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LAYERS IN ATM NETWORKS

Dissertation Submitted to
JAWAHARLAL NEHRU UNIVERSITY
in partial fulfilment of requirements
for the award of the degree of
Master of Technology
in
Computer Science

by
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January 1999

Dedicated to

My Beloved Parents

JAWAHARLAL NEHRU UNIVERSITY
NEW DELHI - 110 067

CERTIFICATE

This is to certify that the dissertation entitled

"LAYERS IN THE ATM NETWORKS"

Which is being submitted by **Mr. Anand Kumar Agarwal** to the **School of Computer & Systems Sciences, Jawaharlal Nehru University, New Delhi** for the award of degree in **Master of Technology** in Computer Science, is a record of bonafide work under the supervision and guidance of **Dr. R. C. Phoha.**

This work is original and has not been submitted in part or full to any University or Institution for the award of any degree.

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ACKNOWLEDGEMENTS

I wish to convey my heartfelt gratitude and sincere acknowledgements to my guide ***Dr. R. C. Pholia,*** School of Computers & Systems Sciences for his whole hearted, tireless and relentless effort in helping me for the successful completion of this project.

I would like to record my sincere thanks to my dean, ***Prof. P. C. Saxena,*** School of Computers & Systems Sciences for providing the necessary facilities in the centre for the successful completion of this project.

I take this opportunity to thank all of my faculty members, my friend Mr. Lalit Mohan Pant, all other my friends for their help and special thanks to my wife **Mrs. Sarika Agarwal** for her help and co-opration.

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ABSTRACT

The Asynchronous Transfer Mode (ATM) has been strongly promoted, as the transport structure for future broadband telecommunication networks, within standards bodies CCITT. ATM is a transport technique based on fast packet switching, where the information is packed into fixed size cells. These cells are identified and switched throughout the networks. This contains the virtual connection identification and whatever other addressing is required, as well as perhaps priority and other control data. In the ATM networks three layers are defined.

ATM networks carry fixed-size cells within the network irrespective of the applications being supported. At the network edge or at the end equipment, an ATM adaptation layer (AAL) maps the services required by the application. Many trunking applictions that have voice compression and silence suppression require transmission of small delay-sensitive packets. Existing AALs are very inefficient for this purpose. In this dissertation, we discuss the AAL-2, which allows very high efficiency for carrying small packets. We describe the basic principles and compare several alternatives with respect to transmission error performance, bandwidth efficiency and delay/jitter performance. The results show that the AAL-2 adds significant value to packet telephony applications over ATM networks.

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CHAPTER 1

INTRODUCTION

1.1. BASIC PRINCIPLES OF ATM

ATM offers a flexible transfer capability common to all services because of its independence of the bit-rate and data structure of the services carried. For these reasons, additional functionalities, are needed to accommodate the various services and these are added at the edge of the ATM network.

ATM is a particular packet-oriented transfer mode that uses asynchronous time division multiplexing techniques with the multiplexed information flow being organized into blocks of fixed size, called cells. A cell (as following figure) consists of an information field carrying user information (cell payload) and a header containing network information, for example routing information. Because cells from more than one connection are multiplexed together, there has to be a means to identify cells belonging to the same connection and this is done by information in the header.

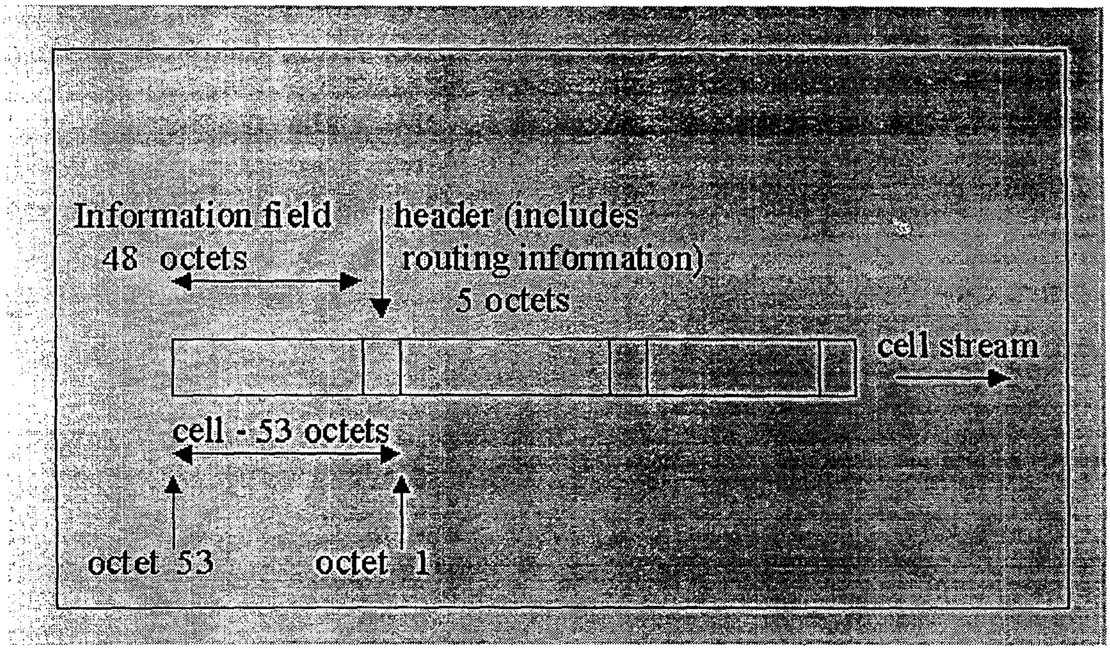

Figure 1-1. Basic ATM cell format

The information field is transported transparently by the ATM network and no processing (for example error control, used in conventional packet networks) is performed on this field by the ATM network. The cell sequence is preserved by the ATM networks, cells being received in the same order as they are sent (cell sequence integrity).

Because ATM is a connection-oriented technique, a path has to be established between the users before information can be exchanged. This is done by the connection set-up procedure at the start and by a clear-down procedure at the end. The set-up procedure uses a signaling protocol for on-demand connection and other means, for instance a network-management procedure, for semi -permanent or for permanent connections. A broadband call can be a multi-media call having a number of components. As each component generally requires a separate connection, it is usual to discuss what happens at a 'connection' level rather than 'call' level.

Each connection has a transfer capacity (a bandwidth) assigned to it according to the user's request, subject to there being sufficient capacity available. This is usually done during the connection set-up procedure using a process called connection admission control (CAC); this process determines the parameters that the connection will be allowed to have depending on the user's needs. There is another process, usage parameter control(UPC) that monitors the connection and takes action if the connection attempts to exceed the limits that have been allocated to it.

1.2. PROTOCOL REFERENCE MODEL

In a similar way to the familiar OSI 7-layer model, the B-ISDN also has a protocol reference model (as following figure), which consists of a user plane, a control plan and a management plane. The concepts of service access points (SAPs), service data units(SDUs) and protocol data units(PDUs) that are found in the OSI layered approach also apply to this protocol reference model.

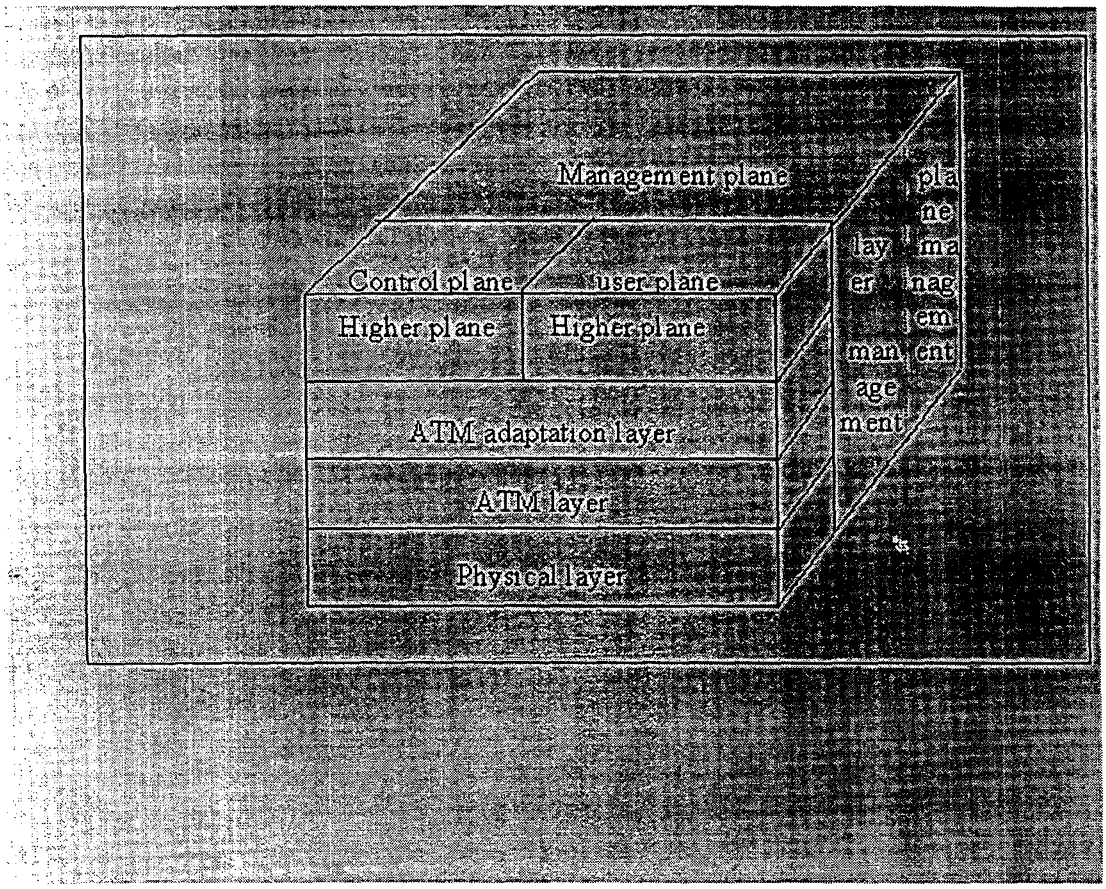

Figure 1-2. B-ISDN protocol reference model and the functions of the layer

The user plane (for user information transfer) and the control plane (call control and connection control functions) are structured in layers. Above the physical layer, the ATM layer provides cell transfer for all services and the ATM adaptation layer (always referred to as the AAL) provides service-dependent functions to the layer above the AAL. The management plane provides network supervision functions. As would be expected with such a layered protocol, the characteristics of the ATM layer are independent of the physical medium.

CHAPTER 2

LAYERS IN ATM NETWORKS

2.1. PHYSICAL LAYER

2.1.1. Principles

The physical layer consists of two sublayers: the physical medium sublayer(PM) and the transmission convergence sublayer(TC).

The physical medium sublayer includes only physical-medium-dependent functions and provides bit transmission capability, including bit-transfer and bit alignment. It includes line coding and electrical-optical transformation.

The transmission convergence sublayer performs all those functions necessary to transform a flow of cells into a flow of data units (i.e. bits) which can be transmitted and received over a physical medium.

Going from the physical layer to the ATM layer, the flow of data (strictly, in OSI terms, the flow of service data units) crossing this boundary is a flow of valid cells. Valid cells are those whose headers have no errors, error checking on the header having been performed in the transmission convergence sublayer.

Going in the opposite direction, from the ATM layer to the physical layer, the ATM cell flow is merged with the appropriate information for cell delineation and it also carries operation, administration and maintenance (OAM) information relating to this cell flow.

2.1.2. Transfer Capacity

CCITT Recommendation I.432 defines two bit-rates for the physical layer at the T_B reference point: 155.520 Mbps in both direction and 622.080 Mbps in at least one direction. The interfaces may be electrical or optical and it may use a cell-based structure or SDH framing.

The bit-rates above are the gross bit-rates of the physical layer and the overhead due to physical-layer frame-structure octets or physical-layer cells must be subtracted. The values available for ATM cells are given below and these correspond to the payload of SDH. The maximum user bit-rate is 48/53 of that available for cells and this too is shown below. However, the bit-rate available for ATM cells must also be used for signaling cells and OAM information cells for the ATM and higher layers. The bit-rate available for user information is consequently less than the values in the table.

Physical layer parameter values

	lower bit-rate (Mbps)	higher bit-rate (Mbps)
gross physical-layer bit-rate	155.520	622.080
max bit-rate available for ATM cells	149.760	599.040
max bit-rate available for cell payload	135.631	542.526

2.1.3. Physical Layer for Cell-based Interface

The interface structure consists of a continuous stream of cells. The maximum spacing between successive physical-layer cells is 26 ATM-layer cells. Physical-layer cells are also inserted when no ATM-layer cells are available. The physical layer cells which are inserted can be either 'idle cells' or physical-layer OAM cells depending on the OAM requirements.

Physical-layer OAM cells are used for conveyance of the physical-layer OAM information. How often these cells are inserted should be determined by OAM requirements. However, there must be not more than one physical-layer OAM cell every 27 cells and not less than one physical-layer OAM cell every 513 cells on an operating link.

The physical-layer OAM cells have a unique header so that they can be properly identified by the physical layer.

2.1.4. Header Error Control

The header error control (HEC) covers the entire cell header. The code used for this function is capable of either single-bit error correction or multiple-bit error-detection.

The transmitter calculates the HEC value (an 8-bit sequence) across the entire ATM cell header and inserts the result in the HEC field. The value is the remainder of the division (modulo 2) by the generator polynomial (x^8+x^2+x+1) of the product x^8 multiplied by the content of the header excluding the HEC field.

To significantly improve the cell delineation performance in case of bit slips, the CCITT recommends that the check bits calculated by the use of the check polynomial are added (modulo- 2) to an 8-bit pattern(01010101), the left bit being the most significant) before being inserted in the HEC field. The receiver must subtract (equal to add modulo 2) the same pattern from the 8 HEC bits before calculating the syndrome of the header. This operation does not affect the error detection/correction capabilities of the HEC.

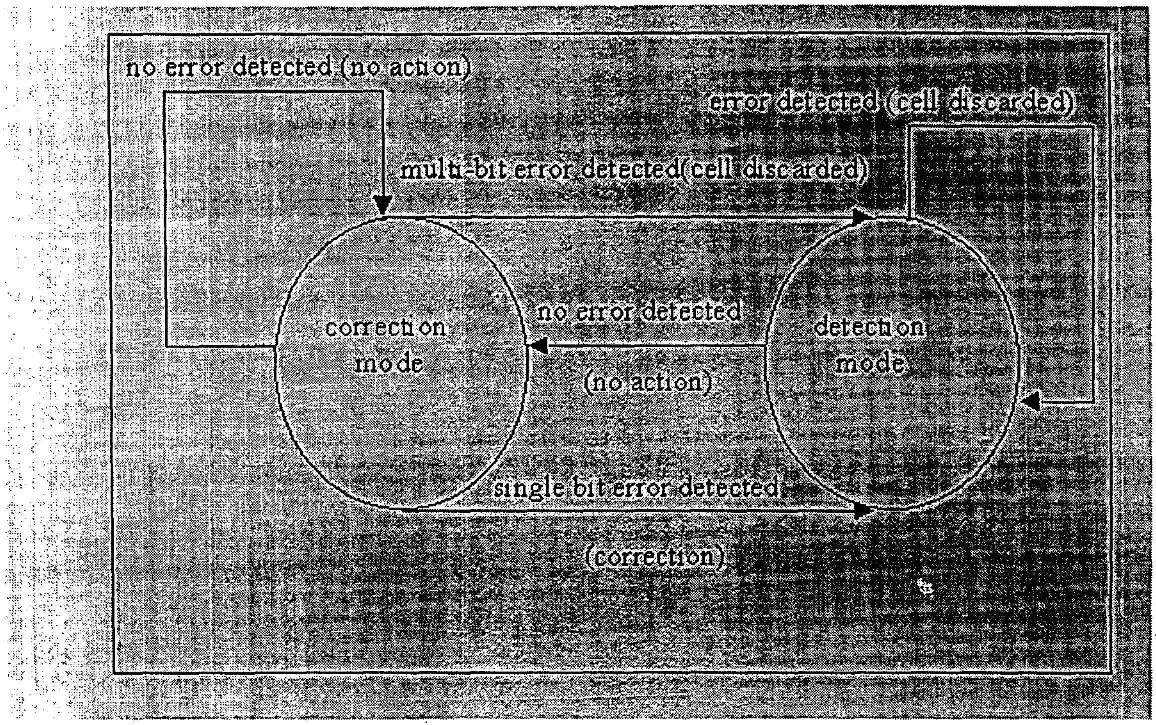

Figure 2-1. HEC: receiver modes of operation

The receiver has two modes of operation. The default mode provide for single-bit error correction. Each cell header is examined and, if an error is detected, one of two actions takes place depending on the state of the receiver. In 'correction mode'. In 'detection mode', all cells with detected header errors are discarded. When a header is examined and found not to be in error, the receiver switches to 'correction mode'.

