the value of delay variance, making ATM acceptable for timing sensitive information like voice.

3- The quality of transmission links has lead to the omission of overheads, such as error correction, in order to maximize efficiency.

4- There is no space between cells. At times when the network is idle, unassigned cells are transported.

The structure of the cell is important for the overall functionality of the ATM network. A large cell gives a better payload to overhead ratio, but at the expense of longer, more variable delays. Shorter packets overcome this problem, however the amount of information carried per packet is reduced. An agreement between these two conflicting requirements was reached, and a standard cell format chosen. The ATM cell consists of a 5-ocet header and a 48-octet information field after the header.

information contained in the header is dependent on whether the cell is carrying rmation from the user network to the first ATM public exchange (User-Network rface - UNI), or between ATM exchanges in the trunk network (Network-Node rface - NNI). The formats of the two types of header are shown below Figure 3.2.

8

1

GFC	VPI	
VPI	VCI	
VCI		
VCI	PT	CLP
HEC		

Octet 1

Octet 5

Bit 1 Bit 8

VPI		
VPI	VCI	
VCI		
VCI	PT	CLP
HEC		

User-Network Interface (UNI) **Network-Network**
erface (NNI)

Figure 3.2 Structure of UNI and NNI

e VPI is the first field in the ATM header at the NNI, and the second field in the der at the UNI. This value identifies the virtual path used for this connection. The ameters associated with the VPI include bandwidth and input or output port used. A h may consist of several channels, each carrying a portion of the total bandwidth ocated tothat path. The virtual path identifier is a decimal number in the range 0-4095 hin the NNI and 0-255 in the UNI.

e VCI is the second field in the ATM header at the NNI and the third at the UNI. The ameters associated with the VCI include bandwidth, VPI, in and out portsand in/out CI. The Virtual Channel Identifier is a decimal number in the range 0-65535.

th the VPI and VCI may be used to route traffic through the switch. In some tances, cells are switched on the VPI value only.

s important to note that VPIs and VCIs are unique on a link-by-link basis. In other rds, Virtual Path 10 on switch A is not same as Virtual Path 10 on switch B. Further, tual Path 10 on switch A, Port 1a1, is not the same as Path 10 on switch A, port 1a2. e VPI and VCI values are interpreted at each switch and used to determine the output nk, and outgoing VPI and VCI values. The fields within an ATM cell at the UNI level re defined as Figure 3.2.

GFC (Generic Flow Control)

The first four bits of byte 1 in the header are defined as GFC and currently they are not ised. Exact mechanisms for flow control are still under development, and no explicit lefinition for this field exists at this time.

VPI (Virtual Path Identifier)

The last four bits of byte 1 and the first four bits of byte 2 are reserved for the VPI.

VCI (Virtual Channel Identifier)

The second half of byte 2, all of byte 3 and first half of byte 4 are reserved for the VCI.

PTI (Payload Type Indicator)

The next three bits of byte 4 are reserved for the PTI. PTI is used to indicate the type of information carried in the cell. Codes within this field are defined as: values 0-3 identify various type of user data, values 4 and 5 indicate management information, value 6 and 7 are reserved for future definition. These reserved values are expected to be used in the future implementations of a flow control algorithm called BECN.

CLP (Cell Loss Priority)

CLP is contained in the last bit of byte 4. If set, this bit indicates the eligibilityof the cell for discarding under congested conditions. Currently, this bit is not used until a clear definition of who sets and decides whether this bit should be set is determined.

HEC (Header Error Check)

The last byte of the header (byte 5) contains a header error control. This error-correction is calculated across the previous four bytes of the header. It is designed to detect multiple header errors and to correct single bit errors. It provides protection against mis-delivery of cells due to address errors. It does not contain any indication of the quality or integrity of the data within the payload field.

Payload

The remaining 48 bytes of the cell contain user information. The ATM adaptation layer (AAL) accomplishes inserting data into the payload field. It should be noted that depending on the AAL proces, not all-48 bytes would contain user information. Up to four bytes may be the AAL itself.

<table>
<tr><td colspan="3">VPI</td></tr>
<tr><td>VPI</td><td colspan="2">VCI</td></tr>
<tr><td colspan="3">VCI</td></tr>
<tr><td>VCI</td><td>PTI</td><td>CLP</td></tr>
<tr><td colspan="3">HEC</td></tr>
<tr><td colspan="3">PAYLOAD
48 Bytes</td></tr>
</table>

Figure 3.3 ATM NNI Cell Format

The cell structure at the NNI level is essentially the same as the UNI. The only notable difference is the GFC field, which is eliminated, and four bits of byte 1 are pretended to the VPI field (Figure3.3). For this reason, at the NNI levle the range of values avaliable for the VPI is 0-4095. At the UNI level the range is from 0-255.

NEAR EAST UNIVERSITY

FACULTY OF ENGINEERING

DEPARTMENT OF COMPUTER ENGINEERING

CONGESTION AND ADMISSION CONTROL IN ATM NETWORKS

GRADUATION PROJECT
COM 400

Student : Harun USLU

Supervisor : Mr. Halil ADAHAN

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ABSTRACT

CONGESTION AND ADMISSION CONTROL IN ATM NETWORKS

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The advent of B-ISDN has been heralded as the next generation of communication systems. These networks will be based on ATM technologies. With the proleferance of multimedia traffic over the internet it seems natural to move over to the ATM technology which have been designed specifically to support integration of data, voice and video applications with guarantees of QoS. Workstations have been used to introduce multimedia applications to the desktop, including components of voice, video and image, besides growing amount of data. This development requires networks of greater bandwith than commonly present today with the capability of handling multiservice traffic on the same network.

So with the need of high bandwith and high quality of data, video traffic and quality of service (QoS) makes the ATM technology much more popular and attractive in this fields. The basic component of an ATM network is the switches. It is a switching network technology. The ATM technology is a bit expensive due to the prices of switches and the lack of API (application programming interface) standards, therefore designing cheaper ATM switches is the main target and a key point for the success of ATM technology.

Keywords: ATM, Quality of Service (QoS), Congestion and Admission Control.

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LIST OF ABBREVIATION

ABR	Available Bit Rate
ATM	Asynchronous Transfer Mode
B-ISDN	Broadband Integrated Services Digital Network
CAC	Connection Admission Control
CBR	Constant Bit Rate
CLR	Cell Lose Ratio
HB	High-Bursty
ISDN	Integrated Services Digital Network
LB	Low-Bursty
MPEG	Moving Picture Experts Group
nrtVBR	Non-real-time Variable Bit Rate
PCR	Peak Cell Rate
QoS	Quality of Service
RSVP	Resource Reservation Protocol
rtVBR	Real-time Vriable Bit Rate
UBR	Unspecified Bit Rate
VBR	Variable Bit Rate
VCI	Virtual Channel Identifier
VPI	Virtual Path Identifier
WAN	Wide Area Network
LAN	Local Area Network
UNI	User Network Interface
NNI	Network Network Interface
HEC	Header Error Control
FIFO	First In First First Out
SF	Switch Fabric
SONET	Synchronous Optical Network
BS	Buffer Size
DRAM	Dynamic Random Access Memory
ATD	Asynchronous Time Division

TDM	Time Division Multiplexing
CLP	Cell Lose Priority
UPC	Usage Parameter Control
TES	Transform-Expand-Sample
PCM	Pulse Code Modulation
ADPCM	Adaptive Differantials Pulse Code Modulation
SAD	Speech Activity Detector
DSI	Digital Speech Information

CHAPTER I

1. INTRODUCTION

In this chapter, the basic ATM models and feature of ATM networks are explaned. The basic characteristics of an ATM network, how it is working and the topology of such a network are explained in detail.

1.1. Introduction to ATM

Computer networks have a dramatic effect on the way they interact and communicate with one another. Fast, effective communication is essential in our information-oriented society and effects all aspects of our lives. Valuable information service, in both the public and private private sectors, are currently being offered over computer networks. Voice networks were integrated with data networks (computer and communication integration) during the 1980s where as during the 1990s we are experiencing the integration of computer, communication, and consumer electronics.

Although today's computer networks are capable of supportting many useful application (e-mail and file sharing, for example) , they often inadequately address the needs of emerging multimedia applications are becoming the norm rather than the expection, it is crucial that future computer networks are designed to support these application.

1.2. ATM Insight

Asynchronous transfer mode (ATM) is often described as the technology that will allow total flexibility and efficiency to be achieved in tomorrow's high-speed, multiservice, multimedia networks. ATM has received a great deal of attention in recent years, many "network experts" predict that ATM will be the technology that finally enables high bandwidth, and time-critical applications. It is clear that the ATM technology will play a central role in the evaluation of current workgroup, campus and enterprise networks.

ATM delivers important advantages over existing LAN and WAN technologies, including the promise of scalable bandwidths and Quality of Service (QoS) guarantees, which facilitate new classes of applications such as multimedia.

If the users are in the same building, and own the cable system that the network runs over, then it's likely that we're using a Local Area Network, or LAN. LAN's tend to operate at pretty high speeds simply because we can be sure that own cabling is of good quality, and we don't have to send signals very far.

If the users are spread over different buildings, or different sites, or different cities or different countries then we're using a Wide Area Network, or Wan. A key characteristic of WANs is that we never "own" them, we can only ever rent them because the cables we're using for the WAN connections are actually owned by Telecommunications providers or PTT's.

In the past, specific communications technologies were developed for users over a LAN, like Ethernet or Token Ring, while totally different technologies were developed for the WAN, like Frame Relay or X.25. In other words, if we want to communicate from one Ethernet to another that's located in a different town, we can't use Ethernet as the WAN link. We'd have have to convert the connection to a WAN technology for the journey between towns, and then back to Ethernet at the other end.

ATM is different because it can be used in the LAN or the WAN with no conversion steps between the two. ATM is the first truly integrated network technology to emerge and it is likely to become its ultimate benefit.

ATM bases it's sharing ability on the use of very small information unit called cells. An ATM a cell is 53 bytes long, which means that in a single cell we can't even fit one line of text from a book. However by transmitting lots of cell at very high speeds, we can move a whole encyclopedia around in a few seconds.

The reason we can move these cells so quickly is that each cell is always exactly the same length, and has exactly the same format of address. The address is the part of the cell that tells us where the information is going, and the more regular this address format, the easier it is for us to perform this decision in hardware, rather than software.

Connectionless Communication

- Each Packet Fully Addressed.
- All Addresses Long and Complex.
- Each Packet Routed Individually.
- Routing in Software.

Connection-Oriented Communication

- Each Call Fully Addressed.
- Call Addresses Long and Complex, Cell Addresses Short and Simple.
- Each Cell Routed Over Existing Virtual Circuit.
- Routing in Hardware.

Figure 1.1 Connectionless and Connection-Oriented Communication

To keep address really simple, ATM is based on connection-oriented technology. In telephone system, we lift the receiver, dial a number and wait for a connection. When we are finished with the cell, we clear the connection by putting the phone down. ATM works in exactly the same way. When an ATM system wants to send information, it dials the other system, makes a connection and sends the information. This figure 1.1 compares two systems.

By operation this way, the complicated task of routing the cell is performed only once for whole call. In LAN systems, routing has to be performed on every single frame and this adds complexity and cost to the devices, called routers that are used to direct and share information streams. When we make an ATM connection, we have the chance to ask the network for a specific service quality. We could request the bandwidth we think we need, or the maximum delay on the we can tolerate, and many other service parameters.

So the combination of very efficient sharing, and the chance to ask for a connection quality means that ATM is also the first network technology that truly supports all forms of digital information.

1.3. Quality of Service (QoS) in ATM

When an ATM end station connects to the ATM network, it is essentially making a *contract* with the network based on quality of service (QoS) parameters. This constract specifies an envelope that describes the intended traffic flow. This envelope specifies values for peak bandwidth, average sustained bandwidth, and burst size.

It is the responsibility of the ATM device to adhere to the contract by means of *traffic shaping*. Traffic shaping is the use of queues to constrain data burst, limit peak data rate, and smooth jitter so that the traffic will fit within the promised envelope.

ATM swithes have the option of using *traffic policing* to enforce the contract. The switch can measure the actual traffic flow and compare it against the agreed upon traffic envelope. If it finds that traffic is outside of the agreed upon parameters, the switch can set the CLP bit of the offending cells. Setting the CLP bit makes the cell *discard eligible*, which means that the switch, or any other switch handling the cell, is allowed to drop the cell during periods of congestion.