The error-protection function of the HEC providers both recovery from single-bit header-errors, and a low probability of the delivery of cells with errored headers under bursty error conditions. The error characteristics of fibre-based transmission systems appear to be a mix of single-bit errors and relatively large burst errors.

2.1.5. Cell Delineation and Scrambling

Cell delineation is the process which allows identification of the cell boundaries and is performed using a procedure based on the header error control (HEC) field.

Scrambling is used to improve the security and robustness of the HEC cell delineation mechanism. In addition it helps to randomize the data in the information field for possible improvement of the transmission performance. Any scrambler specification must not alter the ATM header structure, header error control and cell-delineation algorithm.

The recommended cell delineation method is performed by using the correlation between the header bits to be protected (32 bits) and the relevant control bits (8 bits) introduced in the header by the HEC (header error control) using a shortened cyclic code with generating polynomial x^8+x^2+x+1.

Following figure shows the state diagram of the HEC cell delineation method.

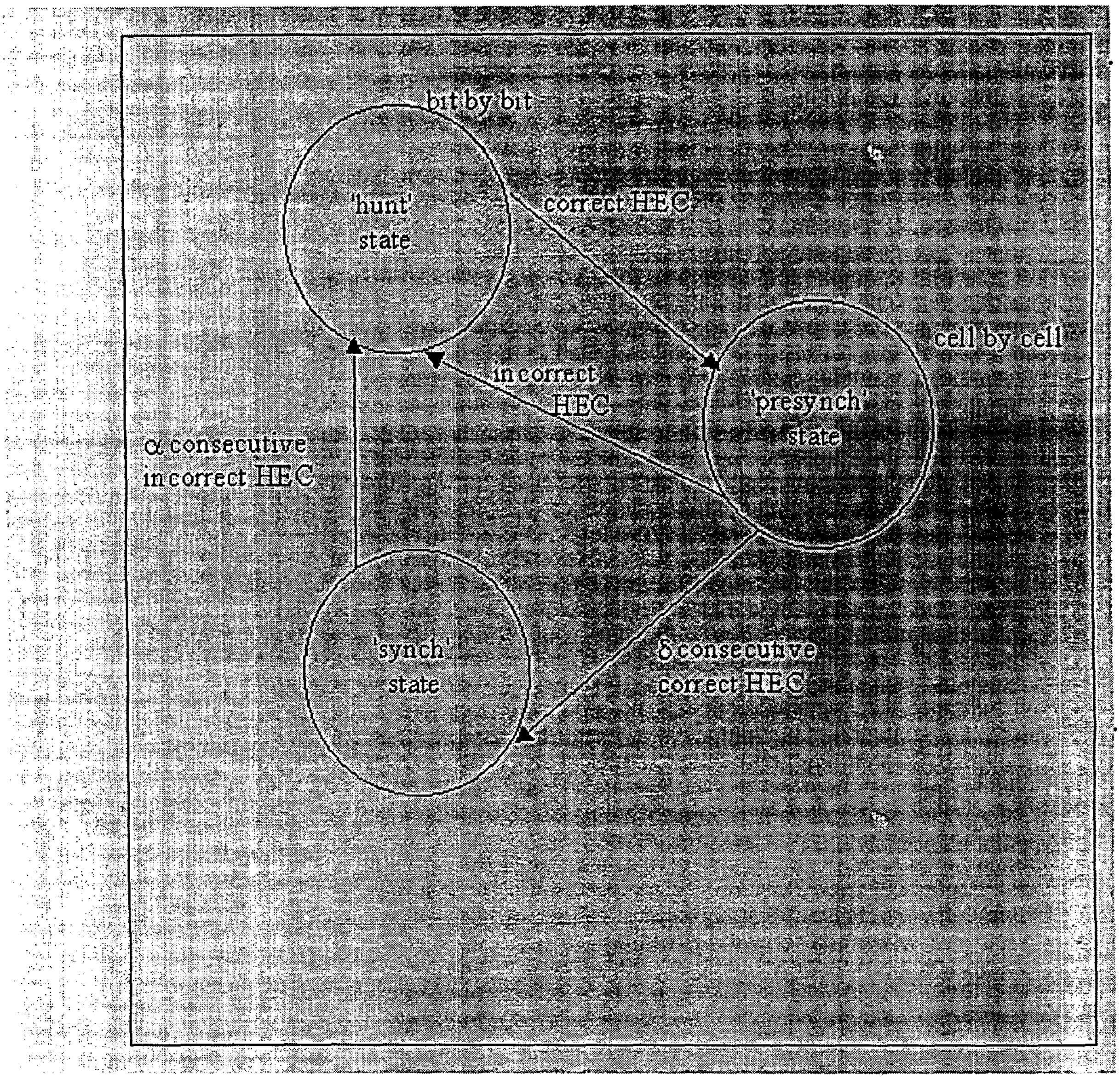

Figure 2-2. Cell delineation state diagram

(i) In the hunt state, the delineation process is performed by checking bit by bit for the correct HEC (i.e. syndrome equals zero) for the assumed header field. For the cell-based physical layer, prior to scrambler synchronization, only the last six bits of the HEC are used for cell delineation checking. Once such an agreement is found, it is assumed that one header has been found, and the method enters the presynch state. When octet boundaries are available within the receiving physical layer prior to cell delineation, the cell delineation process may be performed octet by octet.

(ii) In the presynch state, the delineation process is performed by checking cell by cell for the correct HEC. The process repeats until the correct HEC has been confirmed δ times consecutively. If an incorrect HEC is found, the process returns to the hunt state.

(iii) In the synch state the cell delineation will be assumed to be lost if an incorrect HEC is obtained α times consecutively.

The parameters α and δ have to be chosen to make the cell delineation process as robust and secure as possible and able to satisfy the required performance. Robustness against false misalignments due to bit errors depends on α; robustness against false delineation in the re-synchronization process depends on the value of δ.

2.1.6. Scrambler Operation

For the cell based UNI, the distributed sample scrambler is recommended. This is an example of a class of scrambler in which randomization of the transmitted data stream is achieved by modulo addition of a pseudo-random sequence. Descrambling at the receiver is achieved by modulo addition of an identical locally generated pseudo-random sequence having phase synchronization with the first in respect of the transmitted cells. The scrambler does not affect the performance of the 8-bit HEC mechanism during steady-state operation.

2.2. ATM LAYER

It is worth considering the ATM layer first as this is the layer that is concerned with transporting information across the network which consist the transmission path.

2.2.1. Transmission Path

The transmission path extends between network elements that assemble and disassemble the payload of a transmission system (the payload will be used to carry user information; together with the necessary transmission overhead it forms the complete signal).

ATM uses virtual connection for information transport and these connections are divided into two levels: the virtual path level and the virtual channel level. This subdivision of the transport function is one of the powerful features of ATM.

Virtual channel (VC)

' A concept used to describe unidirectional transport of ATM cells associated by a common unique identifier value. ' This identifier is called the **virtual channel identifier** (VCI).

Virtual path (VP)

' A concept used to describe unidirectional transport of cells belonging to virtual channels that are associated by a common identifier value. ' This identifier is called the **virtual path identifier** (VPI).

A transmission path may comprise several virtual paths and each virtual path may carry several virtual channels. The virtual path concept allows the grouping of several virtual channels.

2.2.2. ATM Objects

There are four ATM objects.

(i) **Virtual channel link** ' A means of unidirectional transport of ATM cells between a point where a VCI value is assigned and the point where that value is translated or removed. '

(ii) **Virtual path link** ' A means of unidirectional transport of ATM cells between a point where a VPI value is assigned and the point where that value is translated or removed. '

(iii) **Virtual channel connection** A connection of VC links is called a virtual channel connection (VCC).

(iv) **Virtual path connection** A connection of VP links is called a virtual path connection (VCC).

The functions of the ATM layer are considered in more detail below.

2.2.3. Cell Multiplexing and Demultiplexing

In the transmit direction, the cell multiplexing function combines cells from individual virtual paths (VPs) and virtual channels (VCs) into one cell flow. In the receive direction, the cell demultiplexing function directs individual cells to the appropriate VP or VC.

2.2.4. Cell VPI and VCI Translation

The VPIs and VCIs are labels that identify a particular VP and VC on that link. The switching node uses the these values to identify a particular connection and will then, using the routing information established at connection set-up, route the cells to the appropriate output port. The switching element changes the value of the VPI and VCI fields to new values that are used on the output link.

2.2.5. Cell Header Generation/Extraction

These functions apply at points where the ATM layer is terminated. In the transmit direction, the cell-header generation function receives a cell-information field from a higher layer and generates an appropriate ATM cell header, except for the HEC (header error control) sequence which is calculated and inserted by the physical layer. In the receive direction, the cell header extraction function removes the ATM cell header and passes the cell information field to the ATM adaptation layer.

2.2.6. Generic Flow Control

The generic flow control (GFC) function is used only at the user-network interface (UNI). It may assist the customer network in the control of cell flows towards the network but it does not perform flow control of traffic from the network. The GFC protocol information is not carried through the network.

2.2.7. Cell Structure

The ATM cell consists of a 5-octet header and a 48-octet information field immediately following the header; it is fully specified in CCITT Recommendation I.361 (B-ISDN ATM-layer specification). Two cell formats have been specified, one for the user-network interface (UNI) and the other for the network-network interface (NNI). The UNI format is used between the user installation and the first ATM exchange as well as within the user's own network. The NNI format is used between the ATM exchanges in the trunk network.

The header is divided into the following fields.

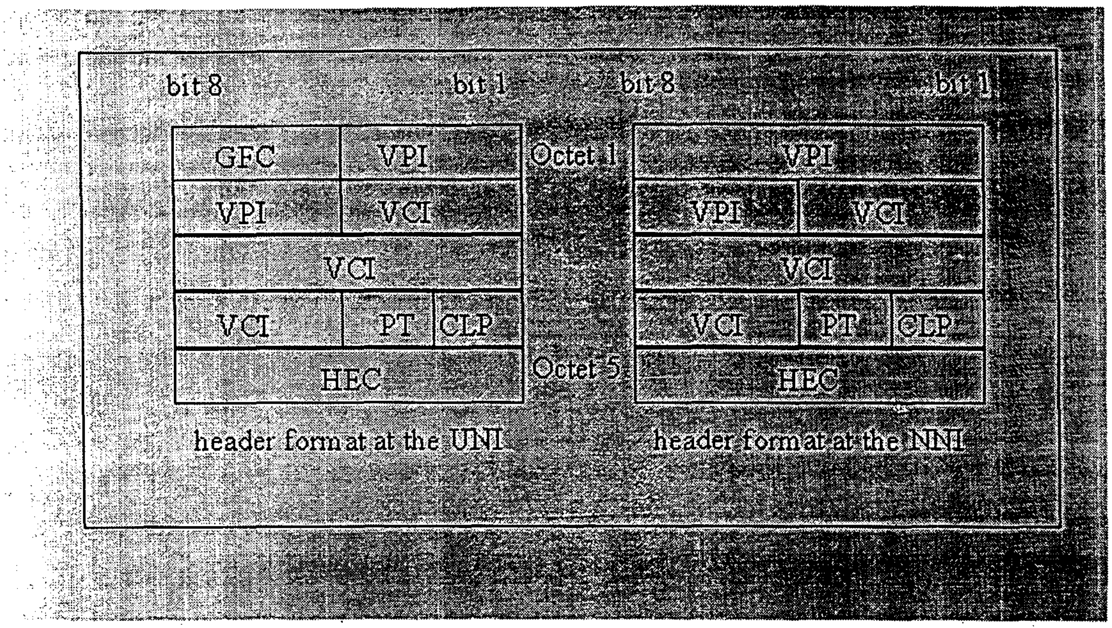

Figure 2-3. Format of the ATM header

Generic flow control (GFC) field

The GFC field contains 4 bits at the UNI and, as explained above, it can provide flow control information towards the network.

Routing field (VPI/VCI)

It contains 24 bits (8 for VPI and 16 for VCI) are available for routing at the UNI and 28 bits (12 for VPI and 16 for VCI) at the NNI.

Payload type (PT) field

In the cell 3 bits are available for payload type identification. The payload type field is used to provide an indication of whether the cell payload contains user information or network information; this is specified in CCITT Recommendation I.361. When the PT field indicates that the cell contains network information, the network processes the information field of the cell.

Cell loss priority (CLP) field

This bit may be set by the user or service provider to indicate lower-priority cells. Cells with the CLP bit set are at risk of being discarded depending on the conditions in the network.

Header error control field (HEC) field

The HEC field consists of 8 bits. This field is processed by the physical layer to detect errors in the header, the error control covering the entire cell header. The code used for this function is capable of either single-bit error-correction, or multiple-bit error-detection.

2.2.8. Switching Functions

Switching defines how a transmission path is routed through a network from the source terminal to the destination terminal, and how channels of intermediate links are associated with each other to form a connection between end points. As explained earlier, the ATM transport network is structured as two layers, the ATM layer and the physical layer. The transport functions of the ATM layer are independent of the physical-layer implementation and are subdivided into two levels: the VC (virtual channel) level and the VP (virtual path) level. Switching in an ATM network is very different from that in the N-ISDN.

Although a connection is established at call set-up, it does not consist of a fixed bandwidth path that is exclusively available for the particular connection. Instead, cells from many connections are multiplexed onto a link, then queued at the switching node at the switching node before each cell is individually switched to the appropriate destination link.

The bulk of the traffic in the network will be generated by the business customer, with a significant proportion being based on semi-permanent connections or leased lines. Managing this share of the traffic in an optimal manner results in significant cost savings and high revenue: such traffic management is possible with the use of flexible and re-configurable ATM cross-connects in the network.

In general, a connection in an ATM network can be established on-demand of the connection can be semi permanent or permanent. With on-demand switching, the establishment and release of a connection is done by using signaling procedure in the control-plane of the protocol reference model. Semi-permanent connection between agreed points may be provided for an indefinite period of time after subscription, for a fixed period or for agreed periods during a day, week or other interval. Permanent connection are available to the customer to the customer at any time during the period of subscription between the fixed points requested by the customer when the permanent connection was ordered. Both semi-permanent and permanent connections are set up by the management plane.

In a more formal way, it can be said that in control plane communication a user manages (establishes, releases and maintains) a VPC/VCC by sending control-plane messages through a signaling VCC that is terminated at a VC-switch. In management-plane communication, the connections are established, released and maintained by the VP or VC cross-connect using the network-management function.