To deliver QoS guarantees, ATM switches implement a function known as Connection Admission Control (CAC) (Figure 1.2). Whenever a connection request is received, the switch performs the CAC function. That is, the switch determines whether setting up the connection violets the QoS parameters of the requested connection. The switch accepts the connection only if it can commit the resources necessary to support that traffic level while at the same time maintaining the agreed QoS of existing connections. By accepting the connection, the network forms a traffic contract with the user. Once the connection is accepted, the network continues to provide the agreed QoS as long as the user complies with the traffic contract. CAC is a local switch function and is dependent on the architecture of the switch and local decisions on the strictnes of QoS guarantees.

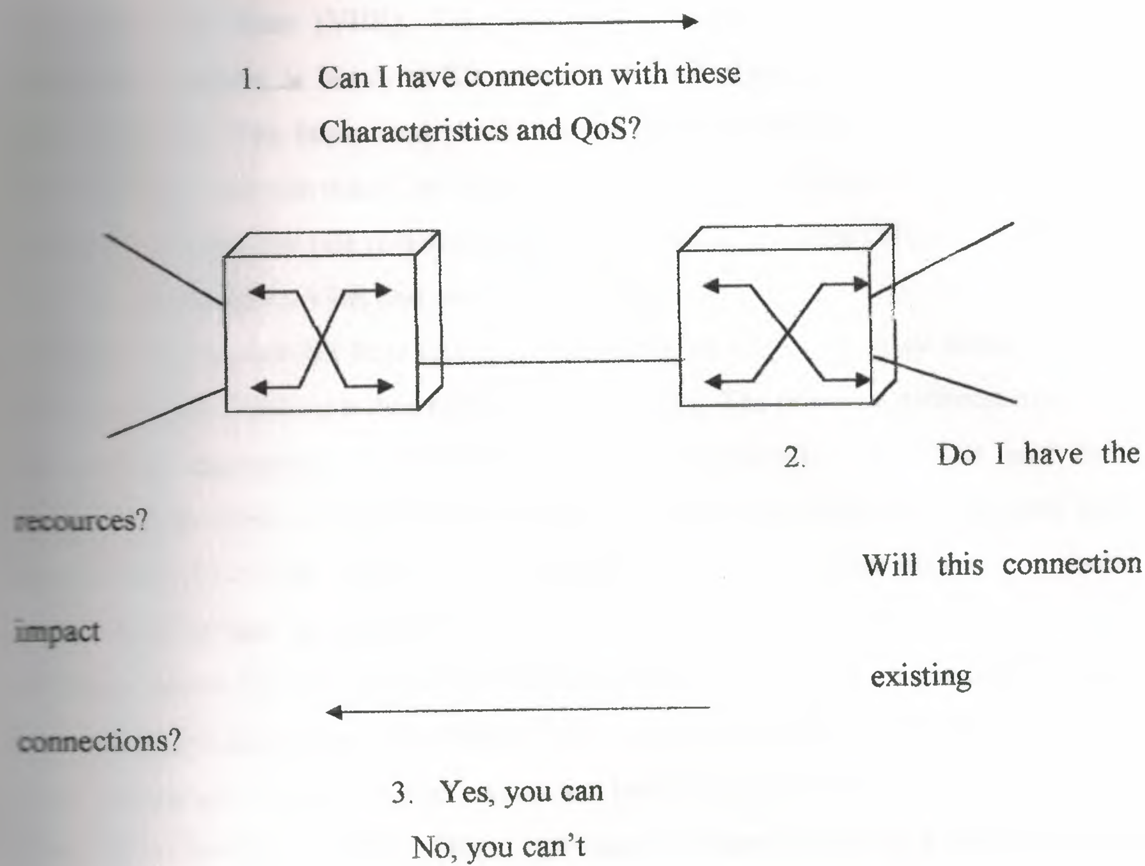


Figure 1.2. Connection Admission Control

For a given connection the major traffic contract parameters consist of the following:

1- Connection traffic descriptor, consisting of:

a-)Peak Cell Rate (PCR), which is the maximum cell rate a source is allowed to maintain.

b-)Sustainable Cell Rate (SCR), which is the average cell rate source is allowed to maintain.

c-)Maximum Burst Size (MBS), which is the maximum number of cells that a source is allowed to send consecutively at the PCR.

2- Service category, consisting of:

a-)Constant Bit Rate(CBR). This is the highest-priority category, designed for traffic that must meet strict throughput and delay requirements. such traffic includes voice and interactive video. Because of the strict delay and bandwidth requirements of this kind of service, the network must be able to maintain the PCR for every connection throughout the connection life.

b-)Variable Bit Rate (VBR). This category is designed for applications whose information transfer is bursty. VBR connections are characterized in terms of PCR, SCR and MBS. The basic idea is that bursty transmissions at rates higher than the sustainable cell rate can occur, but must be offset by periods of lower transmission rates so that hte average cell rate remains no higher than the sustainable cell rate. VBR comes in two types, real-time VBR and non-real-time VBR.

c-)Real-time Variable Bit Rate (rtVBR). This category can carry delay-sensitive traffic while using less bandwidth than CBR services require. The principle difference between applications appropriate for rt-VBR and those appropriate for rtVBR and those appropriate for CBR is that rtVBR application transmit at a rate that varies with time. Equivalently, a rtVBR source can be characterized as somewhat bursty (e.g., voice applications that use compression).

d-)Not-real-time Variable Bit Rate (nrtVBR). This service category is intended for non-real-time applications that have busty traffic and do not have strict delay guarantees (e.g., applications like airline reservations and banking transaction).

e-)Available Bit Rate (ABR): This category is intended for data traffic, such as Internet traffic. ABR traffic management defines flow control mechnaisms to allocate background bandwidth fairly among applications that do not have rigorous cell transfer delay tolerances but do have low cell loss requirements.

f-)Unspecified Bit Rate (UBR): At any given time, a certain amount of the capacity of an ATM network is consumed in carrying CBR and two types of VBR traffic. All of this unused capacity could be made available fot the UBR. This category is intended for non-real-time applications, e.i., those not requiring tightly constrained delay and delay variation. Examples are electronic mail and file transfer. UBR service does not specify traffic related service guarantees.

1.4. Switching in ATM

ATM is the transfer mode select by CCITT (ITU) for integrated broadband communication. The services carried by ATM all have their different requirements on cell delay, cell jitter and cell loss. The most important part of an ATM Network is the ATM switch. No single architecture for ATM switches has so far emerged as the "right" way to build a switch that fulfils these sometimes confilicting requirements. Rather, the different architectures proposed in the literature each have their strengths and weaknesses.

CHAPTER 2

2. ATM Architecture

B-ISDN Protocol Reference Model

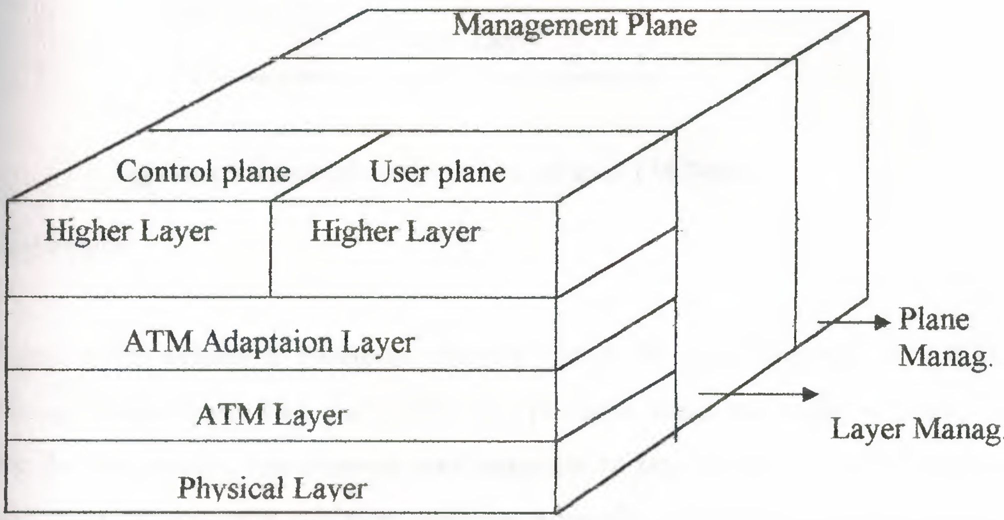


Figure 2.1 B-ISDN Protocol Reference Model

The ATM protocol reference model is based on standards developed by the ITU. The protocol reference model for ATM is divided into three planes: a **user plane** to transport user information, a **control plane** to manage signaling information, and a **management plane** to maintain the network and carry out operational function. The management plane is further subdivided into layer management and plane management to manage the different layers and planes (Figure 2.1). Protocols of the control plane and the user plane include following layers: the **physical layer**, **the ATM layer**, and the **ATM adaptation layer**. The ATM layered network architecture is shown in Figure 2.2.

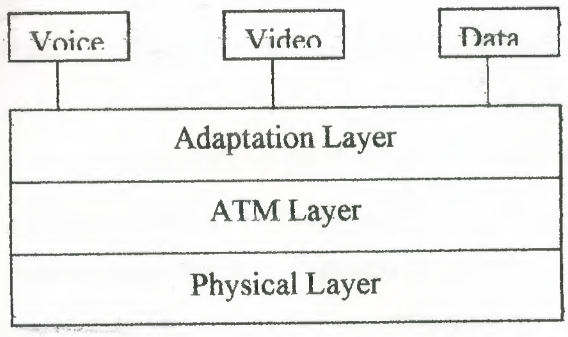


Figure 2.2 Layered Architecture of an ATM Network

2.1. Phsical Layer

The physical layer defines a transport method for ATM cells between two ATM entities. It is divided into two sublayers: the physical medium sublayer which is responsible for the correct transmission and reception of bits on the physical medium, and transmission convergence sublayer which is primarily responsible for the framing of data transported over the physical medium.

2.1.1 Physical Medium (PM) Sublayer

The physical medium sublayer performs medium dependent functions. For example, it provides bit transmission capabilities including bit alignment, line coding and electrical/optical conversion. The PM sublayer is also responsible for bit timing, that is the insertion and extraction of bit timing information. The PM sublayer currently supports two types of interface: optical and electrical.

2.1.2 Transmission Convergence (TC) Sublayer

Above the physical medium sublayer is the transmission convergence sublayer. The ITU-T recommendation specifies two options for for TC sublayer transmission frame structure: cell based and Synchronous Digital Hierarchy (SDH). In the cell based case, cells are transported continuously without any regular frame structure. Under SDH, cells are carried in a special frame structure based on the North American SONET

(Synchronous Optical Network) protocol. The TC sublayer is responsible for the following four functions: cell rate decoupling, header error control, cell delineation, and transmission frame adaptation. **Cell rate decoupling** is the insertion of idle cells at the sending side to adapt the ATM cell stream's rate to the rate of the transmission path. **Header error control** is the insertion of an 8 bit CRC polinomial in the ATM cell header to protect the contents of the ATM cell header. **Cell delineation** is the detection of cell boundaries. **Transmission frame adaptation** is the encapsulation of departing cells into an appropriate framing structure (either cell-based or SDH-based).

2.2 ATM Layer

The ATM layer is a unique layer that carries all the different classes of services supported by B-ISDN within a 53-byte cell. The ATM layer is responsible for cell relaying between ATM-layer entities, cell multiplexing of individual connections into composite flows of cells, cell demultiplexing of composite flows into individual connections, cell rate decoupling or unassigned cell insertion an deletion, priority processing and scheduling of cells, cell loss priority marking and reduction, and generic flow control access.

The fields present in the ATM cell header, shown in Table 2.1 define the functionality of the ATM layer. The cell header contains a generic flow control (GFC) field, the VPI/VCI fields, a payload type indicator (PTI) field, a cell loss priority (CLP) field, and a header checksum field.

Cell (53 Bytes)					
Header					Information
Generic Flow Control	VCI/VPI Field	Payload Type Indicator	Cell Loss Priority	Header Checksum	Payload
4 bits	24 bits	2 bits	2 bits	8 bits	48 bytes

Table 2.2.1 ATM cell structure

The GFC field is used by the user network interface (UNI) 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. The VCI/VPI fields are used for channel identification and simplication of the multiplexing process. The PTI field is used to distinguish between cells and control cells. This allows control and signalling data to be transmitted on a different sub-channel from user data. The CLP field is used to indicate whether a cell may be discarded during periods of network congestion. The header checksum field is used to protect the header field from transmission errors.

It is important to realize that the functions performed by the ATM layer are designed to be carried out in hardware at very high data rates.

2.2.1 ATM Layer Functions

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

An ATM switch may either be a VP switch, in which case it only translates the VPI values contained in cell headers, or it may be a VP/VC switch, in which case it translates the incoming VCI value into an outgoing VPI/VCI pair. Since VPI and VCI values don't represent a unique end-to-end virtual connection, they can be reused at different switches through network. This is important, because the VPI and VCI fields are limited in length and would be quickly exhausted if they were used simply as destination addresses.

The ATM layer supports two types of virtual connection: switch virtual connections (SVC) and permanent or semipermanent virtual connections (PVC). SVC are established and torn down dinamically by an ATM signaling procedure. That is, they only exist for the duration of a single call. PVC on the other hand, are established by network administrators and continue to exist as long as the administrator leaves them up, even if they are not used to transmit data.

Other important functions of the ATM layer include cell multiplexing and demultiplexing, cell header creation and extraction, and generic flow control. Cell multiplexing is the merging of cells from several calls onto a single transmission path, cell header creation is the attachment of a 5-octet cell header to each 48-octet block of user payload, and generic flow control is used at the UNI to prevent short-term overload conditions from occuring within the network.

Layer Management	Higher Layer Functions	Higher Layers	
	Convergence	CS	AAL
	Segmentation and Reassembly	SAR	
	Generic Flow control Cell header generation/extraction Cell VPI/VCI translation Cell multiplex and demultiplex	ATM Layer	
	Cell Rate Decoupling Header Error Control (HEC) Cell Delination Transmission Frame adaptation Transmission frame generation/recovery	TC	Physical Layer
	Bit Timing Physical medium	PM	

Table 2.2.1.1 Functions of each layer in the protocol reference model

2.2.2 ATM Layer Service Categories

The ATM forum and ITU-T have defined several distinc service categories at the ATM layer. The categories defined by the ATM Forum include constant bit rate (CBR), real-time variable bit rate (VBR-rt), non-real time variable bit rate (VBR-nrt), avsliable bit rate (ABR), unspecifie bit rate (UBR). ITU-T defined four service categories, namely, deterministic (DBR), statistical bit rate (SBR), avaliable bit rate (ABR) and ATM block transfer (ABT). The first of the three ITU-T service categories correspond roughly to the ATM forum's CBR, VBR and ABR classification, respectively. The fourth service category, ABT is solely defined by ITU-T and is intended for bursty data aplications. The UBR category defined by the ATM Forum is for calls that request no quality of service guarantees at all. Figure 2.5 lists the ATM service categories, their quality of service (QoS) parameters, and the traffic descriptors required by the service category during call establishment.

The constant bit rate CBR (or deterministic bit rate) service category provides a very strict QoS guarantee. It is targeted at real-time application, such as voice and raw video, which mandatesevere destrictictions on delay, delay variance (jitter) and cell lose rate. The only traffic descriptors required by the CBR service are the peak cell rate and the cell delay variation tolerance. A fixed amount of bandwidt, determined primarily by the call's peak cell rate, is reserved for CBR connection.

The rea-time variable bit rate VBR-rt (or statistical bit rate SBR) service category is intended for real time bursty applications (e.g., compressed video), which also require strict QoS guarantees. The primary difference between CBR and VBR-rt is in the traffic descriptors they use. The VBR-rt service requires the specification of the sustained (or avarage) cell rate and burst tolerence (i.e., burst length) in addition to the peak cell rate and the cell delay variation tolerence. The ATM Forum also defines a non-real time VBR-nrt service category, in which cell delay variance is not guaranteed.

The avaliable bit rate (ABR) service category is defined to exploit the network's unutilized bandwidth. It is intended for non-real time data applications in which the source is amenable to enforced adjustment of its transmission rate. A minimum cell rate is reserved for the ABR connection and therefore guaranteed by the network. When the network has utilized bandwidth, ABR sources are allowed to increase their cel rates up to an allowed cell rate (ACR), a value which is periodically updated by the ABR flow control mechanism. The value of ACR always falls between the minimum and the peak cell rate for the connection and is determined by the network.

The ATM Forum defines another service category for non-real-time applications called the unspecified bit rate (UBR) service category. UBR service is entirely best effort; the call is provided with no QoS guarantees. The UTI-t also defines an additional service category for non-real-time data applications. The ATM block transfer (ABT) service category is intended for transmission of short bursts, or blocks, of data. Before transmitting a block, the source request a reservation of bandwidth from the network. If the ABT service is being used with immediate transmission option (ABT/IT); the block of data is sent at the same time as the reservation request. If bandwidth is not avaliable for transporting the block, then it is simply discarded, and source must retransmit it. In the ABT service with delayed transmission (ABT/DT), the source waits for confirmation from the network that enough bandwidth is avaliable before transmitting the block of data. In both case, the network temporarily reserves bandwidth according to the peak cell rate for each block. Immediately after transporting the block, the network releases the reserved bandwidth.

ITU-T Service Categories →	DBR	SBR	ABT	ABR	
ATM Forum → Service Categories	CBR	VBR-rt	VBR-nrt	ABR	UBR
Cell Loss Rate	specified				unspecified
Cell Transfer Delay	specified			unspecified	
Cell Delay Variation	specified		unspecified		
Traffic Descriptors	PCR/CDVT	PCR/CDVT SCR/BT		PCR/CDVT MCR/ACR	PCR/CDVT

PCR = Peak Cell Rate SCR = Sustained Cell Rate

CDVT = Cell Delay Variation Tolerance BT = Burst Tolerance

MCR = Minimum Cell Rate ACR = Allowed Cell Rate

Table 2.2.2.1 ATM layer service categories

2.3 ATM Adaptatiton Layer

The ATM adaptation layer (AAL), which resides atop the ATM layer, is responsible for mapping the requirements of higher layer protocols onto the ATM network. The purpose of the AAL is to provide a link between the services required by higher network layers and the generic ATM cells used by the ATM layer. It operates in ATM devices at the edge of the ATM network and is totally absent in ATM switches. The adaptation layer is divided into two sublayers: the convergence sublayer (CS), which performs error detection and handling, timing and clock recovery; and the segmentation and reassembly (SAR) sublayer, which performs segmentation of convergence sublayer protocol data units (PDUs) into ATM cell-sized SAR sublayer service data units (SDUs) and vice versa.

In order to support different service requirements. the ITU-T has proposed four AAL-specific service classes. Table 3 depicts the four service classes defined in recommendation I.362 [1]. Note that while these AAL service classes are similar in many ways to the ATM layer service categories defined in the previous section, they are not the same; each exists at a different layer of the protocol reference model, and each requires a different set of functions.

AAL service class A corresponds to constant bit rate (CBR) services with a timing relation required between source and destination. The connection mode is connection-oriented. CBR audio and video belong to this class. Class B correspons to variable bit rate (VBR) services. This class also requires timing between source and destination, and its mode is connecting –oriented. VBR audio and video examples of class B services. Class C also corresponds to VBR connection-oriented services, but the timing between soure and destination needs not be related. Class includes connection-oriented data transfer such as X.25, signaling and future high speed data services. Class D corresponds to connectionless services. Connectionless data services such as those supported by LAN's and MAN's are examples of class D services.

Four AAL types (Types 1,2,3/4 and 5),each with a unique SAR sublayer and CS sublayer, are defined to support the four service classses. AAL Type 1 suports constant bit rate services (classA). and AAL Type 2 supports variable bit rate services with a timing relation between source and destination (clas B). AAL Type ¾ was origanally specified as two different AAL types (Type 3 and Type 4), but due to their inherent similarities they were eventuaaly merged to support both class C and class D services. AAL Type 5 also supports class C and class D services.

2.3.1 AAL Type 5

Currently the most widely used adaptation layer is AAL Type 5. AAL Type 5 supports connection-oriented and connectionless services in which there is no timing relation between source and destination (classes C and D). It's functionality was intentionally made simple in order to support high speed data transfer. AAL Type 5 assumes that the layers above to ATM adaption layer can perform error recovery, retransmision and sequence membering when required, and thus, it does not provide these funtions. Therefore only non-assumed operation is provided; lost or corrupted AAL Type 5 packets will not be corrected by retransmission.

Figure 2.3.1.1 depicts the SAR-SDU format for ALL Type 5 (5,6). The SAR sublayer of ALL Type 5 performs segmentation of a CS-PDU into a size suitable for the SAR-SDU payload. Unlike other AAL Types , type 5 devotes to entire 48-octed payload of the ATM cell to the SAR-SDU; there is no overhead. An AAL specific flag (End of frame) in the ATM Payload Type (PT) field of the cell header is set when yhe last cell of a CS-PDU is sent. The reasembly of CS-PDU frames at the destination is controlled by using this flag.

Cell Header	SAR-SDU Payload

Figure 2.3.1.1 SAR-SDU format for AAL Type 5

Figure 2.3.1.2 depicts the CS-PDU format for AAL Type 5 (5,6). It contains the user data payload, along with any necassary padding bits (PAD) and a CS-PDU trailer, which are added by the CS sublayer when it receives the user information from the higher layer. The CS-PDU is padded using 0 to 47 bytes of PAD field to make the length of the CS-PDU an integral multiple of 48 bytes (the size of the SAR-SDU payload). At the receiving end, a reassembled PDU is the passed to the CS sublayer from the SAR sublayer, and CRS values are then calculate and compared. If there is n error, the PAD field is removed by using the value of length field (LF) in the CS-PDU trailer, and user data is passed to the higher layer. If no error is detected, the erroneous information is either delivered to the user or discarded according to the user's choice.

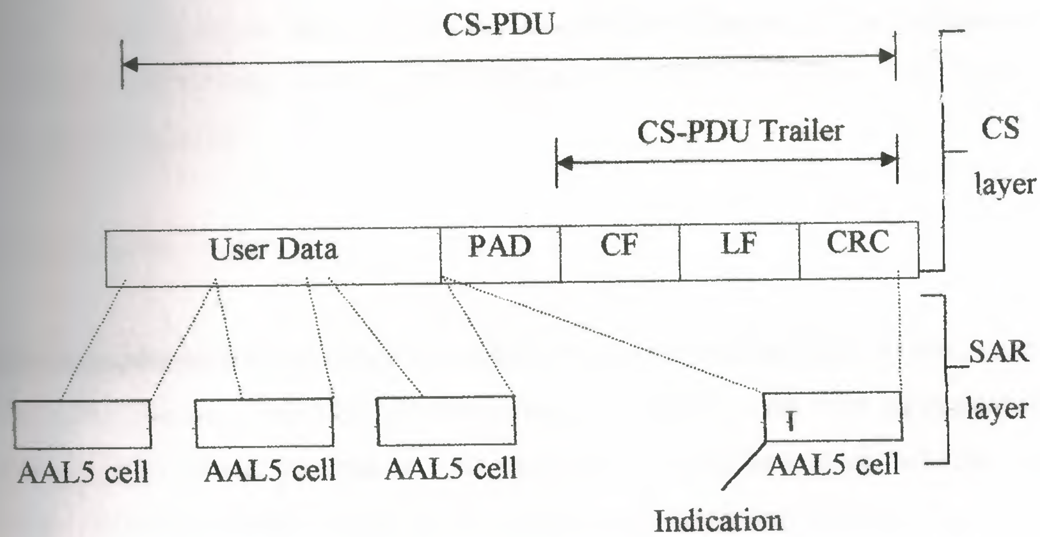


Figure 2.3.1.2 CS-PDU format, segmentation and reassembly of AAL Type 5

2.3.2 AAL Type 1

AAL Type 2 supports constants bit rate services with a fixed timing relation between source and destination users (class A). At the SAR sublayer, it defines a 48-octet service data unit (SDU), which contains 47 octets of user payload, 4 bits for a sequence number, and a 4-bit CRC value to detect errors in the sequece number field. AAL Type 1 performs the following services at the CS sublayer: forward error correction to ensure high quality of audio and video applications, clock recovery b monitoring the buffer filling, explicit time indication by inserting a time stamp in the CS-PDU, and handling of lost and misinserted cels which are recognized by the SAR. At the time of the writing, the CS-PDU format has not been decided.

2.3.3 AAL Type 2

AAL Type 2 supports variable bit rate services with a timing relation between source and destination (class B). AAL Type 2 is nearly identical to AAL Type 1, except that it transfers service data units at a variable bit rate, not a constant bit rate. Furthermore, AAL Type 2 accepts variable length CS-PDUs, and thus, there may exist some SAR-SDUs which are not completely filled with user data. The CS sublayer for AAL Type 2 perform the follow functions: forward error corecion for audio and video services, clock recovery by inserting a time stamp in the CS-PDU, and handling of lost andmisinserted cells. At the time of writing, both the SAR-SDU and CS-PDU formats for AAL Type 2 are still under discussion.