There are three types of switch defined in an ATM network and, as explained above, they are directed by control plane functions; the equivalent cross-connects are directed by management-plane functions. A VP switch is a network element that connects VP links and it translates VPI values between input and output; a VC switch is a network element that connects VC links ,translating VCI values, and terminates VPCs; a VP -VC switch is a network element that acts both as a VP switch and as a VC switch.

2.2.9. Routing

The main function of the ATM layer is routing packets from the source machine to the destination machine. In most subnet, packets will require multiple hops to make the journey. The algorithm that choose the routes and the data structures that they use are a major area of network layer design.

The routing algorithm is that part of the network layer software responsible for deciding which output line an incoming packet should be transmitted on. If the subnet uses datagrams internally, this decision must be made a new for every arriving data packet since the best route may have changed since last time. If the subnet uses virtual circuits internally, routing decisions are made only when a new virtual circuit being set up. Thereafter, data packets just follow the previously established route. The latter case is sometimes called session routing, because a route remains in force for an entire user session (e.g. a login session at a terminal or a file transfer).

Route Modification Methods

When designing a network, it is important to implement at least one route modification method to cope with variation in traffic load or with system failure. The three techniques applicable to an ATM network are described below.

(i.) Alternative Routing

This method provides a call with a choice of two or more routes. It is particularly economical for routes that are expensive to provide: these need to be as fully occupied as possible. In peak traffic conditions, the alternative-routing algorithm switches calls over second-choice overflow routes if the first-choice high-usage route is full. In an ATM network, these overflow routes may not involve further call switching, because they could be set up as VPCs which just involve a greater number of hops. The ability to employ a number of alternative paths greatly enhances network availability and security, and does not involve reserving transmission bandwidth.

(ii.) Adaptive Routing

This can be incorporated as an extension to the alternative-routing method, where the second choice paths are not pre-determined but are established according to the current traffic conditions. Such adaptation to the network state obviously requires more control processing, and may incur greater delays in call establishment. This is traded against the gain in efficient use of transmission resources (at normal network loads).

(iii.) Dynamic Routing

Dynamic routing is complementary to the above two, being similar to the adaptive method. However the time scale over which traffic conditions are assessed is different (hours rather than minutes/seconds). In fact, load estimates rather than traffic measurements may be used. The method can exploit, for example, the non-coincidence of busy hours across a large network. If the VP concept is used to the full, dynamic routing can effectively mean the reconfiguring of the logical network structure.

2.2.10. Traffic Shaping

Traffic shaping alters the traffic characteristics (i.e. the inter-arrival times of the cells) of a VCC or a VPC to achieve a desired modification of the traffic characteristics. In its simplest form, this is spacing to smooth out the peaks in cell-rate at the expense of adding in more delay. Following figure illustrates the principle. Traffic shaping must maintain the cell-sequence integrity of a connection.

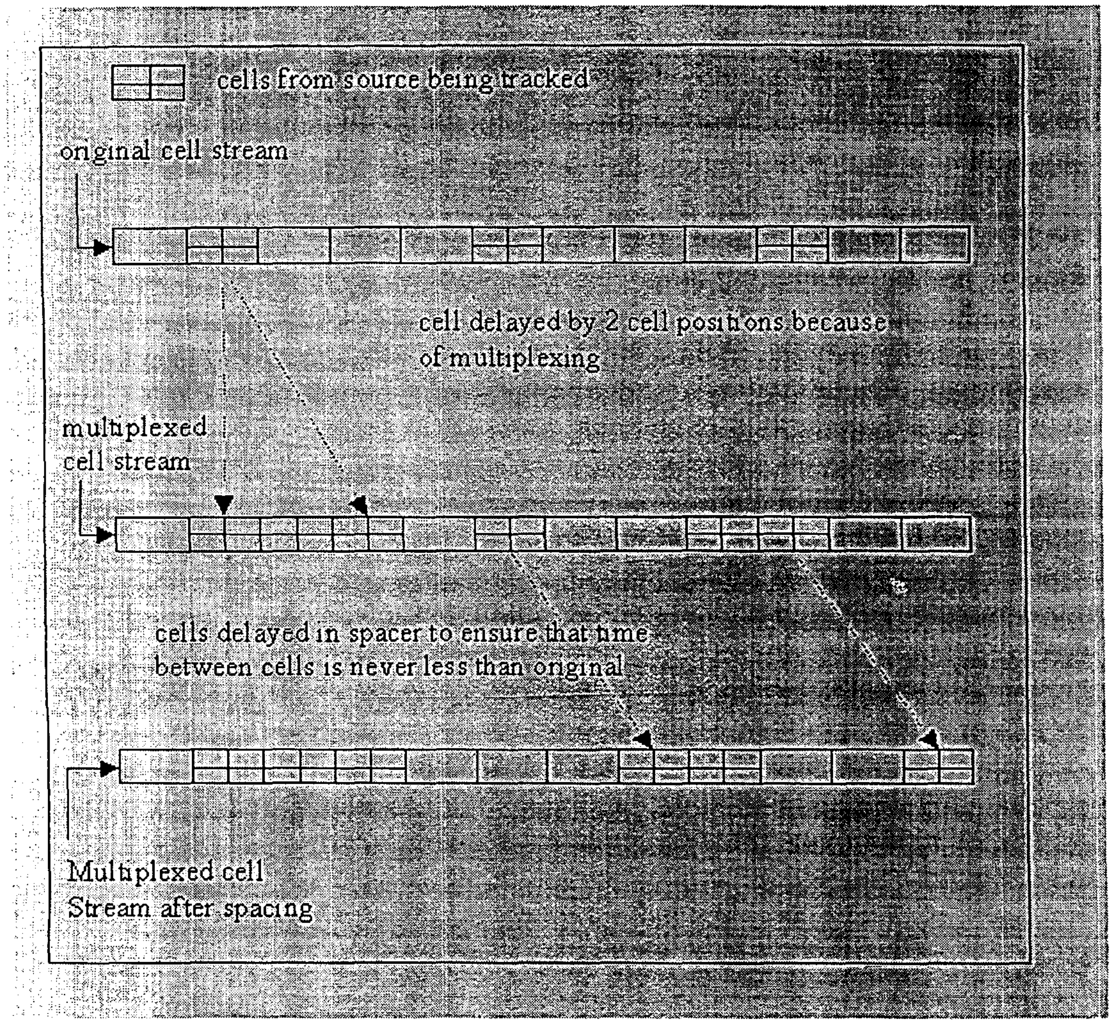

Figure 2-4. Principle of ' spacing '

Several mechanisms, focusing on different aspects of the traffic stream, are candidates for traffic shaping.

The cell spacer compensates for cell delay variation and its consequent impact on peak cell-rate by emitting cells at a declared maximum cell-rate (the inverse of the minimum inter-arrival time). This avoids the cell bunching caused by variation in cell delay, and leads to a considerable performance improvement. (Cell bunching reduces the admissible traffic load within the network.)

Another mechanism aims at an optimal scheduling of cells of different connections originating within the same access network. The phase relationship between different periodic cell steams with identical period lengths is adjusted by means of a suitable shaping function.

A traffic-shaping mechanism that adapts the peak cell-rate at which the next burst will be sent into the network has been used to enforce an effective cell-rate used by the connection admission control. This mechanism is directly related to the linear CAC (Connection Admission Control) scheme.

2.2.11. Service Categories

The service category depends upon the bit rate from source to destination. There are following service category.

(i.) **Constant bit rate (CBR)** Bits are put on one end and they come off the other end. No error checking, flow control or other processing is done.

(ii.) **Variable bit rate (VBR)** VBR is divided into two subclasses, for real time and non-real time is intended for service that have variable bit rates combined with stringent real-time requirements, such as interactive compressed video (i.e. videoconferencing). The other VBR subclass is for traffic where timely delivery is important but a certain amount of jitter can be tolerated by the application.

(iii.) **Available bit rate (ABR)** ABR service category is designed for bursty traffic whose bandwidth range is known roughly. A typical example might be used for in a company that currently connects its offices by a collection of leased lines.

(iv.) **Unspecified bit rate (UBR)** UBR makes no promises and gives no feedback about congestion. This category is well suited to sending IP packets, since IP also makes no promises about delivery. All UBR cells are accepted, and if there is capacity left over, they will also be delivered.

2.2.12. Quality of Service

Quality of service is an important issue for ATM networks, in part because they are used for real-time traffic, such as audio and video. When a virtual circuit is established, both the transport layer (typically a process in the host machine; the "customer") and the ATM network layer (i.e. a network operator, the "carrier") must agree on a contract defining the service.

2.2.13. Congestion Control

When too many packets are present in the subnet, performance degrades. This situation is called congestion. When the number of packets dumped into the subnet by the hosts is within its carrying, capacity, they are all delivered, and the number delivered is proportional to the number sent. However, as traffic increases too far, the routers are no longer able to cope, and they begin losing packets. This tends to make matters worse. At very high traffic, performance collapses completely, and almost no packets are delivered.

2.3. ATM ADAPTATION LAYER

2.3.1. AAL Principles

The ATM adaptation layer (AAL) performs the necessary mapping between the ATM layer and the next higher layer. This is done in the terminal equipment or terminal adapter, i.e. at the edge of the ATM network.

The ATM network, the part of the network which processes the functions of the ATM layer, is independent of the telecommunications services it carries. This means that the user payload is carried transparently by the ATM network and the ATM network does not process the user payload and does not know the structure of the data unit. This is known as semantic independence. There is also time independence as there is no timing relationship between the clock of the network, the network having to cope with any application bit-rate.

The consequence of this independence is that all the functions specific to the services are provided at the boundary of the ATM network and are performed by the AAL. The functions within the AAL have the task of providing the data flow sent by the user to the upper layers at the receiving end, taking into account any effects introduced by the ATM layer. Within the ATM layer, the data flow can be corrupted by errors in transmission or it can suffer cell delay variation as a result of variable delay in buffers or through congestion in the network. Loss of cells or misdelivery of cells are consequences of these effects and these consequences will have an impact on the application, whether it is data transfer, video or voice communication. The AAL protocols must cope with these effects. It could be thought that for each telecommunications service there should be a separate AAL developed. However, taking into account the common factors within possible telecommunications services it is possible to devise a smart set of AAL protocols that should be sufficient for what is currently envisaged.

The functions performed in the AAL depend upon the higher-layer requirements and, as the AAL supports multiple protocols to fit the needs of the different AAL service users, it is therefore service dependent. It is obviously sensible to minimize the number of different AAL protocols required and, to do this, a telecommunication service classification is defined based on the following parameters:

(i) Timing relationship between source and destination
(ii) Bit-rate
(iii) Connection mode.

Other parameters, such as assurance of the communication, are treated as quality-of-service parameters and do not lead to different classes.

Using these parameters, four classes of service have been defined and these are shown as following.

(i) **Class A** Real time, Constant bit-rate and connection-oriented.
(ii) **Class B** Real-time, Variable bit-rate and connection-oriented.
(iii) **Class C** Non real-time, Variable bit-rate and connection-oriented.
(iv) **Class D** Non real-time, Variable bit-rate and connection-less.

Examples of telecommunication services in each of the classes are:

(i) **Class A** circuit emulation, constant bit-rate video
(ii) **Class B** variable bit-rate video and audio

(iii) **Class C** connection-oriented data transfer

(iv) **Class D** connection-less data transfer.

These classes of service are a fairly general concept, but the classes can be mapped onto the different specific AAL types as shown below in table. In addition, there is an empty AAL for those users for whom the performance of the ATM layer may be sufficient for their requirements.

Mapping between classes of service and AAL types

Class	
A	AAL 1
B	AAL 2
C & D	AAL 3/4(the original AAL 3 and AAL 4 combined)
C & D	AAL 5 is alternative to AAL 3/4 for class D services; it is simpler and therefore has less overhead.

The AAL is organized in two sublayers as following figures: the convergence sublayer(CS) and the segmentation and reassembly sublayer(SAR).

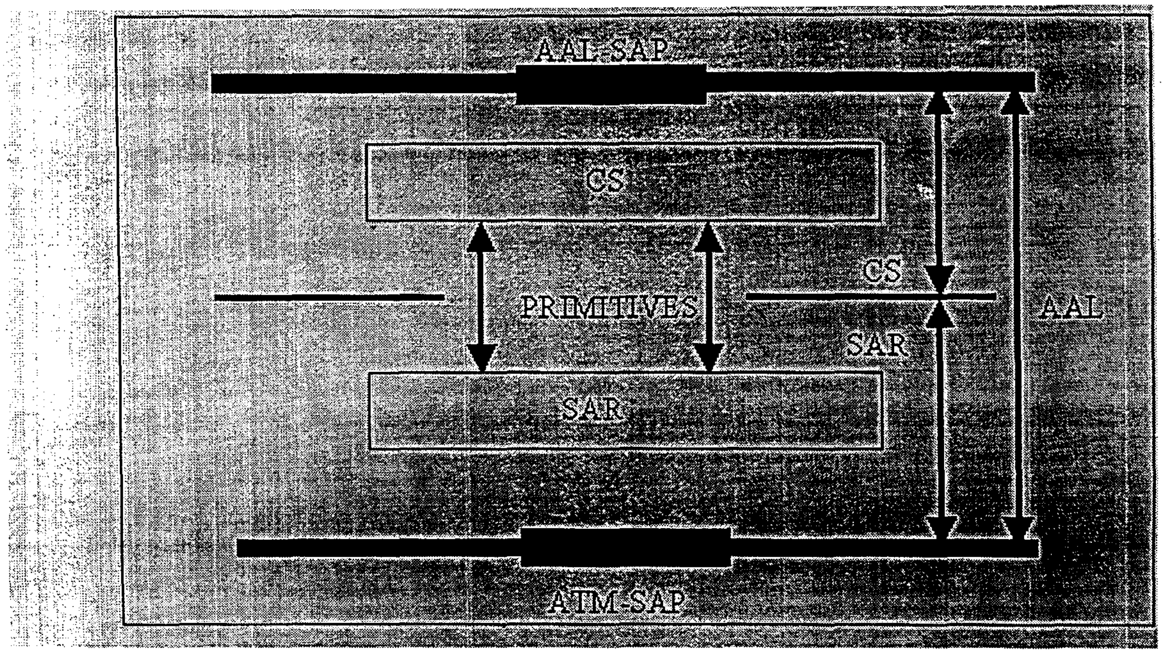

Figure 2-5.General structure of AAL

The SAR layer is concerned with the segmentation of higher-layer information into a suitable size for the information field of an ATM cell and for reassembly of the contents of ATM cell information fields into higher-layer information.

This is not enough to reconstitute the information sent and other functions have to be performed. These include processing of cell-delay-variation, end-to-end synchronization, handling of loss and misinserted cells. Such functions are performed by the convergence sublayer. This sublayer is service specific and different convergence sublayers may be used on top of the same SAR.

The functions of the AAL may be empty if the performance of the ATM layer is sufficient for the requirements of that particular telecommunications service. In this case, users are able to make use of all 48 octets in the information field. The addition of headers and trailers in each sublayer, together with the segmentation of user information to fit the ATM cell.

For specific needs in two AAL types, the convergence sublayer has been subdivided into two parts as following figure, the common part CS(CPCS) and the service specific CS(SSCS).Different protocols, to support specific AAL user services, or group of services, may be defined. The SSCS may be null, in the sense that it only provides for the mapping of the mapping of the equivalent primitives of the AAL to CPCS and vice-versa.