2.3.4 AAL Type ¾

Aal Type ¾ mainly supports services that require no timing relation between the source and destination (classes C and D). At the sublayer, it defines a 48-octet services data unit, with 44 octet of user payload, a 2-bit payload type field to indicate whether the SDU is the beginning, middle, or end of a CS-PDU, a 4-bit cell sequence number, a 10-bit multiplexing identifier that allows several CS-PDUs to be multiplexed over a single VC, a 6-bit cell payload lenght indicator, and a 10-bit CRC code that covers the payload. The CS-PDU format allows for up 65535 octets of user payload and contains a header and trailer to delineate the PDU.

The functions that AAL Type ¾ performs include segmentation and reassembly of variable-lenght user data error handling. It supports message mode (for framed data transfer) as well as streaming mode (for streamed data trnasfer). Since Type ¾ mainly intended for data services, it provdes a retransmission mechanism if necessary.

CHAPTER 3

3.1 Cell Structure of ATM

ATM provides a good bandwith flexibility and can be used efficiently from desktop computers to local area and wide area networks. As itis shown Figure 3.1. All packets are of fixed length 53 bytes (5 bytes for header and 48 bytes for information). No processing like error control is done on the information field of ATM cells inside the network and it is carried transparently in the network.

Header of Cell [5 Bytes]	Payload of Cell [48 Bytes]

Figure 3.1 A typical ATM Cell of 53 bytes

The main characteristics of ATM can summarized as follows:

1- ATM provides cell sequence integrity. That is cells arrive at the destination in the same order as they left the source. This may not be the case with other packet-switched networks.

2- Cells are much smaller than standard packet switched networks. This reduces the value of delay variance, making ATM acceptable for timing sensitive information like voice.

3- The quality of transmission links has lead to the omission of overheads, such as error correction, in order to maximize efficiency.

4- There is no space between cells. At times when the network is idle, unassigned cells are transported.

The structure of the cell is important for the overall functionality of the ATM network. A large cell gives a better payload to overhead ratio, but at the expense of longer, more variable delays. Shorter packets overcome this problem, however the amount of information carried per packet is reduced. An agreement between these two conflicting requirements was reached, and a standard cell format chosen. The ATM cell consists of a 5-ocet header and a 48-octet information field after the header.

information contained in the header is dependent on whether the cell is carrying rmation from the user network to the first ATM public exchange (User-Network erface - UNI), or between ATM exchanges in the trunk network (Network-Node erface - NNI). The formats of the two types of header are shown below Figure 3.2.

8

1

GFC	VPI	
VPI	VCI	
VCI		
VCI	PT	CLP
HEC		

Octet 1

Octet 5

Bit 1 Bit 8

VPI		
VPI	VCI	
VCI		
VCI	PT	CLP
HEC		

User-Network Interface (UNI) **Network-Network**
erface (NNI)

Figure 3.2 Structure of UNI and NNI

e VPI is the first field in the ATM header at the NNI, and the second field in the der at the UNI. This value identifies the virtual path used for this connection. The ameters associated with the VPI include bandwidth and input or output port used. A h may consist of several channels, each carrying a portion of the total bandwidth ocated tothat path. The virtual path identifier is a decimal number in the range 0-4095 hin the NNI and 0-255 in the UNI.

e VCI is the second field in the ATM header at the NNI and the third at the UNI. The ameters associated with the VCI include bandwidth, VPI, in and out portsand in/out CI. The Virtual Channel Identifier is a decimal number in the range 0-65535.

th the VPI and VCI may be used to route traffic through the switch. In some tances, cells are switched on the VPI value only.

s important to note that VPIs and VCIs are unique on a link-by-link basis. In other rds, Virtual Path 10 on switch A is not same as Virtual Path 10 on switch B. Further, tual Path 10 on switch A, Port 1a1, is not the same as Path 10 on switch A, port 1a2. e VPI and VCI values are interpreted at each switch and used to determine the output nk, and outgoing VPI and VCI values. The fields within an ATM cell at the UNI level re defined as Figure 3.2.

GFC (Generic Flow Control)

The first four bits of byte 1 in the header are defined as GFC and currently they are not used. Exact mechanisms for flow control are still under development, and no explicit definition for this field exists at this time.

VPI (Virtual Path Identifier)

The last four bits of byte 1 and the first four bits of byte 2 are reserved for the VPI.

VCI (Virtual Channel Identifier)

The second half of byte 2, all of byte 3 and first half of byte 4 are reserved for the VCI.

PTI (Payload Type Indicator)

The next three bits of byte 4 are reserved for the PTI. PTI is used to indicate the type of information carried in the cell. Codes within this field are defined as: values 0-3 identify various type of user data, values 4 and 5 indicate management information, value 6 and 7 are reserved for future definition. These reserved values are expected to be used in the future implementations of a flow control algorithm called BECN.

CLP (Cell Loss Priority)

CLP is contained in the last bit of byte 4. If set, this bit indicates the eligibilityof the cell for discarding under congested conditions. Currently, this bit is not used until a clear definition of who sets and decides whether this bit should be set is determined.

HEC (Header Error Check)

The last byte of the header (byte 5) contains a header error control. This error-correction is calculated across the previous four bytes of the header. It is designed to detect multiple header errors and to correct single bit errors. It provides protection against mis-delivery of cells due to address errors. It does not contain any indication of the quality or integrity of the data within the payload field.

Payload

The remaining 48 bytes of the cell contain user information. The ATM adaptation layer (AAL) accomplishes inserting data into the payload field. It should be noted that depending on the AAL proces, not all-48 bytes would contain user information. Up to four bytes may be the AAL itself.

VPI		
VPI	VCI	
VCI		
VCI	PTI	CLP
HEC		
PAYLOAD 48 Bytes		

Figure 3.3 ATM NNI Cell Format

The cell structure at the NNI level is essentially the same as the UNI. The only notable difference is the GFC field, which is eliminated, and four bits of byte 1 are pretended to the VPI field (Figure3.3). For this reason, at the NNI levle the range of values avaliable for the VPI is 0-4095. At the UNI level the range is from 0-255.

.2 ATM QoS Classes

ATM guarantees the quality of service of the connection that has been established, this s one of the most important properties of the ATM technology. The ATM forum has lefined four Atm layer service classes, each with scalable quality of service (QoS) evels:

Class A:

Class A, or constant bit rate (CBR), traffic is characterized a continuous stream of bits t a steady rate, such as time division multiplexing (TDM) multiplexer traffic. Class A raffic is low-bandwidth traffic that is highly sensitive to delay and intolerant to cell oss.

lass B and C:

lass B and C traffic, defined as VBR, has a bursty nature and can be characterized by oice or video applications that use compression. Class B traffic is real-time VBR, (RT-BR) , where end-to-end delay is critical, such as interactivevideo conferencing. Class is non-real time (VBR-NRT) traffic, where delay is not so critical, such as video ayback, training tapes and video mail messages.

lass D:

ass D traffic is split into two classes: ABR and UBR. These classes are for bursty AN traffic and data that is more tolerant of delays and cell lose. UBR is a 'best effor' rvice that does not specify bit rate or traffic parameters and has no quality of service arantees. Originally devised as way to make use of exceesbandwith, UBR is sun-bject increased cell loss and the discard of whole packets. ABR, like UBR, is also a best fort service, but differs in that it is a managed service, based on MCR and with a low loss. No delay variation guarantee is currently envisioned for either UBR or ABR rvice classes.

ATM Switches for the Market

e most important component of an ATM network is the switches. They are based on e switching architectures of the 1970s and 1980s, which have also been proposed for ecommunication switching gear or as the interconnect fabric of multiprocessor stems. Various objectives, such as blocking, routing, performance, and VLSI nplementation, motivated past work on the switches. Although these attributes are sential, they don't address many concerns important to the commercial ATM market, ich as the general cost of ownership and incremental deployment. In a world with 20 illion computers connected by over one million Ethernets, coexistence with present AN/WAN technologies, stability, reliability, and efficient utilization of bandwidth are rucial.

CHAPTER 4

4. ATM Traffic Management

One of the primary benefits of ATM networks is that they can provide users with guaranteed Quality of Service (QoS). To do this, the user must inform the network, upon connection set-up, of both the expected natures of the traffic that will be sent along connection, and of the type of quality of service that the connection requires.

ATM networks offer a specific set of service classes, and at connection set-up, the user must request a specific service class from the network for that connection. ATM networks to differentiate between specific types of connection use service classes, each with a particular mix of traffic and QoS parameters, since such traffic may need to be differentiated within the network, for instance, by using priorities to allow for the requested behaviour. The current set of QoS classes, which the ATM Forum is defining for UNI 4.0, is as follows.

1- Continuous Bit Rate [CBR]: End systems would use CBR connection types to carry constant bit rate traffic with a fixed timing relationship between data samples, typically for circuit emulation.

2- Variable Bit Rate-Real Time [VBR (rt)]: The VBR (rt) service class is used for connections that carry variable bit rate traffic, in which there is a fixed timing relationship between samples; for example, variable bit rate video compression.

3- Variable Bit Rate-Non-Real Time [VBR (nrt)]: The VBR (nrt) service class is used for connections that variable carry variable bit rate traffic in which there is no timing relationship between data samples, but guarantee of QoS (on bandwidth or latency) is still required. Such a service class might be used in Frame Relay internetworking.

4- Available Bit Rate [ABR]: The ATM Forum is currently focusing its work on the ABR service, ABR supports variable data transmissions and does not preserve any timing relationships between source and destination. Unlike VBR (nrt) service, however, the ABR service does not provide any guaranteed bandwidth to the user. Rather, the network provides a "best effort" service, in which feedback (flow control echanism) is used to increase the bandwidth available to the user. The ATM Forum is ırrently working on a "Rate Based" mechanism for the ABR congestion control.

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

ın ATM connection that is set up with specified traffic descriptors constitutes a traffic ontract between the user and the network. The network offers the type of guarantee ppropriate to the service class, as long as the user keeps the traffic on the connection vithin the envelope defined by the traffic parameters. The network can enforce the raffic contract by a mechanism known as usage parameter control (UPC), better known s traffic policing.

.1. ATM Traffic Control

The control of ATM traffic is complicated due to ATM's high link speed and small cell size, the diverse service requirements of ATM applications, and the diverse characteristics of ATM traffic. Furthermore the configuration and size of the ATM environment either local or wide area, has a significant impact on the choice of traffic control mechanisms.

The factor, which most complicates traffic control in ATM, is its high link speed. Typical ATM link speeds are 155.52 Mbit/s and 622.08 Mbit/s. At these high link speeds, 53 byte ATM cells must be switched at rates greater than one cell per 2.726 μs or 0.682 μs respectively. It is apparent that the cell processing required by traffic control must perform at speeds comparable to these cell-switching rates. Thus, traffic control should be simple and efficient, without excessive software processing.

Such high speed renders many traditional traffic control mechanisms inadequate for use in ATM due to their reactive nature. Traditional reactive traffic control mechanisms attempt to control network congestion by responding to it after it occurs and usually involves sending feedback to the source in the form of a choke packet. However, a large ropagation-bandwidth product (i.e., the amount of traffic that can be sent in a single ropagation delay time) renders many reactive control schemes ineffective in high-peed networks.

Vhen a node receives feedback, it may have already transmitted a large amount of data. Consider across continental 622 Mbit/s connection with a propagation delay of 20 ms propagation-bandwidth product of 12.4 Mbits). If a node at one end of the connection xperiences congestion and attempts to throttle the source at the other end by sending it feedback packet, the source will already have transmitted over twelve megabits of nformation before feedback arrives. This example illustrates the ineffectiveness of raditional reactive traffic control mechanisms in high-speed networks and argues for ovel mechanisms that take into account high propagation-bandwidth products.

Not only is traffic control complicated by high speeds, but it is made more difficult by ne diverse quality of service (QoS) requirements of ATM applications. For example, nany applications have strict delay requirements and must be delivered within a pecified amount of time. Other applications have strict loss requirements and must be elivered reliable without an inordinate amount of loss. Traffic controls must address ne diverse requirements of such applications.

Another factor complicating traffic control in ATM networks is the diversity of ATM raffic characteristics. In ATM networks, continuous bit rate traffic is accompanied by ursty traffic. Bursty traffic generates cells at a peak rate for very short period of time nd then immediately becomes less active, generating fewer cells. To improve the fficiency of ATM network utilization, bursty calls should be allocated an amount of andwidth that is less then their peak rate. This allows the network to multiplex more alls by taking advantage of the small probability that a large number of bursty calls vill be simultaneously active. This type of multiplexing is referred to as statistical nultiplexing. The problem then becomes one of determining how best to statistically nultiplex bursty calls such that the number of cells dropped due to excessive burrstones s balanced with the number of bursty traffic streams allowed. Addressing the unique emands of bursty traffic is an important function of ATM traffic control.

r the reasons mentioned above, many traffic control mechanisms developed for isting networks may not be application to ATM networks, and therefore novel forms traffic control are required. One such class of novel mechanisms that work well in gh-speed networks falls under the heading of preventive control mechanisms. reventive control attempts to manage congestion by preventing it before it occurs. reventive traffic control is targeted primarily at real-time traffic. Another class of affic control mechanisms has been targeted toward non-real-time data traffic and relies n novel reactive feedback mechanisms.

.2. Preventive Traffic Control

Preventive control for ATM has two major components: call admission control and isage parameter control. Admission control determines whether to accept or reject a iew call at the time of call set-up. This decision is based on the traffic characteristics of he new call and the current network load. Usage parameter control enforces the traffic parameters of the call once it has been accepted into the network. This enforcement is necessary to insure that the call's actual traffic flow conforms with that reported during call admission. Before describing call admission and usage parameter control in more detail, it is important to first discuss the nature of multimedia traffic. Most ATM traffic belongs to one of two general classes of traffic: Continuous traffic and bursty traffic. Source of continuous traffic (e.g., constant bit rate video, voice without silence detection) are easily handled, because their resource utilization is predictable and they can be deterministically multiplex. However, bursty traffic (e.g.. voice with silence detection, variable bit rate video) is characterized by its unpredictability, and it is this kind of traffic, which complicates preventive traffic control.

Burrstiness are a parameter descrying how densely or sparsely cell arrivals occur. There are a number of ways to express traffic burrstones, the most typical of which are the ratio of peak bit rate to average bit rate, and the average burst length. Several other measures of burrstones have also been proposed. It is well known that burrstones plays a critical role in determining network performance, and thus, it is critical for traffic control mechanisms to reduce the negative impact of bursty traffic.

. Call Admission Control

l admission control is the process by which the network decides whether to reject a / call. When a new call requests access to the network, it provides a set of traffic criptors (e.g., peak rate, average rate, average burst length) and a set of quality of vice requirements (e.g., acceptable cell loss rate, acceptable cell delay variance, eptable delay). The network then determines, through signaling, if it has enough ources (e.g., bandwidth, buffer space) to support the new call's requirements. If it s, the call is immediately accepted and allowed to transmit data into the network. erwise it is rejected. Call admission control prevents network congestion by limiting number of active connections in the network to a level where the network resources adequate to maintain quality of service guarantees.

of the most common ways for an ATM network to make a call admission decision use the call's traffic descriptors and quality of service requirements to predict the uivalent bandwidth" required by the call. The equivalent bandwidth determines how y resources need to be reserved by the network to support the new call at its iested quality of service. For continuous, constant bit rate calls, determining the valent bandwidth is simple. It is merely equal to the peak bit rate of the call. For t connections, however, the process of determining the equivalent bandwidth should into account such factors as call's burrstones ratio (the ratio of peak bit rate to age bit rate), burst length, and burst inter arrival time. The equivalent bandwidth for t connections must be chosen carefully to ameliorate congestion and cell loss while imizing the number of connections that can be statistically multiplexed.

Usage parameter control

admission control is responsible for admitting or rejecting new calls. However, call ission by itself is ineffective if the call does not transmit data according to the ic parameters it provided. Users may intentionally or accidentally exceed the traffic meters declared during call admission, thereby overloading the network. In order to ent the network users from violating their traffic contracts and causing the network to enter a congested state, each call's traffic flow is monitored and, if necessary, restricted. This is the purpose of usage parameter control.

To efficiently monitor a call's traffic, the usage parameter control function must be located as close as possible to the actual source of the traffic. An ideal usage parameter control mechanism should have the ability to detect parameter-violating cells, appear transparent to connections respecting their admission parameters, and rapidly respond to parameter violations. It should also be simple, fast and cost effective to implement in hardware. To meet these requirements several mechanisms have been proposed and implemented.

The leaky bucket mechanism is a typical usage parameter control mechanism used for ATM networks. It can simultaneously enforce the average bandwidth and the burst factor of a traffic source. One possible implementation of the leaky bucket mechanism is to control the traffic flow by means of tokens. A conceptual model for the leaky bucket mechanism is illustrated in Figure 4.1.

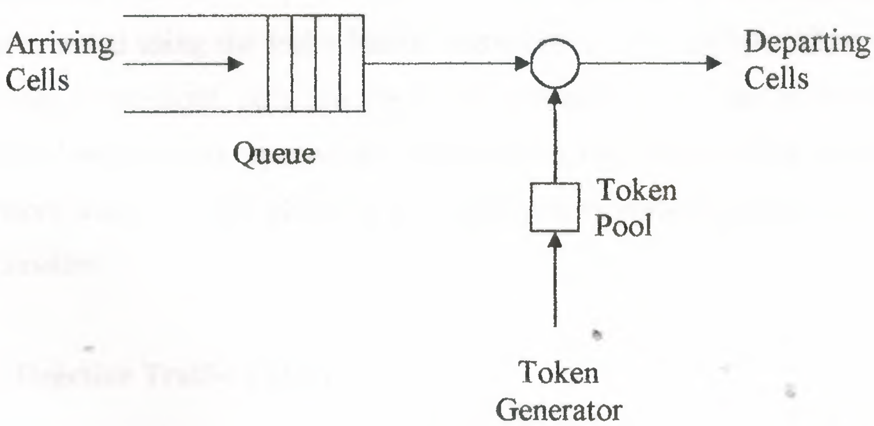


Figure 4.1. Leaky bucket mechanism

In Figure 4.1, an arriving cell first enters a queue. If the queue is full, cells are simply discarded. To enter the network, a cell must first obtain a token from the token pool; if there is no token a cell must wait in the queue until a new token is generated. Tokens are generated at a fixed rate corresponding to the average bit rate declared during call admission. If the number of tokens in the token pool exceeds some predefined threshold value, token generation stops. This threshold value corresponds to the burstiness of the transmission declared at call admission time; for larger threshold values, a greater degree of burstiness is allowed. This method enforces the average input rate while allowing for a certain degree of burstiness.

One disadvantage of the leaky bucket mechanism is that the bandwidth enforcement introduced by the token pool is in effect even when the network load is light and there is no need for enforcement. Another disadvantage of the leaky bucket mechanism is that it may mistake non-violating cells for violating cells. When traffic is bursty, a large number of cells may be generated in a short period of time, nevertheless conforming to the traffic parameters claimed at the time of call admission. In such situations, none of these cells should be considered violating cells. Yet in actual practice, leaky bucket may erroneously identify such cells as violations of admission parameters. To overcome these disadvantages, a virtual leaky bucket mechanism (also referred to as a marking method) has been proposed. In this mechanism violating cells rather than being discarded or buffered, are permitted to enter the network at a lower priority (CLP=1). These violating cells are discarded only when they arrive at a congested node. If there are no congested nodes along the routes to their destinations, the violating cells are transmitted without being discarded. The virtual leaky bucket mechanism can easily be implemented using the leaky bucket method described earlier. When the queue length exceeds a threshold, cells are marked as "droppable" instead of being discarded. The virtual leaky bucket method not only allows the user to take advantage of a light network load, but also allows a larger margin of error in determining the token pool parameters.

4.5. Reactive Traffic Control

Preventive control is appropriate for most types of ATM traffic. However there are cases where reactive control is beneficial. For instance, reactive control is useful for service classes like ABR, which allow sources to use bandwidth not being utilized by calls in other service classes. Such a service would be impossible with preventive control, because the amount of unutilized bandwidth in the network changes dynamically, and the sources can only be made aware of the amount through reactive feedback.

There are two major classes of reactive traffic control mechanisms: rate-based and credit-based. Most rate-based traffic control mechanisms establish a closed feedback loop in which the source periodically transmits special control cells, called resource management cells, to the destination (or destinations). The destination closes the feedback loop by returning the resource management cells to the source. As the feedback cells traverse the network, the intermediate switches examine their current congestion state and mark the feedback cells accordingly. When the source receives a returning feedback cell, it adjusts its rate, either by decreasing it in the case of network congestion, or increasing it the case of network underutilization. An example of a rate-based ABR algorithm is the Enhanced Proportional Rate Control Algorithm (EPRCA), which was proposed, developed, and tested through the course of ATM Forum activities.

Credit-based mechanisms use link-by-link traffic control to reduce loss and optimize utilization. Intermediate switches exchange resource management cells that contain "credits", which reflect the amount of buffer space available at the next downstream switch. A source cannot transmit a new data cell unless it has received at least one credit from its downstream neighbor. An example of a credit-based mechanism is the Quantum Flow Control (QFC) algorithm, developed by a consortium of researchers and ATM equipment manufacturers.

CHAPTER 5

5. Switch Design Principles

From the preceding section, it can be seen that each design alternative has its own merits, drawbacks, and considerations. The general design principles and issues exposed in the last section are analyzed in more detail here. The main points those are important:

5.1 Internal Blocking

A fabric is said to be internally blocking if a set of N cells addressed to N different outputs can cause conflicts within the fabric. Internal blocking can reduce the maximum possible throughput. Banyan networks are blocking, while TDM buses where the bus operates at least N times faster than the port speed are internally non-blocking. By the same concept, shared memory swiches which can read and write at the rate of NV(N*V) cells per second are interal blocking, since if N cells arrive for N different outputs, no conflicts will occur.

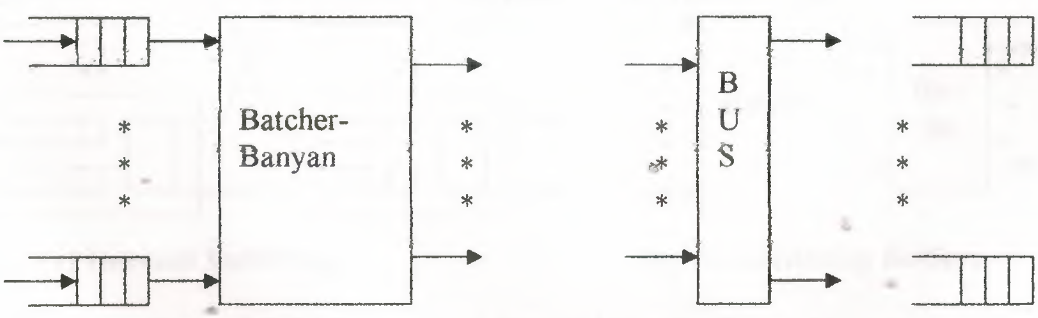
Hence, to prevent internal blocking, shared resources must operate at some factor greater than the port speed. Applying this to banyan networks, is discussed, the internal links need to run square root of N times faster than the highest speed incoming link (12). This foctor limits the scalability and throughput of the switch. Coppo (13) have developed a mathematical model for analyzing the preventing the cells behind it from being admitted. Thus capacity is wasted. Several methods have been proposed to tackle the head-of-the-line blocking problem, but they all exhibit complex design. Increasing the intemal speed of the space division fabric by a factor of four, or changing the FIFO discipline are two examples of such methods.

. 2 Buffering Approaches

uffering is necessary in all design approaches. For instance, in a banyan network, if vo cells addressed to the same output successfully reach the last switching stage at the ame time, output contention occurs and must be resolved by employing buffering. The ocation and size of buffers are important issues that must be decided. There are four asic approaches to the placement of buffers. These basic approaches are illustrated in igure 5.2.1.

.2.1. Internal Queuening

uffers at the input of an internally non blocking space division fabric (such as Batcher anyan network) illustrate this type of buffering. This approach suffers from head-of- ıe-line blocking. When two cells arrive at the same time and are destined to the same utput, one of them must wait in the input buffers preventing the cells behind it from eing admitted. Thus capacity is wasted. Several methods have been proposed to tackle ıe head-of-the-line blocking problem, but they all exhibit complex design. Increasing ıe internal speed of the space division fabric by a factor of four, or changing the FIFO iscipline are two examples of such methods as seen in Figure 5.2.1.