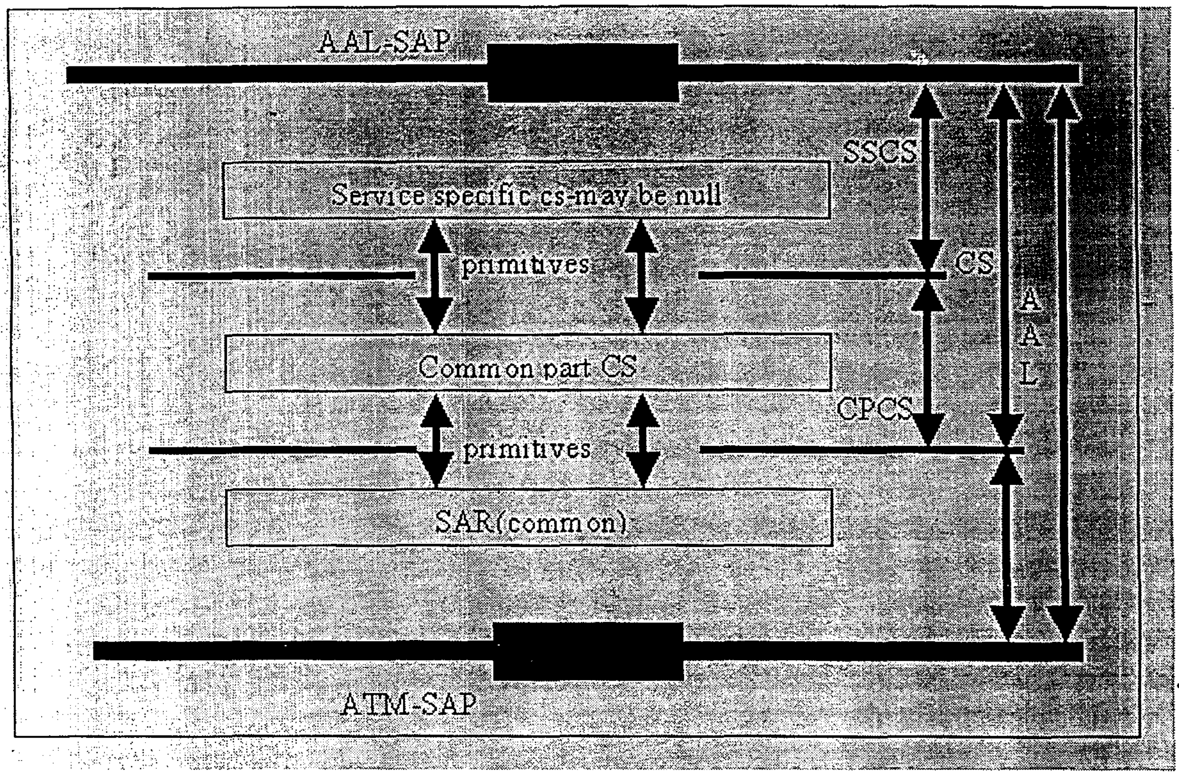

Figure 2-6. Structure of AAL with SSCS and CPCS (used in AALs 3/4 and 5)

Definitions of the functions and protocol within the AAL follow the usual naming conventional associated with layered systems with protocol data units (PDUs) and service data units (SDUs). The SDU passes across service access points (SAPs) whereas the PDU is the data unit between peer layers. For AAL type 3/4 and type 5, which both have the SSCS and CPCS, there is a PDU defined for each of this parts, with the appropriate header and trailer. Because no service access point is defined between the CS and SAR sublayers, there is no 'SAR SDU'.

2.3.2. AAL Type 1

AAL 1 provides the following layer services to the AAL user:

(i.) Transfer of service data units with a constant source bit-rate and their delivery with the same bit-rate.

(ii) Transfer of timing information between source and destination.

(iii) Transfer of structure information between source and destination.

(iv) Indication of lost or errored information which is not recovered by AAL 1, if needed.

The functions listed below may be performed in the AAL in order to enhance the layer service provided by the ATM layer. The SAR-PDU format is given in following figure.

(i) Segmentation and reassembly of user information.

(ii) Handling of cell delay variation.

(iii) Handling of cell payload assembly delay.

(iv) Handling of lost and misinserted cells.

(v) Source clock frequency recovery at the receiver.

(vi) Recovery of the source data structure at the receiver.

(vii) Monitoring of AAL-PCI(protocol control information)for bit errors.

(viii) Handling of AAL-PCI bit errors.

(ix) Monitoring of the user information field for bit errors and possible corrective action.

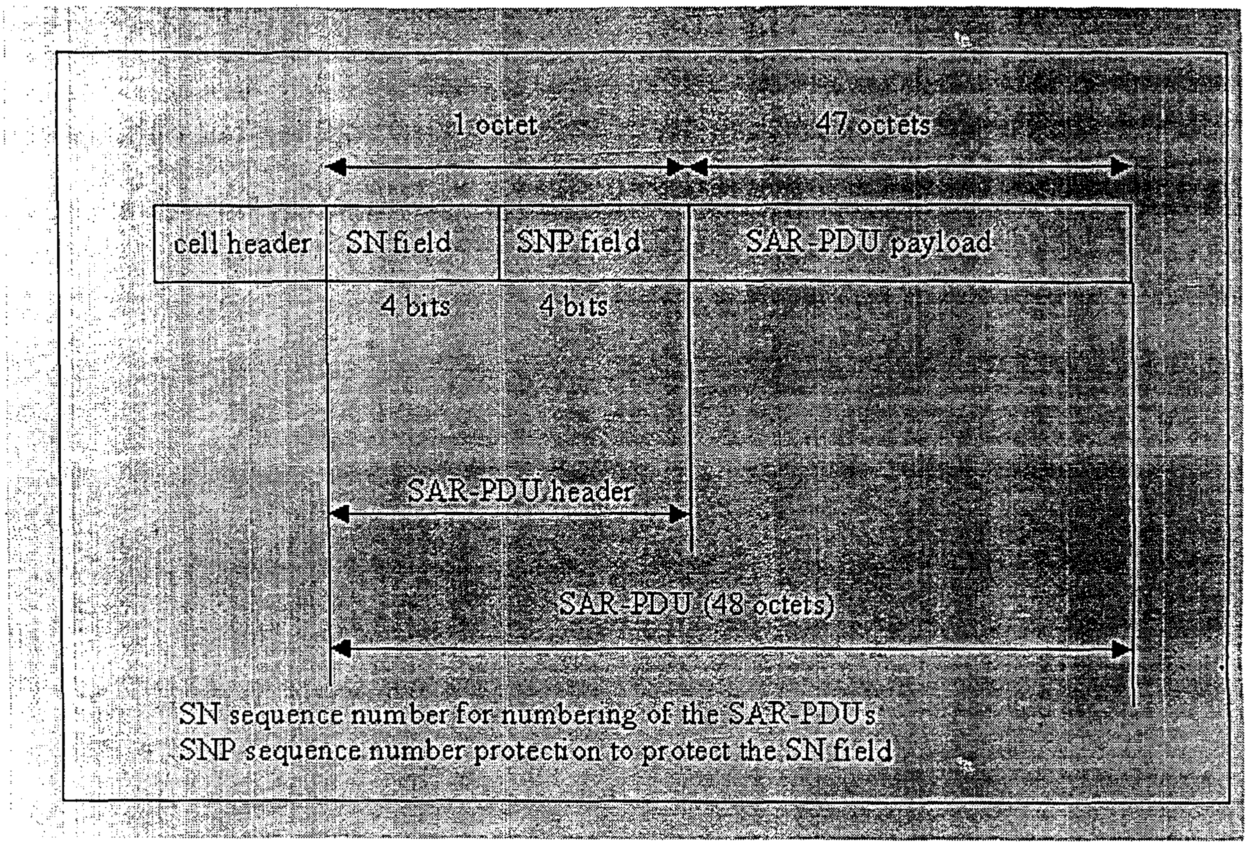

Figure 2-7. SAR-PDU format of AAL 1

At the SAR level, the SNP field, which provides 1-bit error correction and 2-bit error detection, is processed. If the result is right (no error detected or error detected and corrected), the SN field is sent to the CS level, which processes it depending on the application. Four CS have been identified for the following applications:

(i) Circuit transport to support both asynchronous and synchronous circuits. Examples of asynchronous circuit transport are 1.544, 2.048, 6.312, 8.448, 32.064, 34.368, 44.736 Mbps. Examples of synchronous circuit transport are signals at 64, 384, 1536, and 1920 Kbps.

(ii) Video signal transport for interactive and distributive services.

(iii) Voice-band signal transport.

(iv) High-quality audio signal transport.

2.3.3. AAL Type 2

It will be define later for small packet Encapsulation and Multiplexing in this dissertation.

2.3.4. AAL types 3/4

AAL 3 was designed for class C services (connection-oriented data) and AAL 4 for Class D services(connectionless-oriented data). During the standardization process, the two AALs were merged and are now the same. The current name given to this AAL is AAL 3/4.

Two modes of service have been defined: message and streaming. These are explained below.

In message mode as the following figure, the AAL service data unit is passed across the AAL interface in exactly one AAL interface data unit (AAL-IDU). This service provides the transport of fixed length or variable length AAL-SDUs.

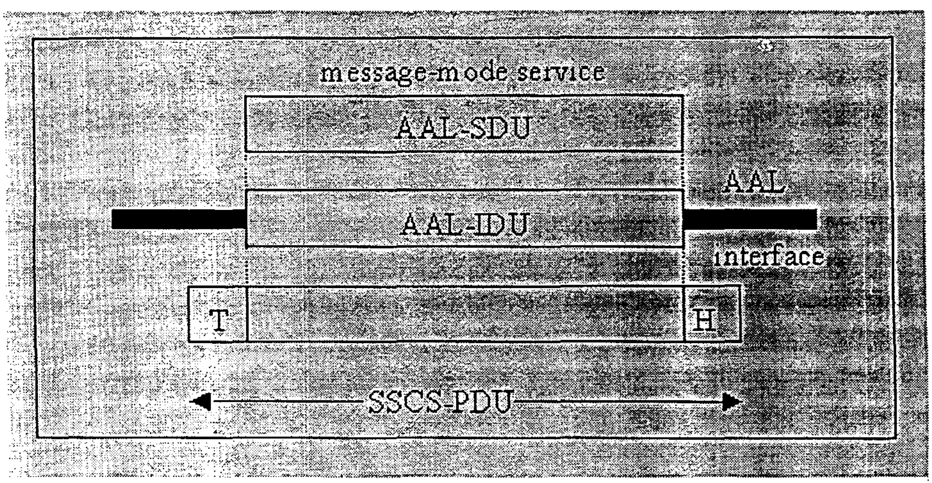

Figure 2-8. Message mode service

In streaming mode as the following figure the AAL service data unit is passed across the AAL interface in one or more AAL interface data units. The transfer of these AAL-IDUs across the AAL interface may occur separated in time and this service provides the transport of variable length AAL-SDUs.

The streaming mode service includes an abort service by which the discarding of an AAL-SDU partially transferred across the AAL interface can be requested.

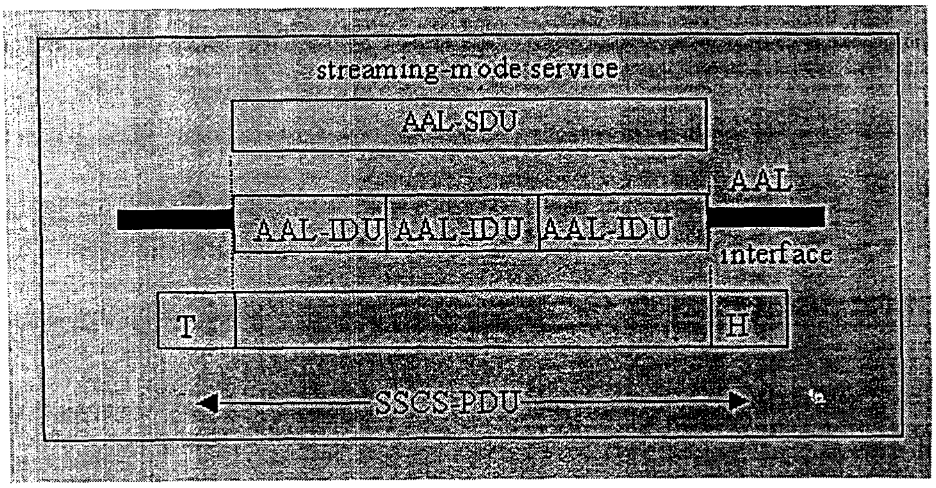

Figure 2-9. Streaming-mode service

<u>The Segmentation and Reassembly Sublayer(SAR)</u>

The SAR sublayer is depicted as following figure. The SAR sublayer accepts variable-length CS-PDUs from the convergence sublayer and generates SAR-PDUs with a payload of 44 octets, each containing a segment of the CS-PDU.

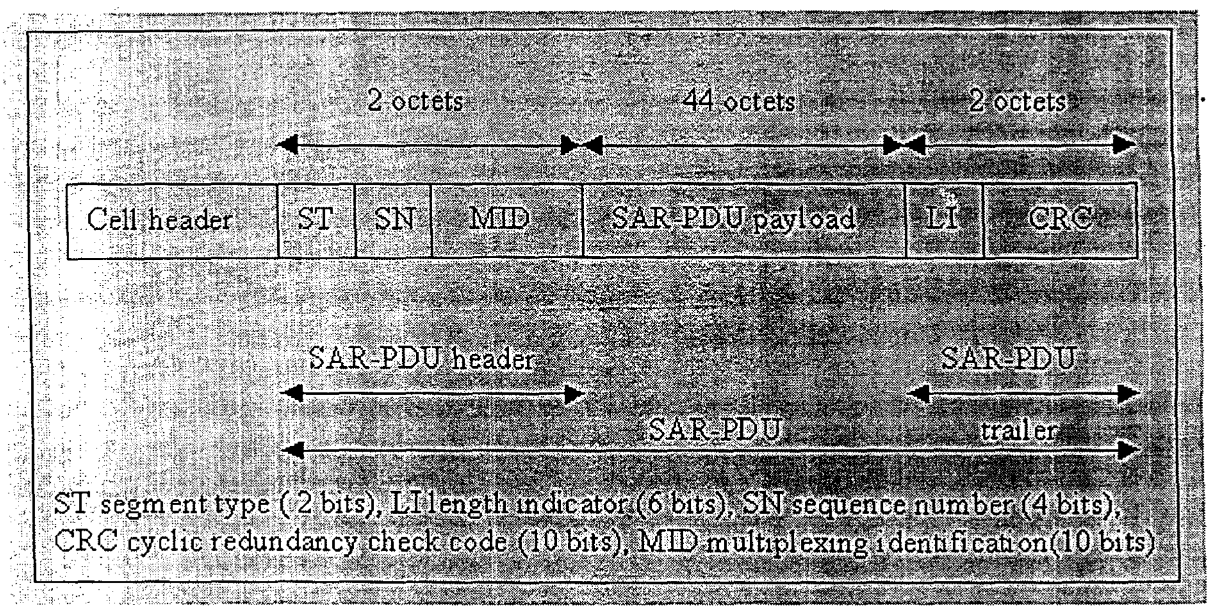

Figure 2-10. SAR format of AAL 3/4

The segment type indication identifies a SAR-PDU as containing a beginning of message, a continuation of message, an end of message or a single segment message. The sequence number allows the sequence of SAR-PDUs to be numbered modulo 16.These two fields enable the segments of the CS-PDU to be re-assembled in the correct sequence and minimize the effect of errors on the reassembly process.

The multiplexing identification is used to identify a CPCS connection on a single ATM-layer connection. This allows for more than one CPCS connection for a single ATM-layer connection. The SAR sublayer, therefore, provides the means for the transfer of multiple, variable-length CS-PDUs concurrently over a single ATM layer connection between AAL entities.

The length indication contains the number of octets of CS-PDU information that are include in the SAR-PDU payload field. This is necessary because the amount of date from the CS-PDU may not completely fill 44 octets available.