(a) Input Buffering **(b) Output Buffering**

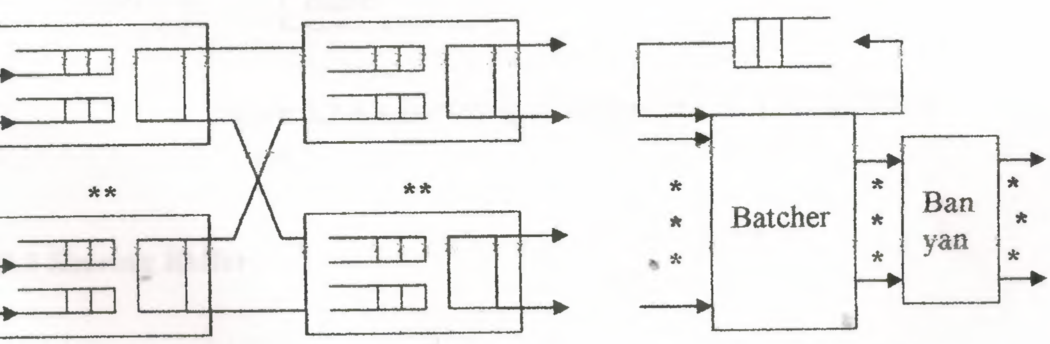
Figure 5.2.1 Input and Output Buffering

.2.2. Output Queuening

'his type of buffering can be evident by examining the buffers at the output ports of a hared bus fabric. This approach is optimal in terms of throughput and delays, but it eeds some means of delivering muitiple celis per celi time to any output. Hence, either he output buffers must operate at some factor times the port speed, or there should be nuitiple buffers at each output. in both cases, the throughput and scalability are limited, ither by the speedup factor or by the number of buffers, Figure 2.4(b)..

.2.3. Internal Queueing

Buffers can be placed within the switching elements in a space division fabric. For nstance, in a banyan network, each switching element contains buffers at its inputs to tore cells in the event of conflict. Again, head-of-the-line blocking might occur within he switching elements, and this significantly reduces throughput, especially in the case of small buffers or larger networks. Internal buffers also introduce random delays within he switch fabric, causing undesirable cells delay variation as seen in Figure 5.2.3.1.



(a) Internal Buffering **(b) Recurcilating Buffers**

Figure 5.2.3.1 Internal and Recurcilating Buffering Mechanism

5.2.4. Recirculated Queuing

This technique allows cells to re-enter the internally non blocking space division network. This is needed when more than one cell is addressed to the same output simultaneously, so the extra cells need to be routed to the inputs of the network through recirculating buffers. Although this approach has the potential for achieving the imal throughput and delay performance of output queueing, its implementation fers from two major complexities. First, the switching network must be large enough accommodate the recirculating cells. Second, a control mechanism is essential to quentially order the cells, Figure 5.2.4.1shows the buffering strategies in ATM itches below.

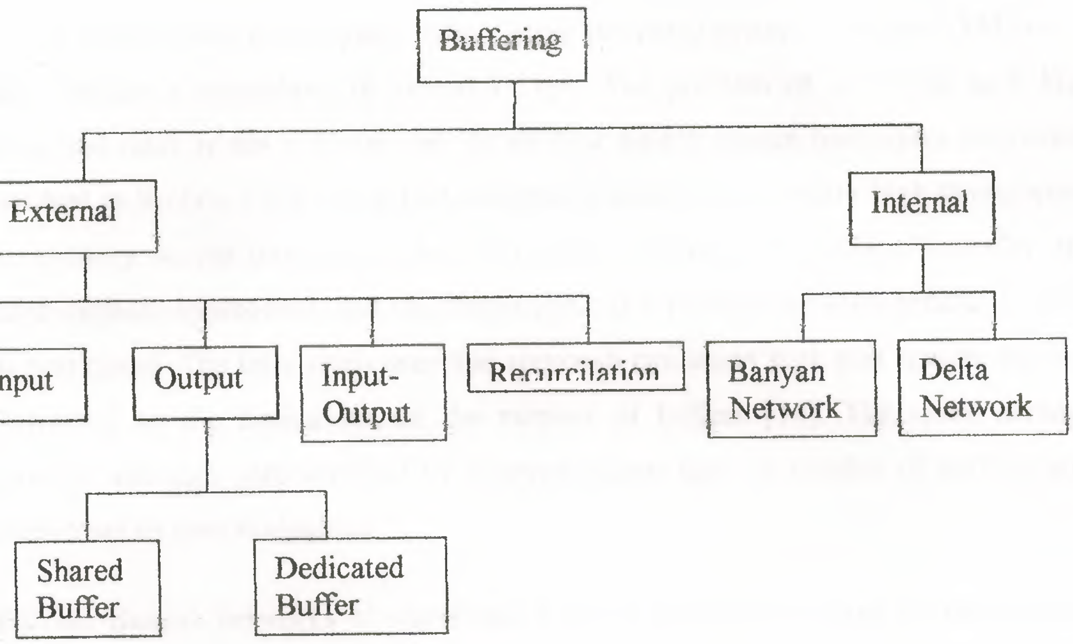


Figure 5.2.4.1 Buffering strategies tree in ATM switches.

3 Sharing Buffer

ne number and size of buffers has a significant impact on switch design. In shared emory switches, the central buffer can take full advantage of statistical sharing, thus sorbing large traffic bursts to any output by giving it as much as is available of the ared buffer space. Hence, it requires the least total amount of buffering. For a random iiform traffic and large values of N, a buffer space of only 12 N cells is required to hieve a cell loss rate of 10^{-9}, under a load of 0.9. For a TDM bus fabric with N output iffers, and under the same traffic assumptions as before, the required buffer space is out 90 N cells. Also a large traffic burst to one output cannot be absorbed by the other output buffers, although each output buffer can statistically multiplex the traffic from the N inputs. Thus buffering assumes that it is improbable that many input cells will be directed simultaneously to the same output. Neither statistical multiplexing between outputs or at any output can be employed with fully interconnected fabrics with N squared output buffers. Buffer space grows exponentially in this case.

.4 Scalability of Switch Fabrics

f ATM switches will ever replace today's large switching systems, then an ATM switch vould require a throughput of almost 1 Tbps. The problem of achieving such high hroughput rates is not a trivial one. In all four switch design techniques previously escribed in Section 2.2.2.1, it is technologically infeasible to realize high throughputs. he memory access time limits the throughput attained by the shared memory and hared medium approaches, and the design exhibits a tradeoff between number of ports nd port speed. The fully interconnected approach can attain high port speeds, but it is onstrained by the limitations on the number of buffers [14]. The space division pproach, although unconstrained by memory access time or number of buffers, also iffers from its own limitations:

Batcher-Banyan networks of significant size are physically limited by the possible rcuit density and number of input/output pins of the integrated circuit. To interconnect veral boards, interconnection complexity and power dissipation place a constraint on e number of that can be interconnected

The entire set of N cells must be synchronized at every stage.

Large sizes increases the difficulty of reliability and repairability.

All modifications to maximize the throughput of space-division networks increase implentation complexity.

us the previous discussion illustrates the infeasibility of realizing large ATM itches with high throughput by scaling up a certain fabric design. Large fabrics can y be attained by interconnecting small switch modules (of any approach) of limited oughput.

5.5 Multicasting

Many services, such as video, will need to multicast an input cell to a number of elected outputs, or broadcast it to all outputs. Designing multicasting capability can be one by either adding a separate copy network to the routing fabric, or designing each nterconnected switching module for multicasting.

hared Medium and Fully Interconnected Output-Buffered Approaches:

ere multicasting is inherentiy natural, since input celis are broadcast anyway, and the ldress filters at the output buffers select the appropriate celis. Thus the address filters n simply filter according to the multicast addresses, in addition to the output port ldress.

ıared Memory Approach:

ulticasting requires additional circuitry in this case. The cell to be multicast can be plicated before the memory, or read several times from the memory. Duplicating cells juires more memory, while reading a cell several times from the same memory ation requires the control circuitry to keep the cell in memory until it has been read all output ports in the multicast groups.

CHAPTER 6

. Hardware Switch Architectures for ATM Networks

ATM networks information is segmented into fixed length cells, and cells are ynchronously transmitted through the network. To match the transmission speed of the etwork links, and to minimize the protocol processing overhead, ATM performs the vitching of cells in hardware switching fabrics, unlike traditional packet switching etworks, where switching is largely performed in software.

large number of designs have been proposed and implemented for ATM switches. hile many differences exist, ATM switch architectures can be broadly classified into o categories: asynchronous time division (ATD) and space-switched architectures.

1 Asynchronous Time Division Switches

TD, or single path, architectures provide a single, multiplexed path through the ATM vitch for all cells. Typically a bus or ring is used. Figure 6.1.1 shows the basic ucture of the ATM switch proposed. In this figure, four input ports are connected to ur output ports by a time-division multiplexing (TDM) bus. Each input port is located a fixed time slot on the TDM bus, and the bus is designated to operate at a eed equal to the sum of the incoming bit rates at all input ports. The TDM slot sizes e fixed and equal in length to the time it takes to transmit one ATM cell. Thus, during e TDM cycle the four input ports can transfer four ATM cells to four output ports.

ATD switches, the maximum throughput is determined by a single, multiplexed path. vitches with N input ports and N output ports must run at a rate N times faster than the nsmission links. Therefore, the total throughput of ATD ATM switches is bounded the current capabilities of device logic technology. Commercial examples of ATD itches are the Fore Systems ASX switch and Digital s Vnswitch.

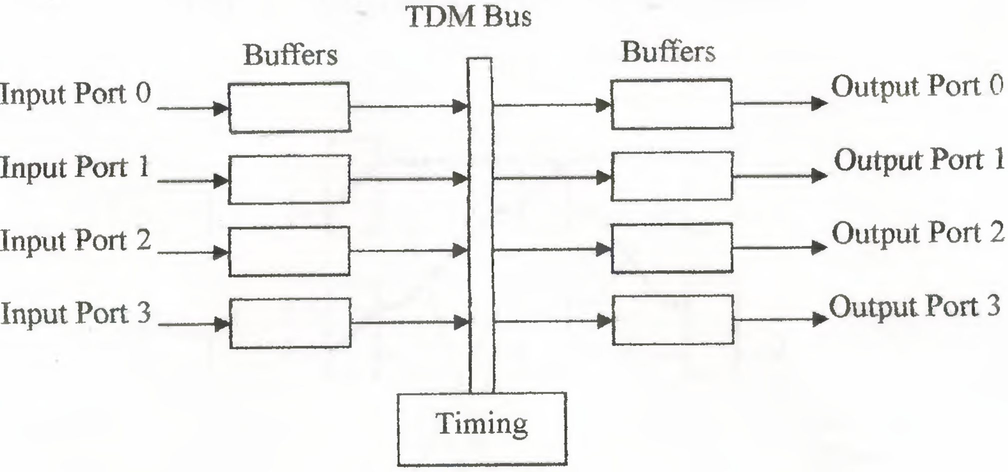


Figure 6.1.1 A 4x4 Asynchronous Time Division Switch

2 Space Division Switches

o eliminate the single path limitation and increase total throughput, space division TM switches implement multiple paths through switching fabrics. Most space-vision switches are based on multi-stage interconnection networks, where small vitching elements (usually 2x2 cross-point switches) are organized into stages and ovide multiple paths through a switching fabric. Rather than being multiplexed onto a ngle path, ATM cells are space- switched through the fabric. Three typical types of ace-division switches are described below.

anyan Switches: Banyan switches are examples of space-division switches An NxN anyan switch is constructed by arranging a number of binary switching elements into veral stages. Figure 6.2.1 depicts an 8x8 self-routing Banyan switch. The switch bric is composed of twelve 2x2 switching elements assembled into three stages. From ny of the eight input ports, it is possible to reach all of the eight output ports. One esirable characteristic of the Banyan switch is that it is self-routing. Since each cross oint switch has only two output lines, only one bit is required to specify the correct utput path. Very simply, if the desired output addresses of an ATM cell is stored in the

:ell header in binary code, routing decisions for the cell can be made at each cross-point ;witch by examining the appropriate bit of the destination address.

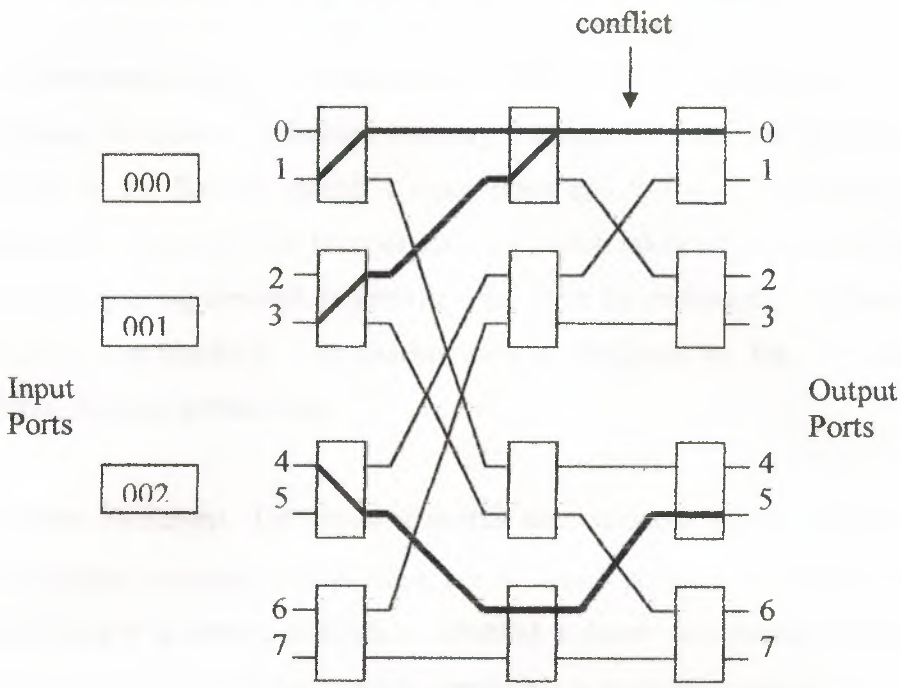


Figure 6.2.1 An 8x8 Banyan switch with binary switching elements

Although the Banyan switch is simple and possesses attractive features such as nodularity, which makes it suitable for VLSI implementation, it also has some isadvantages. One of its disadvantages is that it is internally blocking. In other words, ells destined for different output ports may contend for a common link within the witch. This results in blocking of all cells that wish to use that link, except for one. Hence, the Banyan switch is referred to as a blocking switch. In Figure 6.2.1, three cells re shown arriving on input ports 1, 3 and 4 with destination port addresses of 0, 1, and , respectively. The cell destined for output port 0 and the cell destined for output port 1 nd up contending for the link between the second and third stages. As a result, only ne of them (the cell from input port 1 in this example) actually reaches its destination output port 0), while the other is blocked.

Batcher-Banyan Switches: Another example of space-division switches is the Batcher-Banyan switch. It consists of two multi-stage interconnection networks: a Banyan self-routing network and a Batcher sorting network. In the Batcher-Banyan switch the incoming cells first enter the sorting network, which takes the cells and sorts them into ascending order according to their output addresses. Cells then enter the Banyan network, which routes the cells to their correct output ports.

As shown earlier, the Banyan switch is internally blocking. However the Banyan switch possesses an interesting feature. Namely, internal blocking can be avoided if the cells arriving at the Banyan switch's input ports are sorted in ascending order by their destination addresses. The Batcher-Banyan switch takes advantage of this fact and uses the Batcher soring network to sort the cells, there by making the Batcher-Banyan switch internally non-blocking. The Starlite switch, designed by Bellcore, is based on the Batcher-Banyan architecture.

Crossbar Switches: The crossbar switch interconnects N inputs and N outputs into a fully meshed topology; that is, there are N^2 cross points within the switch. See Figure 6.2.2. Since it is always possible to establish a connection between any arbitrary input and output pair, internal blocking is impossible in a crossbar switch.

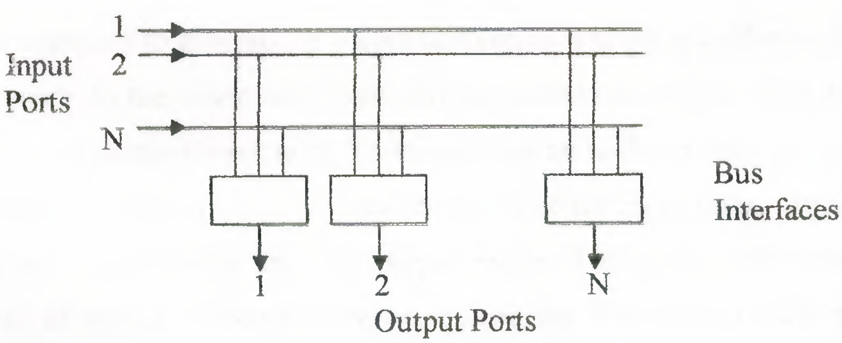


Figure 6.2.2 A Knockout (crossbar) switch

The architecture of the crossbar switch has some advantages. First, it uses a simple two-state cross-point switch open and connected state which is easy to implement. Second, the modularity of the switch design allows simple expansion. One can build a larger switch by simply adding more cross-point switches. Lastly, compared to Banyan-based switches, the crossbar switch design results in low transfer latency, because it has the smallest number of connecting points between input and output ports. One disadvantage o this design, however, is the fact that it uses the maximum number of crosspoints cross point switches) needed to implement an NxN switch.

The Knockout Switch by ATT Bell Labs is a non-blocking switch based on the crossbar design. It has N inputs and N outputs and consists of a crossbar-based switch with a bus interface module at each output. Figure 6.2.2.

.3 Non-blocking Buffered Switches

Although some switches such as Batcher-Banyan and crossbar switches are internally non-blocking two or more cells may still contend for the same output port in a non-blocking switch, resulting in the dropping of all but one cell. In order to prevent such loss, the buffering of cells by the switch is necessary. The buffers may be placed in the inputs to the switch, in the outputs to the switch, or within the switching fabric itself, as a shared buffer. Some switches put buffers in both the input and output ports of a switch.

The first approach to eliminating output contention is to place buffers in the output ports of the switch. In the worst case, cells arriving simultaneously at all input ports can be destined for a single output port. To ensure that no cells are lost in this case, the cell transfer must be performed at N times the speed of the input links, and the switch must be able to write N cells into the output buffer during one cell transmission time. Examples of output buffered switches include the Knockout switch by AT&T Bell Labs, the Siemens&Newbridge MainStreetXpress switches, the ATML's VIRATA switch, and Bay Networks' Lattis switch.

The second approach to buffering in ATM switches is to place the buffers in the input ports of the switch. Each input has a dedicated buffer and cells which would otherwise be blocked at the output ports of the switch are stored in input buffers. Commercial examples of switches with input buffers as well as output buffers are IBM's 8285 Nways switches, and Cisco's Lightstream 2020 switches.

A third approach is to use a shared buffer within the switch fabric. In a shared buffer switch there is no buffer at the input or output ports. Arriving cells are immediately njected into the switch and when output contention happens, the winning cell goes hrough the switch, while the losing cells are stored for later transmission in a shared buffer common to all of the input ports. Cells just arriving at the switch join buffered cells in competition for available outputs. Since more cells are available to select from, t is possible that fewer output ports will be idle when using the shared buffer scheme. Thus, the shared buffer switch can achieve high throughput. However, one drawback is hat cells may be delivered out of sequence, because cells that arrived more recently nay win over buffered cells during contention. Another drawback is the increase in the umber of input and output ports internal to the switch. The Starlite switch with trap by Bellcore is an example of the shared buffer switch architecture. Other examples of hared buffer switches include Cisco's Lightstream 1010 switches, IBM's Prizma witches, Hitachi's switches, and Lucent's ATM cell switches.

.4 Continuing Research in ATM Networks

TM is continuously evolving, and its attractive ability to support broadband integrated ervices with strict quality of service guarantees has motivated the integration of ATM nd existing widely deployed networks. Recent additions to ATM research and chnology include, but are not limited to, seamless integration with existing LANs fficient support for traditional Internet IP networking and further development of flow nd congestion control algorithms to support existing data services. Research on topics lated to ATM networks is

urrently proceeding and will undoubtedly continue to proceed as the technology atures.

CHAPTER 7

7.1. SYSTEM MODELS

7.1.1. Congestion Control

Telecommunication networks provides the management voice, video and data to control multimedia traffic. ATM (Aynchronous Transfer Mode) is a current technology that will lead to such integration. ATM statistically multiplexes bursty traffic, which needs some traffic and congestion control mechanisms that are still under study both by ITU-T and the ATM Forum.

Congestion is a state of a network node. A network node is congested when the total input arrival rate at this nodeexceeds the service rate of the node 6,7,8. Congestion can occur at the user-network interface (UNI) or at the network-network interface (NNI) depending on the bottlenecks and resource-demand relationships of the network. Thus, congestion can occur monthly, daliy, per session, per multiple round trip times or less than one round trip time. As the occurance rate of congestion increases, the complexity of the scheme that will handle congestion increases.

When congestion does occur, the network recovers by simply dropping cells indiscriminately. An alternative solution could be selectively discarding of low priority cells only. Both voice and video traffic can be coded in such a way that cells essential tothe basic signal are identified (H: High priority cells) separately from those that contribute to a higher quality result (L: Low priority cells). So the network can control congestion by discarding less important information.

A way of implementing a discarding policy is to mark packets in priority classes to indicate how important they are. ATM cells have one cell lose priority bit (CLP) reserved in the header, so that every ATM cell can be labeledeither as a lowpriority or a high priority cell. This bit works together with quality of servic parameter CLR (cell loss ratio). Usually CLR is not defined if CBR traffic needs to be transmitted with no loss. In that case CLP is set to 'high'. For the other traffic classes, when the CLP is set 'low' the loss of the cell during congestion is highly dependent on the buffering ethod at the network node and the usage parameter control (UPC) mechanism. Thus e can say that assigning priorities and controlling loss can also be done with buffering ethod.

1.2 The First System Model

gure 7.1.2.1 shows the state transition diagram of one source that generates cells. This ite transition diagram shows the states of the sources that are used for input to the TM multiplexer. This implies that state transition are synchronized to occur at the ot's boundaries according to a tree-state aperiodic discrete-time Markov chain. All urces start at the 'Off-state'. The probability of the transition from off state, to the On te is (1-β). The first generated cell must be at On-H state which is the start of a ocycle, but it can only stay at that state only one time slot. There is no way back to 'F state, and it must now genearte any number of On-Low cells. With a probability is the time slots spent at On-L state, and the number of generated On-Low cells are not iited but has a probability of γ2. All On-Low cells generated belong to the same sub-cle until until a new On-High cell is generated. The cells can pass from On-Low state On-High state with a probability of d(1-γ2). This is the start of a new sub-cycle. insition from the On-L state to the Off-state occurs with a probability of (1-d)(1-γ2) l this is the end of the last On-subcycle of the On Cycle.

.3 An Equivalent System Model

assume that On and Off periods alternate with each other. An On period consist of ı number of sub-periods and each period is geometrically disributed. We assume that t cell of each sub period is high-priority cell, and remaining cells are low-periority s. At the end of each sub period, an independent Bernoulli trial determines whether begin a new sub-period or make a transition to an Off period. This result in metrically distributed. The Markov model shown in figure 7.1.3.1will generate the ımed cell patterns.

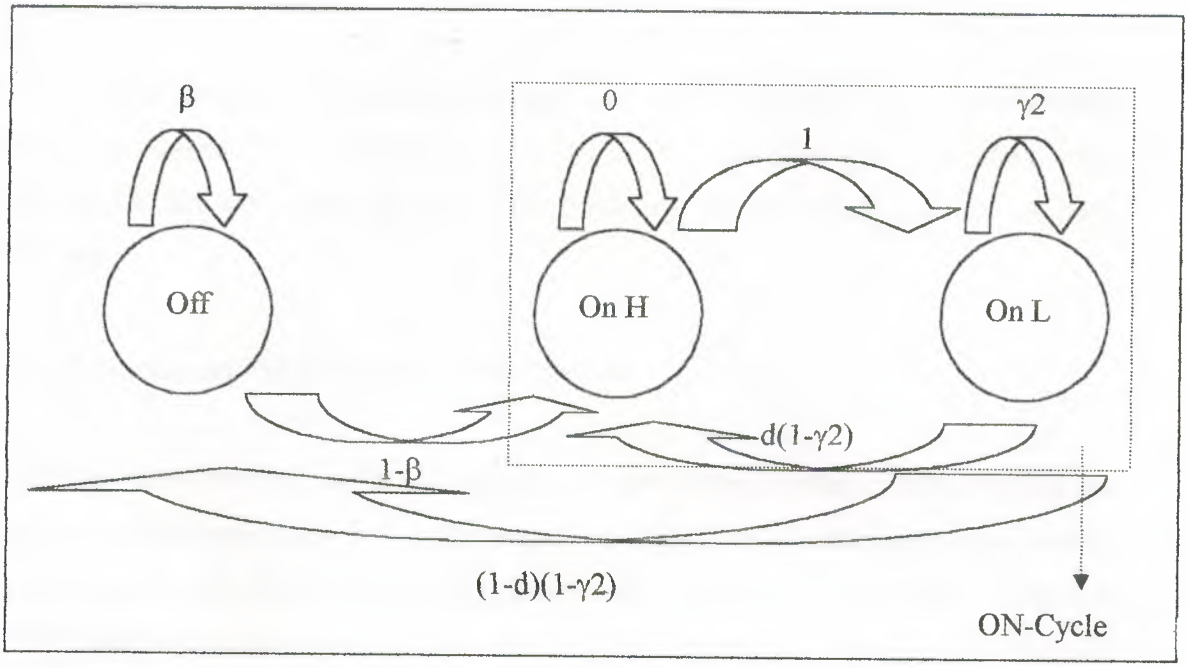


Figure 7.1.2.1 State Transition Diagram of the First System Model.