The CRC field is a 10-bit sequence used to detect bit errors across the whole SAR-PDU. This includes the CS-PDU segment and hence the user data.

The Convergence Sublayer (CS)

The CS has been subdivided into the common-part CS(CPCS) and the service-specific CS (SSCS).

The functions of the CPCS are:

(i) Preservation of SSCS-PDUs
(ii) Error detection and handling
(iii) Buffer allocation size
(iv) Abort

The CPCS requires a 4-octet header and a 4-octet trailer. The padding field provides a 32 bit alignment of the CPCS-PDU payload.

2.3.5. AAL type 5

This AAL was introduced in the study process of CCITT at the end of 1991. Its description is almost complete and will be published in the 1994 CCITT recommendations. Designed for the same class of service as AAL 3/4, it has the advantage of being simpler and requiring less overhead. Unlike AAL 3/4, it allows all 48 octets of the cell information field to be used for the transport of CS-PDU segments, the only SAR protocol information being provided by a bit in the ATM cell header, as explained below. This means that there is neither multiplexing nor error control at the SAR sublayer. However, there is a CRC field in the CS sublayer.

There are also similarities with AAL 3/4. The two modes of service defined, message and streaming, are the same as those for AAL 3/4. Also similar to AAL 3/4, the convergence sublayer of AAL 5 has been subdivided into a CPCS part and a SSCS part. Following figure is showing SAR format.

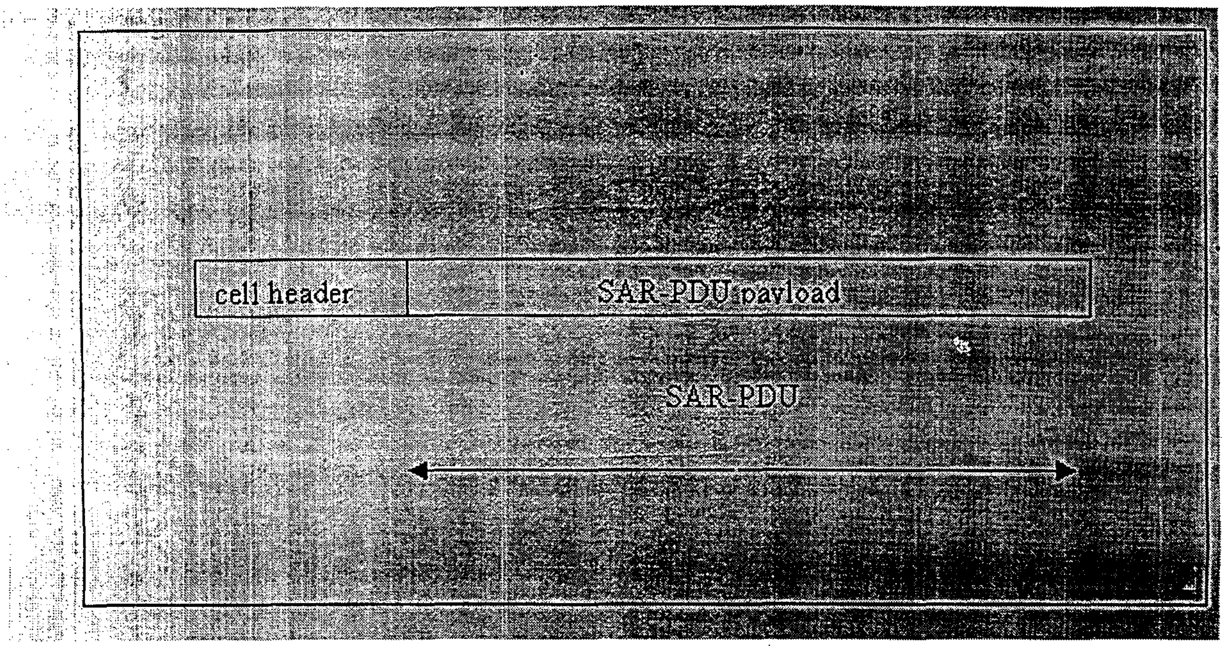

Figure 2-11. SAR format of AAL 5

The protocol control information of the SAR sublayer uses the ATM-layer-user-to-ATM-layer-user parameter (AUU) contained in the ATM header to indicate that a SAR-PDU contains the end of a CS-PDU. When this bit is set to 1, it indicates the end of the CS-PDU, when set to 0 it indicates the continuation or the beginning of a CS-PDU. This is necessary to enable the SAR to cope with reassembly of the CS-PDU in the presence of errors. If no indication of the end of the CS-PDU was provided the loss of a cell, and hence the loss of a segment of the CS-PDU, would mean that all subsequent reassembly operations would be incorrect. By indicating the end of the CS-PDU, the loss of a single cell would limit the error to one CS-PDU, unless the lost cell contained the end indication in which case the error would be restricted to 2 CS-PDUs.

The format of the CPCS is shown in following figure. The main CPCS functions are preserving the SSCP-PDU, providing CPCS user-to-user indication, detecting and handling errors, providing an 'abort' function and also padding where necessary.

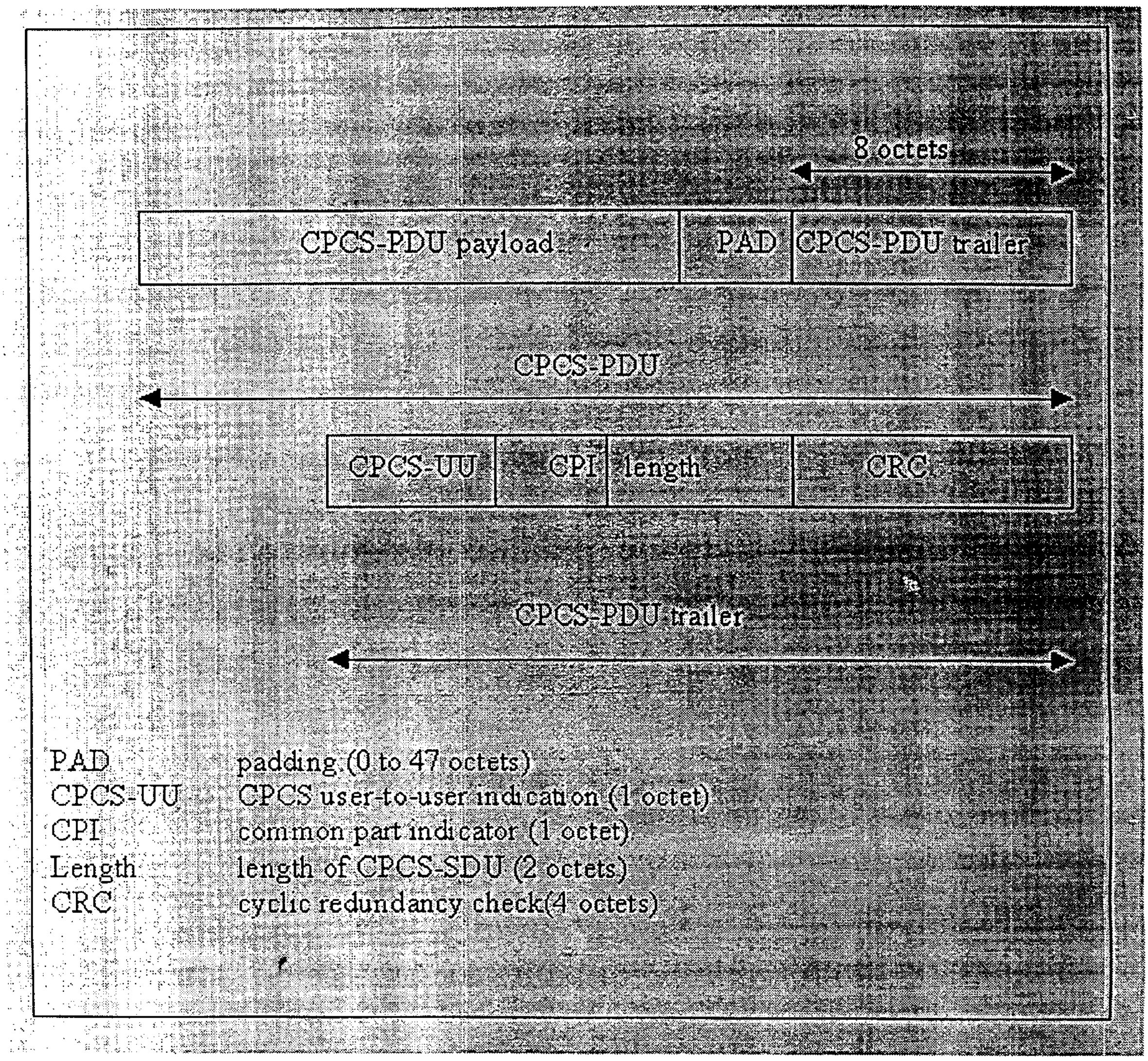

Figure 2-12. CPCS-PDU format for AAL 5

CHAPTER 3

ENCAPSULATION AND MULTIPLEXING IN AAL-2

AAL-1 has been defined for constant bit rate (CBR) traffic requiring tight delay and jitter control. In addition, AAL-3/4 and AAL-5 have been defined for bursty data. These AALs allow simple encapsulation of application packets if each packets fits into one ATM cell. For larger application packets, the segmentation and reassembly layer in AAL-3/4 and AAL-5 allows segmentation of packets at the transmitter so each segment can fit into an ATM cell and the original packet can be assembled from the incoming ATM cells at the receiver. Thus, the existing AALs allow either collection of enough information to fit into one ATM cell payload or segmentation of larger native mode packets into smaller units such that each smaller unit fits into an ATM payload. If native information units are smaller than an ATM payload, these AALs require partial filling of ATM cells.

Many applications require transmission of small packets across an ATM network involving one or more ATM switches in the connection. These applications include private branch exchange (PBX)-to-PBX trunking for compressed voice with or without silence suppression, ATM backbone for cellular systems and personal communications services (PCS) wireless access, ATM trunking on the public-switched telephone network (PSTN, and ATM backbone connectivity to packet telephony.

There are two primary reasons to transmit small packets across ATM networks:

(i) When small native packets are generated away from the ATM network (i.e. in a digital cellular mobile terminal) and the packet boundaries need to be recovered at the destination outside the ATM network.

(ii) When the bit rate of the native application is low and the requirement on the end to end delay prohibits accumulation of bits to fill an ATM cell before sending the cell out to its destination. In the latter case, smaller packets are generated even if the packetization is done at the ATM network edge.

For these application, partial filling of ATM cells when using existing AALs may cause unacceptable loss in bandwidth efficiency. This inefficiency is a concern especially when the total traffic demand requires relatively low speed leased lines, which have high cost relative to bit rate. In many cases, the cost penalty may nullify many advantages offered by an ATM backbone.

We know that many native mode connections (for example, cellular voice calls) may share two end points of the ATM network. In such a case, one ATM connection between two points to carry packets from multiple native connections. The ATM payloads from successive cells of this ATM connection are used as a byte stream on which packets from different native connections (called logical link connections[LLCs]) are packed without regard to the cell boundaries. A connection identification (CID) field is used in the packet header to identify the LLC to which a packet belongs. A length indicator (LI) field is used to identify the boundaries of variable-length LLC packets.

These concepts were incorporated in a protocol called the small packet multiplexed AAL (SMAAL). It became apparent that many other trunking applications would benefit from this protocol. Many alternative formats based on concepts in SMAAL were developed and evaluated. The basic difference in these alternatives are the coverage and amount of error protection and the size of the CID field (equivalently, the number of LLCs to be supported in a single ATM connection). The evaluation resulted in recommendations that are being standardized as the (AAL-2). Following figure shows the concepts involved in SMAAL (and AAL-2) at a high level.

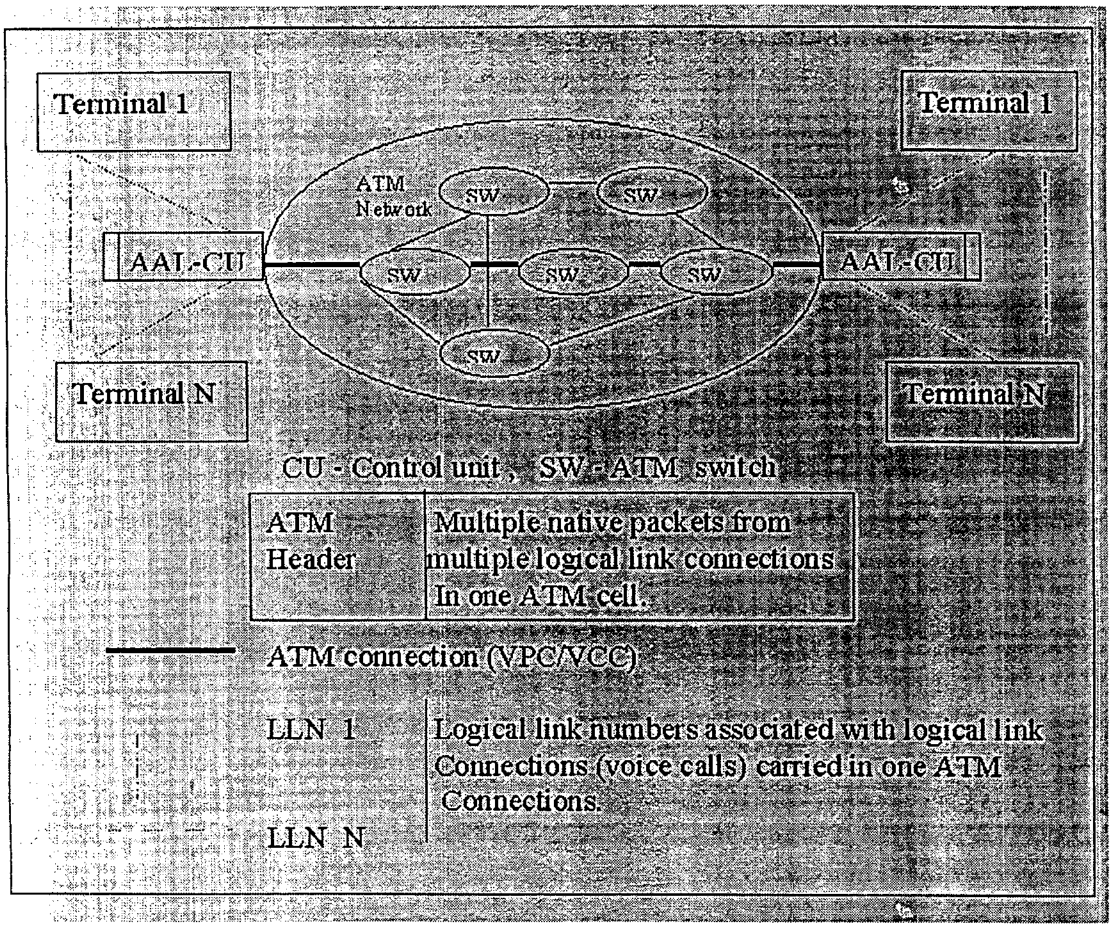

Figure 3-1. The basic concepts in AAL-2

When both multiple sources and destinations are involved, it is sometimes desirable to implement a LLC rebundling function in which AAL-2 is terminated and packets for different LLCs are extracted and rebundled into new ATM connection for transmission to their destinations.

The mix of LLCs in ATM connections will change as calls are set up and torn down. Moreover, it may be desirable to change the coding rate or signal the start or end of silence to the receiver. Doing so would require a signaling protocol that is transparent to the ATM connection.

3.1. SOME APPLICATION SCENARIOS

Details of some of the application scenarios motivating the AAL-2 are now described. The list is not exhaustive, but it represents a few classes of applications with similar characteristics.