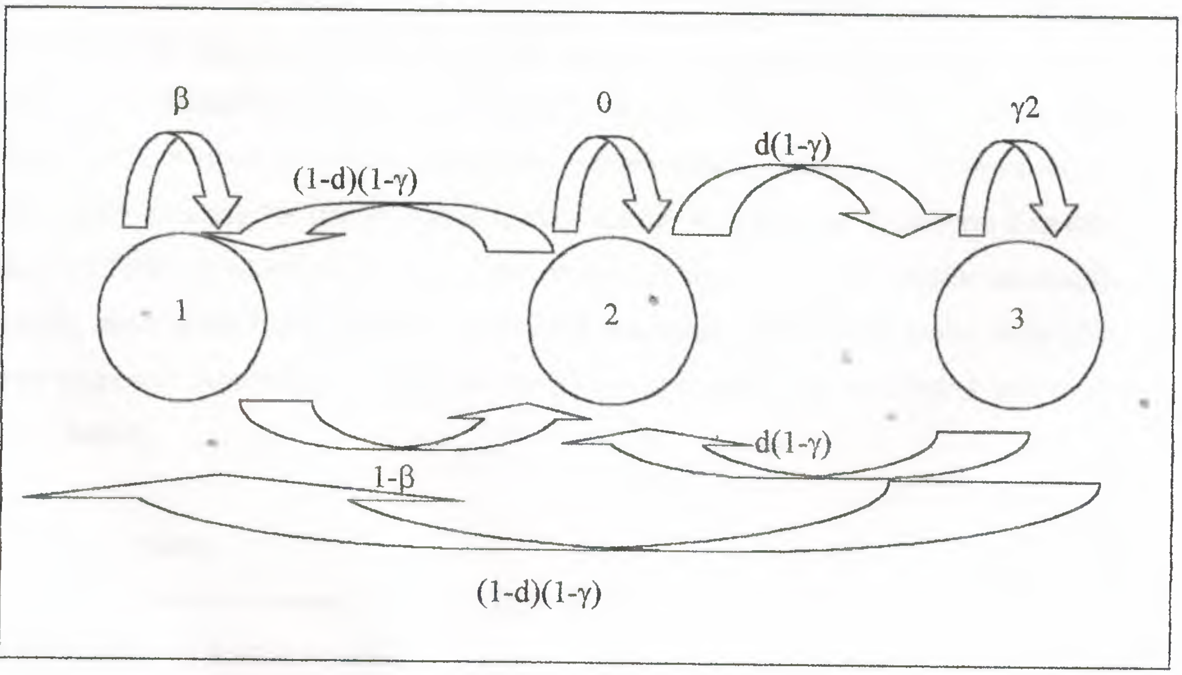


Figure 7.1.3.1. State Transition diagram of the Equivalent Model

.2. MODELLING OF MULTIMEDIA TRAFFIC

ı this section, various approaches that have been used for modifying multimedia traffic ʼill be introduced. As mentioned earlier ATM has the capability to transfer data, voice nd video at the same time. However, the modeling characteristics of these are quite ifferent.

.2.1. Modeling of MPEG Video Traffic Source

tatistical video traffic models are required for generating synthetic video streams for etwork performance studies. A wide range of models, from traditional Markov chains new approaches like fractal models have been proposed in the literature. Several affic models have been proposed to characterize compressed video streams. Maglaris al describe a video source as a first order autoregressive process AR(1) with Gaussian arginal and an exponential autocorrelation function. Skelly et al model video as a larkov-modulated Poisson process. They argue that traffic smoothing on a frame-by-ame basis allows one to model a stream of multiplexed video sources as a 'randomly odulated' Poisson process. It is observed that for video teleconferencing sequences, e number of cells per frame follows the gamma distribution rather than the normal stribution. Moreover the authors suggest that a multi-state Markov chain model, hich can be derived from three parameters, namely mean, variance, and correlation, is fficiently accurate for use in traffic studies. Lazar et al use the Transform-Expand-ample (TES) approach to model a video stream. A number of self-similar stochastic odels, such as fractional Gaussian noise and fractional ARIMA processes, have also een proposed. According to Rose the modeling approaches can be divided into three ain classes:

- Markov chains
- Autoregressive processes (including TES models)
- Self-similar or fractal models

owever, these models are either inadequate to capture the real traffic characteristics, or e analysis are done by multiplexing identical traffic sources, which does not reflect real life applications. Therefore, to achieve adequate and more realist results, analysis should be based on "real" traces representing actual traffic patterns captured from operational multimedia networks. At the beginning of our work we found four video traces that could be used for conducting our video traffic. These traces are *"The Wizard of* Oz", *"The Silence of the Lambs"*, *"Goldfinger"*, and *"Star Wars"*. We decided to use the *"Star Wars"* movie since it represents a realistic full-length sample of entertainment video with a diverse mixture of material ranging from low complexity/motion scene to those with very high action.

7.2.1.1. Overview of Video Source

The MPEG video trace of the movie *"Star Wars"* was obtained from *Bellcore* by anonymous ftp from *ftp.bellcore.com,* directory *pub/vbr.video.trace.* The captured data (in bits per frame) is from two-hour movie with 174,136 video frame, with frame rate of 24 frames per second. The movie has a compression pattern of (12,3) IBBPBBPBBPBB so there are 12 frames in a Group of Pictures (GOP). However, statical analysis showed that the first *I* frame and the *B* frame were missing from the trace. The original video was captured as 408 lines by 508 pels, and then interpolated and filtered to standard CIF frame size, which is 240x352 (Luminance - Y), 120x176 (Chrominance - U & V). The frame is partitioned into blocks of 8x8 pels, on which a Discrete Cosine Transform (DCT) is computed. The DCT coefficients are uniformly quantized into 8 bits and compressed using run-length and Huffman coding. These algorithms comprise essentially the same coding as the JPEG standard. The frame statistics for the *I, P,* and *B* frames are shown in Table 7.2.1.1.

Frame Type	Total Frame	Min (Kb/frame)	Max (Kb/frame)	Mean (Kb/frame)	Std. Dev. (Kb/frame)	Peak/ Mean
I	14511	11.75	185.27	60.38	19.81	3.07
P	43534	2.19	174.29	23.07	14.66	7.56
B	116091	0.48	64.79	7.20	22.96	9.00

Table 7.2.1.1. Frame-level statistics for the *Star Wars* trace

7.2.2. Modeling of Voice Traffic Source

In ATM networks, there are two alternatives for transferring voice traffic. One is to convey it as constant bit rate (CBR) traffic; that is coding at a fixed rate, such as 65 Kbps pulse code modulation (PCM) or 32 Kbps adaptive differentials PCM (ADPCM). PCM and ADPCM are methods used to quantize an analog voice signal and encode the quantization level as digital information. These standards achieve a high voice quality, so called toll quality, but still require relatively high bandwidth. The other method is to use a speech activity detector (SAD) and a digital speech information (DSI) technique. In this case, voice traffic is variable bit rate (VBR) traffic.

When voice signals are coded with a variable bit rate, the process generating cells is as follows: During a talk spurt, a cell is generated at a fixed interval T; and during a silence period, no cells are generated (Figure 7.2.2.1). Consequently, the number of cells transmitted in the network can be reduced by 35% - 40%.

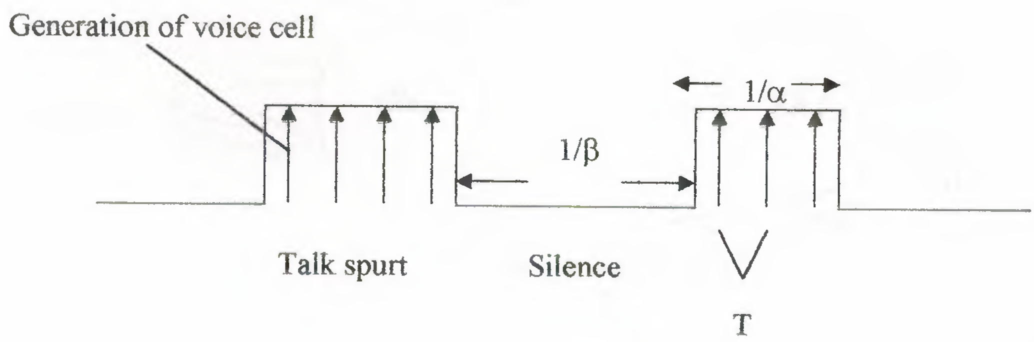


Figure 7.2.2.1. Voice source behavior

In our voice model we used a two ON-OFF model where the holding time in each state is assumed to be exponentially distributed. ON state corresponds to the speech the OFF state corresponds to the silence period. The commonly accepted values for holding time in silent state is 650 ms, and that in the speech state is 352 ms. These values depend on the sensitivity of the silence detectors.

The two-state model has some known limitations. In particular, the simple two-states model does not model a two-way conversion since two-way conversions cannot be modeled by merely superimposing multiple single source speech generators. Events like interruption and double-talking are possible in two-way conversation. Such events will affect any model that tries to approximate speech pattern. Some researchers have proposed a four-state Markov Model to describe the behavior of such system. The four-states represent who is doing talking; no one, one person, the other person, or both. This is only a crude approximation since such a Markov Model has exponential distribution for each state, which may not be realistic. A better model is to add two more states are split into two states making a total of six states. Mutual silence and double-talk states are split into two states, with the identity of the last lone speaker differentiating them. As we stated before, our two-state Markovian model has mean speech and silence times 352 ms and 650 ms respectively. Figure 7.2.2.2.

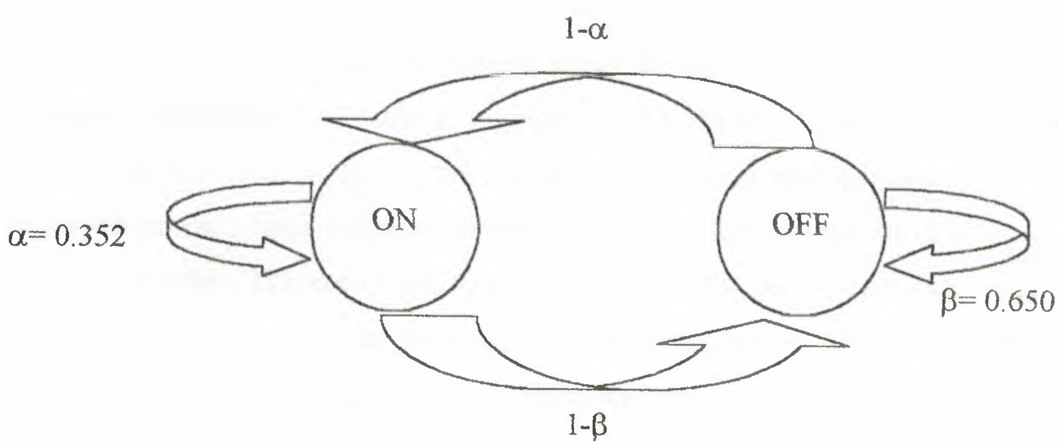


Figure 7.2.2.2. Two-state ON-OFF voice model

CONCLUSION

In this project, I worked congestion and admission control in ATM Networks. In the ATM Networks, the ATM switch is transporting ATM Cell traffic over ATM with the guaranteed level of QoS. Increasing the output buffer size of capacity of 4 Cells does not make any cell lose at the output buffer. For reducing the CLR also we should be sure about the cell arrival rate of the input rates. If there is more cell arriving to the ATM switch than it can transport within a period there will be bottlenck. When this happens the average cell delay is increasing too much, sometimes exceedings the tolerable limits for the ATM switches. The other effect of this problem is the overflowing of the cells inside the ATM switch and causing a high Cell Loss Ratio. This overflow may happen in either the output buffers or happens in the memory. The limited capacities of these buffers and memory are due to reducing the cost of the switch and also reducing the complexity of the switch architecture.

ATM switches based upon this design would be appealing to cable television and telephone companies deploying high-bandwidth WANs. Such networks would conceivable be used to carry simultaneus voice, video and computer data traffic. Deployement on a large scale would require a substantial investment in large numbers of ATM switches. The use of mixed fiber-coaxial WANs would eliminate much of the associated costs of ATM deployment, while still providing an effective solution to the high-bandwidth and high-performance demands of such users.

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