PBX-to-PBX Connectivity over an ATM Backbone Network

ATM has become a promising infrastructure technology for wide area network(WANs) inter- connecting PBXs and other private local networks. Following figure shows a scenario in which two PBXs are connected over an ATM backbone network.

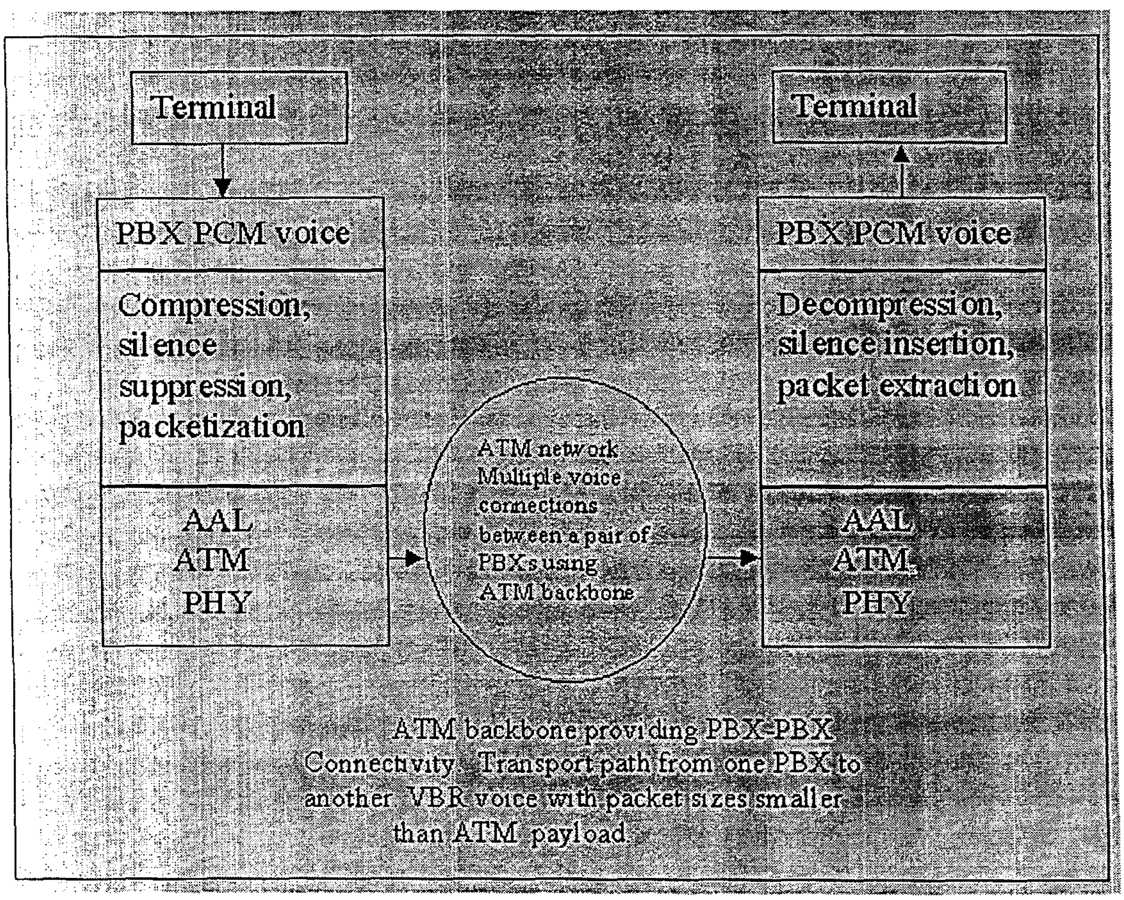

Figure 3-2. PBX-to-PBX connectivity with the ATM backbone

With the current circuit-switched backbone using STM technology, PBXs are connected using leased lines(typically, one or more T1 or E1 lines).The ATM backbone can be used to provide a circuit emulation services(CES) at either the T1 or E1 rate at any other transmission speed depending on the number of 64-Kbps trunks engineered between the two PBXs. In this situation, the main advantage ATM provides is arbitrary sizing of the trunk group. PBXs are not aware of the existence of an ATM backbone treat it like an STM backbone. Leased bandwidth that is not used by the PBX pair is wasted, as it is with the leased lines available today.

An ATM backbone allows further opportunities for efficiency improvement. ATM bandwidth can be allocated as required for active calls. Voice compression and speech activity detection can be used to reduce the required bandwidth further. However, transport of compressed voice poses challenges for existing AALs, especially when silence suppression is also exercised.

With voice compression to 32kbps, a packetization delay of 12ms is incurred to fill an ATM cell. At 8Kbps,the corresponding packetization delay is 48ms. The latter is prohibitive even with echo cancellation and the former is unacceptable without echo cancellation. On the other hand, the smaller packetization interval will result in smaller packets. A AAL is required to carry these small packets efficiently.

A similar situation arises in PSTN trunking. ATM is an attractive technology, especially for international circuits in which expensive bandwidth makes voice compression and silence suppression desirable. Delay consideration results in small packets.

ATM Infrastructure for Wireless Cellular Systems and PCS Access

ATM is a promising technology for the wired infrastructure in cellular systems and PCS. Following figure illustrates an ATM-based infrastructure having transport, switching, signaling, and operations, administration, and maintenance (OA&M) functions.

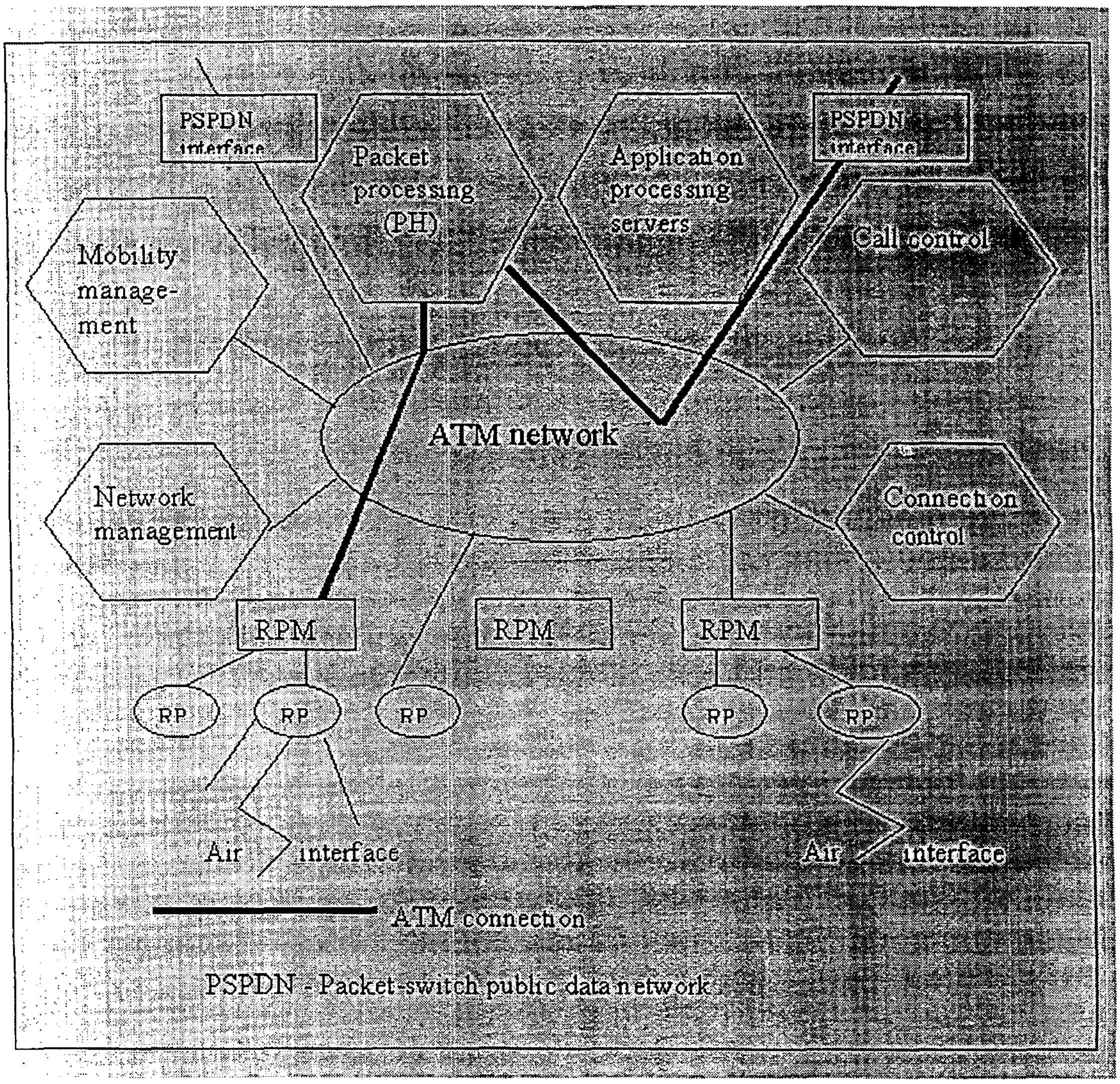

Figure 3-3. ATM infrastructure for wireless cellular systems and PCS access

The air interface is left untouched while the connectivity between the termination for the air interface(radio port[RP]) and the PSTN is provided over an ATM backbone. The ATM backbone itself may consist of ATM multiplexers (RP multiplexers(RPMs]) and ATM switches. For each call, the associated RP becomes one of the two end points of an ATM connection and the home of the AAL. The other end of this ATM connection is at a packet handler (PH), which supports multiples RPs.

The packets on the air interface typically are very small, and they may vary in size(2 to 36 octets for code division multiple access[CDMA]). The packet boundaries need to be preserved across the ATM network. Once again, existing AALs will fill each ATM cell partially and waste a significant fraction of the bandwidth on the ATM connection.

Packet Telephony in Other Places

The existence of the fast packet switching technologies like frame relay and ATM coupled with advances in voice compression technology have motivated the use of packet telephony in other situations.

We consider desktop telephony using a PC or a workstation as the end equipment and the Internet being used to bypass the toll part of the PSTN. The connection between the end equipment and the Internet access provider may consist of "plain old telephone service (POTS) modems", hybrid fiber coax (HFC) cable modems, integrated services digital network(ISDN)access, asymmetric digital subscriber lines(ADSLs), or wireless links. In the case of a POTS modem and ISDN access, the typical mode of operation consists of voice packets from the end equipment in Internet Protocol(IP) datagrams being encapsulated in point-to-point protocol(PPP)frames.

Assuming that IP datagrams are used to carry voice packet from the end equipment to the Internet access providers, many options exists for transporting these packet to their destinations. The Internet access provider implements the IP-to-ATM conversion and uses an ATM backbone network. For high-quality Internet telephony, the voice packet size will decrease, making existing AALs on the ATM backbone unacceptably inefficient.

If ATM is extended all the day to the desktop, voice packets from the end equipment can be carried directly over ATM using AAL-5. However, at the gateway to the backbone, existence of multiple connections creates an opportunity to terminate AAL-5 and carry voice packets from multiple connections using a more efficient protocol.

Even for traditional telephony using analog or ISDN access, a PSTN-to ATM gateway creates an opportunity to compress voice, suppress silence, and carry the resulting lower and variable bit rate voice over the ATM backbone.

3.2. BASIC CONCEPTS AND REQUIRED FEATURES

The primary requirement on the new AAL is to provide efficient transport of small native packets over ATM networks in such a way that allows a very small transfer delay across the ATM network while still allowing the receiver to recover the original packets. It is assumed that multiple LLCs are required between any pair of gateways or TAs, where conversions between the native mode and ATM occur.

One approach is to use an AAL in which a small but fixed number of octets from each LLC are packed in one ATM connection. For example, three octets each from 16 LLCs may be used to pack the 48-byte payloads of ATM cells from a single ATM connection. If at least 16 LLCs are active between two TAs, then packing small 3-byte samples each from 16 different LLCs keeps the packetization delay small and still achieves high bandwidth efficiency. The position of a triplet within the payload of an ATM cell identifies the LLC associated with that triplet. This identification method is referred to as position-based multiplexing and delineation. In the extreme case, 48

LLCs (with one byte from each LLC in each ATM cell) may be multiplexed in a single ATM connection. The mapping between the position and LLC is established by providing or by using an appropriate signaling protocol at call set-up time.

While being very efficient, this approach suffers from a complete lack of flexibility. In particular, it is effective for constant bit-rate applications in which all LLCs have a common rate and a fixed number of LLCs is available. However this perhaps the only scenario in which the method is effective. If different coders are used by different LLCs, this approach will create a significant packing inefficiencies as calls are setup and torn down. If available bit-rate coders or silence suppression are used, then either variable-length packets are generated by LLCs or variable packetization delay is incurred to generate fixed-length packets. In either case, the inflexible position-based multiplexing approach can not be applied with ease.

The AAL-2 should support the transfer of variable size small packets across ATM networks. The selected approach is to treat the payloads from successive ATM cells from the same ATM connection as a byte stream in which variable length LLC packets are multiplexed. The following functions must also be implemented:

(i) Addition of CID field to each LLC packet, to identify the LLC to which the packet belongs.

(ii) Addition of an LI field to each LLC packet, so that the end of the packet can be demarcated. This LI-based approach to the delineation of packets was considered better than flag-based delineation because flags are vulnerable to errors and because the bit stuffing required by flag based protocols conflicts with the byte-oriented nature of ATM transport.

(iii) Inclusion of a pointer to identify the remaining length of the packet that started in the prior cell and overlaps into the current cell when the native occupies part of two (or more) cell payloads. This is similar to an LI but refers to the length of the residual packet in the current cell.

(iv) Incorporation of some mechanism to detect cell loss, because loss of an ATM cell possibly may lead to misconcatenation of native packets.

(v) Use of some mechanism to detect errors in the cell payload, because critical address (CID) and delineation (LI, starting pointer field) information is now part of the unprotected cell payload.

Many alternatives are consistent with the protocol concepts described above. The main decision variables are:

(i) The maximum number of CIDs to be supported in one connection. This in turn determines the length of the CID field.

(ii) The maximum length of the native packet. This determines the length of the LI field.

(iii)self-delineation of packet via the LI field versus the need for a separate mechanism for faster delineation recovery.

(iv) The need foe error protect the packet payload in addition to the protection of the packet header.

(v) The mechanism for providing resiliency to cell losses on a given ATM connection.

3.3 DESIGN CHOICES AND KEY DECISIONS

Because the primary driver for this new AAL is packet telephony and because error detection is not essential for voice coding algorithms, it is not necessary for AAL-2 to provide error detection for native payloads. The purpose of error detection is to guarantee that CID, LI, and the other critical protocol header fields are not misinterpreted. Avoidance of such interpretation could be accomplished either by having a common cyclical redundancy check (CRC) on the ATM payload or a header error check(HEC) for each packet header.

The latter has the advantage of being able to discard only those packets whose headers are corrupted. The former has the advantage of having one CRC for multiple packets and potentially higher bandwidth efficiency while also permitting in-band signaling messages to be carried without additional error detection. The advantages of packet header error detection were determined to outweigh a potential bandwidth penalty, and a per-packet HEC approach was selected for AAL-2.

The maximum number of LLCs to be supported on a single ATM virtual path connection (VCC) affects the size of the CID field. A larger CID field would allow multiplexing more native connections (LLCs) and would provide bandwidth sharing over one more variable bit rate connections. Of course, the numbers of LLCs to be supported also depends on the ratio of the ATM connection bandwidth to the bandwidth required by a single native connection. Using a DS3 link, the number of connections that can be supported increases to the neighborhood of 16,000. Of course, the ATM header allows two levels of addressing (virtual path identifier [VPI] and virtual circuit identifier [VCI]). Thus, it is possible to set up ATM VPCs between AAL-2 end points and to allow them to use VCI and CID to create multiple native connections(LLCs). This situation permits multiplexing with a smaller CID field. As a compromise a CID field of 8 bits supporting a maximum of 255 native connection over a single ATM VCC was chosen. With a 16 bit VCI field an ATM VPC will be able to support up to 255×2^{16} connections, certainly adequate for all applications. In this case, ATM traffic policing will need to be done at the VPC level. The LI field refers to the length of the native packet and is used to delineate the packets. Because AAL-2 was designed for small packets, the length of an ATM cell was thought to be the upper bound on length. In that case, a 6-bit LI field allowing specification of up to 64 octets would suffice. For applications involving occasional packets longer than 64 octets, a MORE function is proposed to indicate an extension beyond 64-octets. While the LI field allows self-delineation once a packet boundary is identified, a cell loss or an error in a packet header will result in the loss of packet delineation. Many approaches to regaining the packet boundaries are possible, including:

(i) Using a packet header HEC to hunt for and identify a packet header. The LI field is used to delineate further the end of the packet and verify that the delineation is achieved. This process is similar to cell delineation at the ATM layer but with variable length packets.

(ii) Using a pointer field at the beginning of each ATM cell to specify the beginning of the first new packet in the current cell payload.

(iii) Using the pointer at the beginning of every m(>1) cells.

The first approach has no additional overhead, but it may take longer to regain packet delineation. In addition, loss of delineation is more frequent. The second approach incurs additional overhead but guarantees packet delineation in one cell time. The third approach adds the overhead only once every m cells, but it may require up to m cells to regain delineation. Because packets received during the time it takes to regain delineation are delineation are discarded, the second approach is preferred. Given that a loss of cell (if not detected at the receiver) can misconcatenate packets, a cell-sequence number field is desirable. Finally, like the packet header (containing CID, LI, and other field), the cell sequence number and pointer also require error detection. A single parity bit provides this error detection. The combination start bit (STF), consisting of the pointer, cell sequence number, and parity is placed at the beginning of every ATM cell payload. The pointer indicates that the first new packet header in the current cell. The HEC in the header of that packet is used to verify synchronization, The robustness afforded by this HEC and those in subsequent packets allows the use of a simple parity check in the STF.

ATM Cells AAL-2 Fields

Following figure shows the placement of the STF and the packet header. The STF is present at the beginning of each ATM cell. The packet header precedes each native packet.

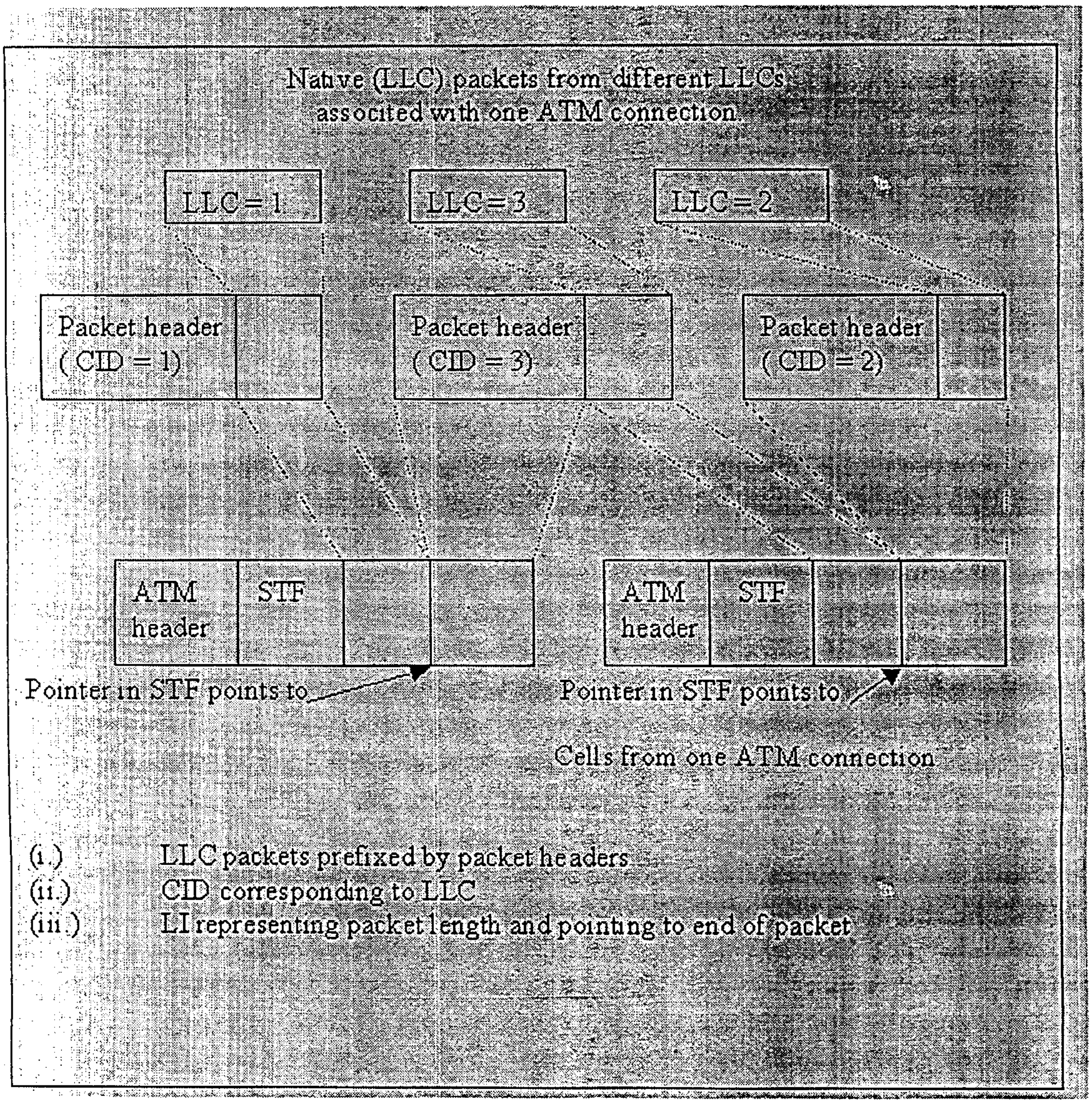

Figure 3-4. Small packet multiplexing using the STF and packet header fields

3.4. AAL-2 COMMON PART DETAILS

The AAL-2 is partitioned into two sublayers: The common-part sublayer (CPS) and the service specific convergence sublayer (SSCS).

The CPS provides the function of multiplexing of variable length packets from multiple sources into a single ATM virtual circuit and relaying these packets to form end to end AAL-2 connections.

The SSCS provides an application specific function, a different instance of it being provided to each AAL user. Some example functions are segmentation and reassembly of user flows into packets suitable for the CPS, forward error control, identifying the end of a speech burst.

Components of the Packet Header

The packet header, which the following figure shows, is 3 octets long. The CID field is 8 bits long and identifies the LLC for the packet. The LI field comprises 6 bits and indicates the length of the LLC packet. When the LI points beyond the end of the current ATM cell, the packet is split between cells. The HEC field comprises 5 bits and provides error detection over the packet header.

Five bits are either reserved (RES) or assigned to the SSCS riding over the AAL-2 common part and are passed transparently from the transmitter's SSCS to the receiver's SSCS. The SSCS can use these bits for specific SSCS functions or for passing higher layer user-to-user communication transparently.

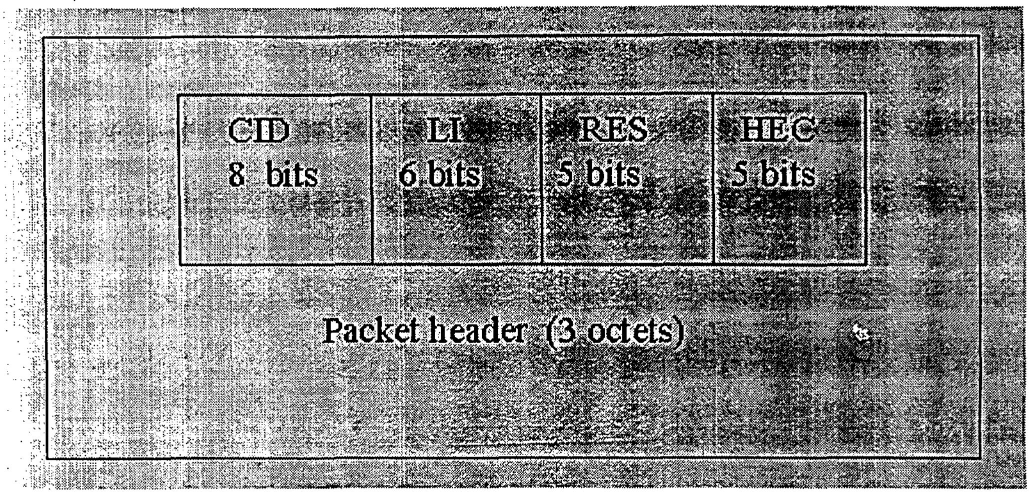

Figure 3-5. Components of the packet header

Components of the Start Field (STF)

The STF is one octet in length and occurs at the beginning of every ATM cell payload. As the figure showing, the offset field (OSF) is 6 bits in length. It indicates the remaining length of the packet that (possibly) started in the preceding cell from this ATM connection and is continuing in the current cell. Thus, the OSF points to the start of the first new packet and provides immediate recovery of the packet boundary after an event causing loss of packet delineation.

The 1-bit sequence number (SN) field provides a modulo-2 sequence numbering of cells. The one parity (P) bit provides odd parity and covers the STF.

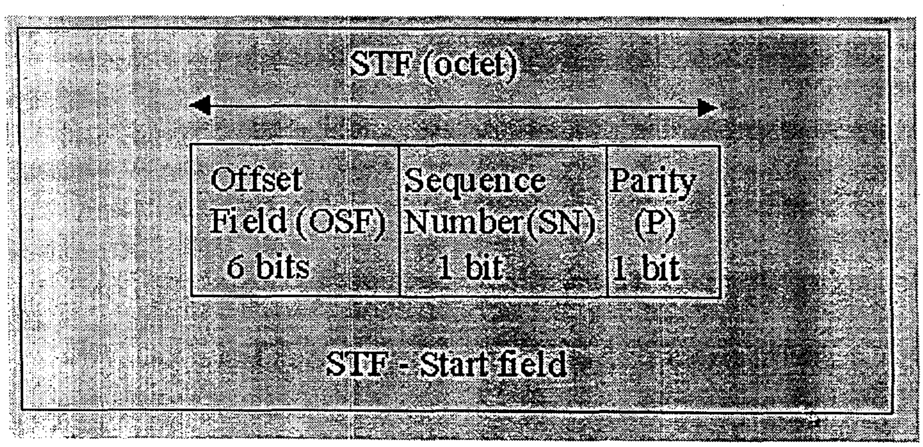

Figure 3-6. Components of the STF

Padding

It may be necessary to transmit a partially filled ATM cell to limit packet emission delay. In such a case, the remainder of the cell is padded with all zero octets. A cell whose payload contains only the STF and 47 padding octets can be transmitted to meet other needs, such as serving a keep alive function an satisfying a traffic contract.

3.5. ANALYSIS OF BANDWIDTH EFFICIENCY

There are two wireless applications the IS-95 CDMA rate set 2 vocoder and the Japanese personal digital cellular (PDC) half rate vocoder. Performance of AAL-2 based transport for these two applications is compared with performance when using frame relay and STM transport. The analysis provides insight into the desired CID field size and into dimensioning ATM virtual connections to take advantage of statistical multiplexing gains made possible by the AAL-2.

Traffic and Models

To determine the bandwidth efficiency and delay/jitter tradeoffs, specific traffic patterns and speech models are used. It is assumed that only voice traffic is carried on the ATM interconnections of interest. In practice, some in band signaling and OA&M traffic may exist but their overall impact will be slight.

CDMA Rate Set 2 Vocoder

Various levels of coding have been defined for the CDMA rate set 2 with a vocoder full rate of 13Kbps. A simplified speech model is used here for the studies. Specifically, 50% of the packets are at the full encoding rate of 14.4 Kbps (active speech, including overhead), while the remaining 50% are at one eighth rate of 1.8Kbps (silence). The overall average rate is 8.1Kbps. This coder produces a measured mean opinion score better than 3.95. Speech is accumulated for 20ms, encoded and transmitted over the air interface as a packet.

PDC Half-rate Vocoder

The same simplified speech activity model for the PDC half-rate vocoder produces an average rate of 2Kbps with a packet produced every 40ms at either the full rate (4Kbps) or complete silence (0Kbps) with equal probability. This vocoder provides an extreme point with a very low bit rate resulting in maximum bandwidth efficiency.

Requirements on Delay Variation

In IS-95 based CDMA technology, voice packets are produced every 20ms. Because of soft handoff, the reception of packets at the mobile must be synchronized perfectly. In addition, because of tight timing requirements, all base station must be synchronized using the Global Positioning System.

It is possible for all the mobile to transmit packets at every 20ms tick. In this case, the delay variation can be up to 20ms depending on the number of active calls. For interactive voice applications, experiments have determined that the one-way end-to-end delay must be in the range of 100 to 150ms. Because coding, interleaving, decoding and deinterleaving can consume a substantial portion this delay budget, an additional delay of 20ms is on the boundary of acceptable performance.

The transmission of packets from mobiles can be staggered such that one set of mobiles transmits at a given time tick, a second set transmits exactly 5ms later, and the last set 15ms later. These are referred to as offset groups. There are two cases, one in which the requirement on the maximum delay variation is 5ms (that is, four different offset groups) and the second in which the maximum permissible delay variation is 20ms (that is one offset group).

Computation of Bandwidth Efficiency

Let the voice packet sizes produced by the sources (CDMA rate set 2 or PDC half rate) during the packetization interval T, be S_1 and S_2 octets. Let p_1 be the probability of producing a packet of size S_1 and p_2 be the probability of producing a packet size S_2. A binomial speech model has been assumed but the analysis is easily generalized to a multinomial model. Let N be the number of voice sources transmitting in an offset group from the mobile to the base station. At the base station, the probability distribution of the number of octets received in an interval of T ms is

$$P\{K=iS_1+(N-i)S_2\} = {}^{N}C_i p_1^{\,i} p_2^{\,N-i}$$

The mean number of octets m received in an offset group is given by

$$m = Np_1S_1+Np_2S_2$$

The variance of the number of octets, denoted by V, is given by

$$V = Np_1p_2(S_1-S_2)^2$$

The standard deviation is given by

$$S = \sqrt{V}$$

The transport network is assumed to engineered to meet a packet loss objective of 0.1% while meeting the delay variation objective of either 5ms or 20ms. Given that the packet loss rate objective on the air interface is 1%, a transport network objective one order of magnitude better than the air interface results in an excellent balance between performance and efficiency.

Let b denotes the capacity in octets allocated to the ATM virtual connection that transports the N voice calls. The capacity, b, is selected so that the maximum delay variation for transmission through the ATM connection is less than y(5 or 20) ms, that is, b=cy, where c is the capacity of the link (T1 or T3) in octets per ms after subtracting the per cell overhead due to the ATM cell header and the AAL-2 STF. To achieve a packet loss less than 0.1% the engineering rule (assuming a normal distribution for number of octets produced) is m+3s=b, which translates to

$x^2+x\alpha-\beta = 0$, where

$$x = \sqrt{N} ,$$

$$\alpha = 3\sqrt{p_1p_2}(S_1-S_2)/(p_1S_1+p_2S_2)$$

and

$$\beta = b/(p_1S_1+p_2S_2)$$

Solving the above equation determine the number of voice calls that can be carried in a single ATM virtual connection to meet the loss objective of 0.1% and the delay variation objective of y ms.

Results and Discussion

First, we consider the CDMA rate set 2, which produces packets containing 36 octets and 5 octets with equal probability. The AAL-2 adds an overhead of 3 octets to each octet, so s1=39 and s2=8. The effective ATM cell payload is 47 octets because the first octet is used as the STF in every cell. In the case of frame relay, there is an overhead of 6 octets for each packet. (The Frame Relay Forum is currently standardizing an approach similar to the AAL-2 to carry multiple small packets within one data link connection identifier (DLCI). The efficiency gains from this are not considered here.) Frame-relay overhead consists of one octet for flag, two octets for the DLCI field, one octet for control, and two octets for the frame check sequence.

Frame relay, with its variable size frame, is ideally suited for carrying variable size packets generated by low bit rate voice. The AAL-2 provides a similar capability over ATM connections. It also allows the use of much higher speed ATM switches and link interfaces, thus allowing further multiplexing gain. Finally with higher speed interfaces ATM transport and switching are much less expensive than the frame relay counterparts. If ATM can achieve bandwidth efficiency comparable to that of frame relay, lower switching cost and the ability to support higher rate interfaces will favor ATM.

3.6. ROBUSTNESS CONSIDERATION IN THE DESIGN OF AAL-2

The transmission error performance seen by AAL-2 users as functions of raw channel transmission performance and various AAL-2 parameters. The analysis also motivates the use of the OSF, SN, and other packet header fields so that the protocols facilitate adequate performance for AAL-2 users. The basic impairment at the physical (PHY) and ATM layers arises from transmission errors and cell losses.

We know that two methods of delineation recovery are possible: self-delineation through HEC hunting and delineation through the OSF field in the start pointer (STF) of every cell. Because we need both the HEC and LI for normal receiver operation after initial delineation is achieved, an obvious question is whether the overhead of the STF is necessary to accelerate the recovery.

Self-delineation

On the surface, it appears that AAL-2 could use only the self-delineation provided by the LI and HEC fields. HEC allows hunting to decide potential packet boundaries. The LI field of the presumed packet header is used to delineate the position of the next packet header and HEC. This is similar to the self-delineation function provided by the ATM HEC. However, variable-length packets and the use of the LI both affect the frequency of delineation loss and recovery delay.

When packet delineation is lost, an octet-by-octet search is used to find the first three-octet packet header (i.e. the location of the first three octets at which the HEC passes). The LI values indicated by the detected packet header are used to find the next packet header. AAL-2 is declared to be in packet delineation after n such correct packet headers, with n chosen for a desired level of confidence.

Delineation Recovery

At the ATM layer, cell delineation is declared when the ATM HEC passes on seven consecutive cells, which corresponds to a confidence level of 10^{13} to 10^{16}. Similar confidence in AAL-2 delineation recovery is desirable. Following table shows the number of packet header HECs (CRC-5) before packet delineation can be declared with a given confidence level.

Number of packets	Probability of incorrect delineation
6	10^{-9}
7	10^{-11}
8	10^{-13}
9	10^{-15}
10	10^{-17}

Table: Self-delineation performance

To achieve a confidence level comparable to ATM header delineation, the number of packet headers that must pass should be set to 9 or higher. Thus, for every event in which packet delineation is lost, nine additional packets must be either discarded or buffered. In case of buffering, an additional delay must be budgeted for speech and other delay-sensitive applications. For voice applications, buffering of nine packets may be unacceptable. The implications of discarding nine packets depend on the frequency of occurrence of delineation recovery.

Loss of delineation

Loss of packet delineation occurs under the following two conditions.

(i) Any detected bit errors in the three-octet packet header: The packet header must be discarded so further packet delineation is not possible.

(ii) Any cell losses: A single cell loss or burst of cell losses implies that packet delineation is lost.

Let p be the probability of a bit error in a packet header and let p_c be the probability of a cell loss. Let k_p denote the average number of packets per ATM cell. Then, assuming nine packet headers must match for recovery of delineation, the packet discard ratio can be approximated by $9 \times (24p + p_c/k_p)$. In the calculation below, k_p is assumed to be 3.

On a noisy T1 link ($p=10^{-6}$), the packet discard rate due to bit errors is as high as 2×10^{-4}. If a VBR or CBR virtual circuit is used to multiplex voice sources, the bandwidth must be determined to obtain a target packet loss rate of 10^{-4}, aggregating the packet losses in lost cells, discarded packets due to bit errors, and discarded packets during delineation recovery.

Misconcatenation

In the absence of a mechanism to detect cell losses, the self-delineation scheme always misconcatenates following a cell loss. That is, in the event of a cell loss, the leading octets of the next cell are always misconcatenated with the trailing packet of the previous received cell. Only after this, when the position of the next expected packet header is examined and the header CRC fails, is the loss of packet delineation detected. Thus, the probability of misconcatenation of a packet split between two ATM cells is $1 \times p_c$.

Delineation Using the Start Field (STF)

For most telephony applications, the number of packets discarded per delineation loss event is excessive only if self-delineation is used. This is mainly due to the recovery period and misconcatenation. The STF, with the OSF, was designed to enhance recovery performance and, hence, reduce the packet discard ratio. The STF does consume an extra byte of overhead per cell. This overhead buys packet delineation recovery at the next cell.

Packet Discard Ratio for Delineation Recovery Using the STF

Assuming that an error could occur in the STF or in any of the k_p packet headers, the packet discard ratio due to bit errors is $p(k_p+1)(3k_p+2)=44p$ for $k_p=3$ packets per cell. When a delineation loss occurs due to a cell loss, resynchronization is immediate at the receipt of the next cell. Thus, no additional packets are discarded. The packet discard rate due to bit errors is comparable to the self-delineation scheme (i.e. 1.8×10^{-4} at $p=10^{-6}$). However, no additional packets are discarded following ATM cell loss/losses. Use of the STF provides improved quality of service by quicker resynchronization and the avoidance of the loss of a burst of packets.

Cell Sequence Number and Parity

Use of the LI from the packet header of the previous cell and the OSF from the current cell allows identification of a possible cell loss if the two LIs do not match. However, there is a significant probability that these two indicators will match. A better way of detecting cell loss is needed to reduce both the misconcatenation probability and the complexity of detection. Two fields are added to the OSF to create the STF. The 1-bit SN provides a modulo-2 SN and allows immediate detection of a single cell loss. In addition, a 1-bit P field allows detection of an odd number of errors in the STF itself.

CHAPTER 4

COMPARISON OF ATM AND TCP/IP NETWORK

The mapping between the TCP/IP layers and ATM network layers is shown in the figure.

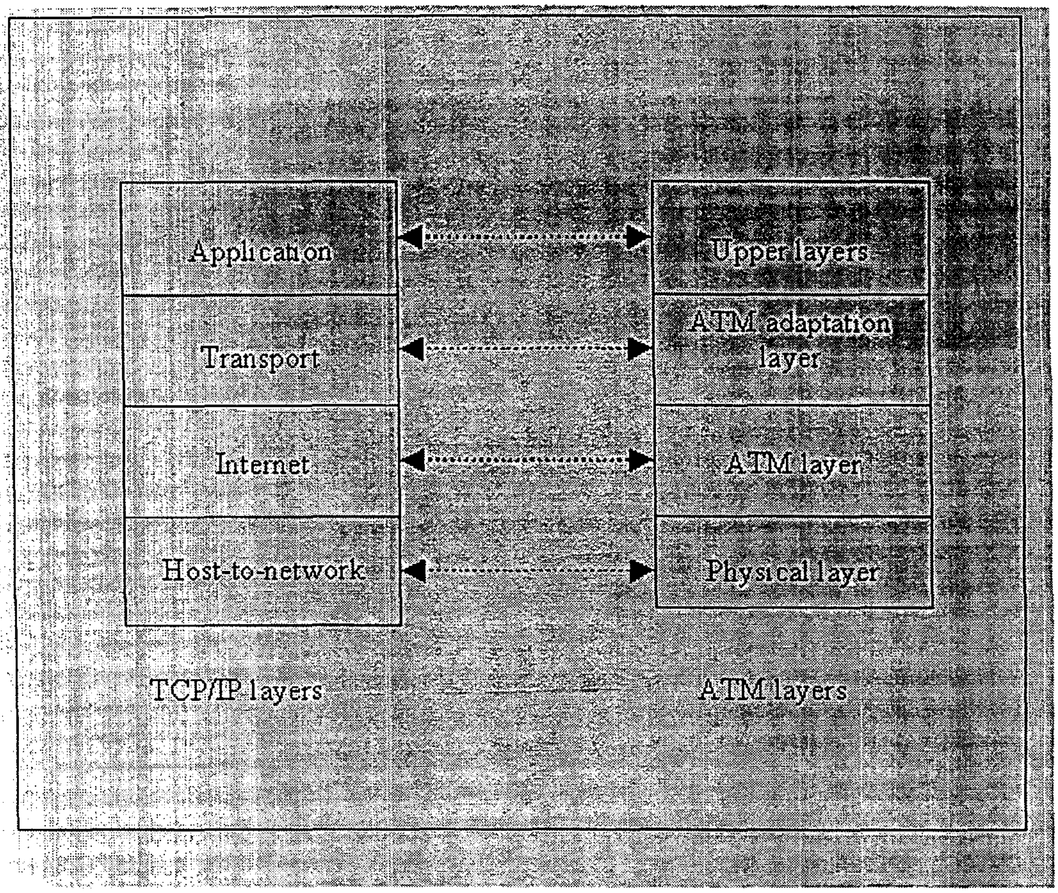

Figure 4-1. mapping between the TCP/IP layers and ATM layers

The comparison among the corresponding layers is as following:

4.1. ATM PHYSICAL LAYER VS TCP/IP PYSICAL LAYER

The basic rate for transmission in the ATM physical layer is standardized by the ITU - T recommendations. Whereas the transmission bit rate for TCP/IP is not standardized. The TCP/IP network is using old channel resources of 64kbps for dialed up connections and more for the other LANs etc.

4.2. ATM LAYER AND IP LAYER

The ATM layer and the IP layer can be compared on the following grounds

4.2.1. Cell Format

The ATM has the fixed sized cells of 53 octets whereas the IP has variable sized packets. In the ATM layer the two interfaces have been distinguished UNI and NNI.

4.2.2. Connection Setup

ATM supports permanent virtual circuit and the switched virtual circuit whereas the IP layer provides connectionless services and uses store and forward techniques. That is why the cell sequence is preserved in the ATM layer and the packet sequence is not preserved in the IP layer.

4.2.3. Service Categories

The service category in the ATM network depends upon the bit rate from source to destination whereas in IP layer there is no such barriers upon the bit rates for different types of services. In present days the main services provided by the TCP/IP network are data services.

4.2.4. Traffic Shaping and Policing

There is no policy for traffic shaping and policing in the TCP/IP model because the packet sequence is not preserved and the services provided by it are not real time services. In ATM networks the services provided are real time so traffic shaping and policing needed. The mechanism for using and enforcing the quality of service parameters is based on specific algorithm named as generic cell rate algorithm (GCRA).

4.3. TRANSPORT LAYER VS AAL LAYER

In the TCP/IP network has only one mode in the network layer (connection-less) but supports both modes (connection-less and connection-oriented) in the transport layer, giving the users a choice. The ATM network has both modes (connection-less and connection-oriented) in the ATM adaptation layer.

4.4. APPLICATION LAYER VS UPPER LAYERS

In TCP/IP network has only one protocol TCP for different type of application, because it supports basically non real-time application but in the ATM network different type of protocol (i.e. AAL-1, AAL-2 etc.) for different type of applications such as transmitting voice and video streams etc.

ATM network sends the transparent cells without checking the cells on every node. Cells are checked in AAL at source as well as at destination. But in the TCP/IP network, it sends the packets and this packets check at each node from source to destination.

CHAPTER 5

CONCLUSION

The study and analysis of ATM layers for BISDN represent crucial elements in the evolution toward a multiservice network. With the range of services requiring that network access grow rapidly and that the rate of growth accelerate, it is essential that designs process the required service flexibility to accommodate many undefined services. This dissertation has presented a viewpoint of ATM, where the key design guideline is service flexibility. Suggested approaches strive to create a service-independent broadband facility that is also sensitive to both service performance and network efficiency.

This dissertation discussed an ATM layer management entity and its functions. The ATM layer management entity conducts activities that control or monitor the use of the ATM cell transport resources. Transferred information can include the cumulative data of detected invalid and errored cells, With these activities and functions, effective management practices are achieved and the end-to-end performance is consequently improved.

AAL-2 achieves high bandwidth efficiency and low packetization delay simultaneously, thus making it ideal for voice and other low bit-rate interactive applications.

Further work on LLC servers is in progress. Specifically, the locations of LLC servers and routing of LLCs using one or more LLC servers as vias are being worked as a network design problem. Another are being investigated is related to engineering ATM VPC and/or VCC bandwidth given the call arrival patterns, blocking criteria, coding rates, degree of real-time adjustability in coding rates and routing strategies no[illegible] used. The basic principles of the AAL-2 are also application [illegible] non-ATM environments. Both the Frame Relay Forum and the IETF are investigating multiplexing protocols similar to the AAL-2 for carrying low-bit-rate voice over frame-relay and networks, respectively.

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Abbreviations

AAL	asynchronous transfer mode adaptation layer
AAL-IDU	AAL interface data unit
AAL-PCI	AAL protocol control information
AAL-SDU	AAL service data unit
ADSL	asymmetric digital subscriber line
ATD	asynchronous time division
ATM	asynchronous transfer mode
ATM-SDU	ATM(layer) service data unit
AUU	ATM-layer-user-to-ATM-layer-user parameter
BER	bit error rate
B-ISDN	Broadband Integrated Service Digital Network
C-4	SDH C-4
CAC	connection admission control
CAD	computer aided design
CBR	constant bit rate
CDMA	code division multiple access
CES	circuit emulation service
CID	connection identification
CLP	cell-loss probability
CLR	cell-loss rate or ratio(context distinguishes)
CPCS	common part convergence sublayer
CPCS-SDU	CPCS service data unit
CPS	common part sublayer
CRC	cyclic redundancy check
CS	convergence sublayer
CS-PDU	CS protocol data unit
DQDB	distributed-queue, dual-bus(MAN standard)
GFC	generic flow control (field in ATM header)
HEC	header error check

MAN	metropolitan area network
MSC	mobile switching center
MSS	man switching system
NNI	network-network interface
OA&M	operations, administration, and maintenance
OSF	offset field
PBX	private branch exchange
PCI	protocol control information
PCS	personal communications services
PDC	personal digital cellular
PDU	protocol data unit
PH	packet handler
PHY	physical layer
PM	physical medium sublayer
POTS	"plain old telephone service"
PPP	point-to-point protocol
PRI	primary rate interface
PSDN	packet switched data network
PSTN	public-switched telephone network
PT	payload type field(in ATM header)
PTI	payload type indicator
QOS	quality of service
RES	reserved
RPM	radio port multiplexer
RP	radio port
SAP	service access point

SAR	segmentation and reassembly (layer)
SAR-SDU	segmentation and reassembly (layer) service data unit
SAR-PDU	segmentation and reassembly (layer) protocol data unit
SMAAL	small packet multiplexed asynchronous transfer mode adaptation layer
SN	sequence number
SSCS	service-specific convergence sublayer
STF	start field
STM	synchronous transfer mode
TA	terminal adapter
TC	transmission convergence
TDM	time division multiplexing
UNI	user-network interface
UPC	usage parameter control
UU	user to user
VBR	variable bit rate (service)
VC	virtual channel
VCC	virtual channel connection
VCI	virtual circuit identifier
vocoder	voice coder
VP	virtual path
VPC	virtual path connection
VPI	virtual path identifier
VP	virtual path
WAN	wide area network