Application of Asynchronous Transfer Mode (Atm) technology to Picture Archiving and Communication Systems (Pacs): A survey

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by

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Specially designed telecommunication networks like public switched telephone network for speech, data networks for computer communications, and broadcast networks for television are very capable in supporting their intended services. These networks, however, are not well suited for other services like multimedia applications which involve the processing and exchange of information in the various media of text, audio, and images. Broadband Integrated Services Digital Network (B-ISDN) provides a range of narrowband and broadband services for voice, video, and multimedia. It provides virtual connections for services which are distributive or interactive, constant bit rate or variable bit rate, and connection oriented or connectionless. Asynchronous Transfer Mode (ATM) has been selected by the standards bodies as the transfer mode for implementing B-ISDN.

The ability to digitize images has lead to the prospect of reducing the physical space requirements, material costs, and manual labor of traditional film handling tasks in hospitals. The system which handles the acquisition, storage, and transmission of medical images is called a Picture Archiving and Communication System (PACS). A complete PACS system will require some means of transmission of the large amounts of digital data. The transmission system will directly impact the speed of image transfer. Today the most common transmission means used by acquisition and display station products is Ethernet. However, when considering network media, it is important to consider what the long term needs will be. Ethernet may be easily available, but sheer image data sizes will require most
sites to move to higher speed network media. Fiber Distributed Data Interface (FDDI) may seem to be a good choice but considering the small incremental increase, most sites will consider moving directly to a much faster medium. This will reduce the need to reinvest in network upgrades after only a few years of use. Although ATM is a new standard, it is showing signs of becoming the next logical step to meet the needs of high speed networks.

This thesis is a survey on ATM, and PACS. All the concepts involved in developing a PACS are presented in an orderly manner. It presents the recent developments in ATM, its applicability to PACS and the issues to be resolved for realising an ATM-based complete PACS. This work will be useful in providing the latest information, for any future research on ATM-based networks, and PACS.
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Chapter 1

Introduction

1.1 Introduction

A computer network is a structure which enables a data processing user at one location to make use of data processing functions or services performed at another location.

Communication networks are generally classified into three broad categories based on the distances they span [1][2][3]. Wide area networks (WANs) cover large geographic areas, metropolitan area networks (MANs) span a few miles, and local area networks (LANs) are generally limited to a single building or a group of buildings.

In a local area network, devices are usually hooked together via private, dedicated cables or other physical communication media. Typical LAN transmission media are twisted-wire-pair telephone wire, coaxial cable, fiber-optic cable, and various forms of wireless transmission. Wide Area Networks often use public, switched telecommunication facilities or packet-switching technologies for communication[4][5].

LANs and WANs also differ in the methods employed to control access to the transmission medium. Whereas LANs employ random medium access control methods like carrier sense multiple access with collision detection (CSMA/CD) and distributed medium access
control methods like token ring and token bus, the form of access control more commonly used in WAN data links is centralised medium access control like polling, circuit switching, packet switching and time division multiple access (TDMA) [6][7].

Conventional circuit switching is not well suited for high-speed networking, because no single user needs more than a small percentage of the total capacity of the transmission channel. Packet switching allows a number of users to share a high capacity transmission channel. Packet switching works well for computer data at low to moderate transmission speeds. But packet switching is not well suited for voice or video communications, because, the delays introduced by the packet switches are too long and too unpredictable for voice or video applications.

Specially designed telecommunication networks like public switched telephone network for speech, data networks for computer communications, and broadcast networks for television are very capable in supporting their intended services. These networks, however, are not well suited for other services like multimedia applications which involve the processing and exchange of information in the various media of text, audio, and images [8].

A new standard called Integrated Services Digital Network (ISDN) was developed to integrate a wide range of services that are inherently different in such network requirements as bandwidth, holding times, end-to-end delays, and error rates [9][10].

The current ISDN standards specify two types of user interfaces called basic rate access and primary rate access. Basic rate access comprises of two 64 kbps B channels and a 16 kbps D channel (2B+D). Primary rate access comprises of 23 B channels and a 64 kbps D channel (23B+D) [11][12].

The highest bit rate ISDN can offer is about 1.5 Mbps, this is not sufficient for
applications which require higher bit rates; like connection of LANs, or transmission of moving images. This led to the development of Broadband Integrated Services Digital Network (B-ISDN) [13][14][15][16]. B-ISDN includes 64 kbps ISDN capabilities and provides the broadband user with bit rates ranging from 50 mbps to hundreds of megabits per second.

Chapter 2 deals with the various types of computer networking technologies developed, the need for a common set of standards to govern the functioning of these networking technologies, the International Standards Organization’s (ISO’s) Open System Interconnection (OSI) model [17][18][19][20], and the ISDN and B-ISDN architectures [21].

B-ISDN provides a range of narrowband and broadband services for voice, video and multimedia. It provides virtual connections for services which are distributive or interactive, constant bit rate, or variable bit rate, and connection-oriented or connection-less [22][23].

A B-ISDN call may consist of a number of virtual connections that may be point-to-point or multipoint, symmetric or asymmetric, unidirectional or bidirectional, and switched or permanent.

With the increasing number of users per network, and increasing user-bandwidth needs, there is a need for developing faster networks. From 1991 to 1994, it was found that the average number of users per network segment has increased from 12 to 21, and the average bandwidth available to the users has decreased by more than 40% [24]. As available bandwidth is dropping, each user’s bandwidth demand is increasing. The growth in processing power at each node, combined with the corresponding growth in data transfer needs, boosts bandwidth demands of current applications. In addition, emerging data-intensive applications like video conferencing, medical imaging, etc., lead to increasing demand for larger network bandwidths [25][26]. Also, time critical applications like audio and video
which need to be transmitted within a set period further complicates network needs. If the network is crowded, conventional shared Ethernet cannot guarantee the timely network access such applications require. Section 2.5 of chapter 2 deals with some of the high speed networking technologies in use.

Asynchronous transfer mode (ATM) has been selected to be the transfer mode for implementing B-ISDN [27][28][29]. The term asynchronous does not refer to physical transmission (which is synchronous in B-ISDN), it refers to the manner in which bandwidth is allocated among connections and users.

In ATM, all information to be transferred is packed into fixed size slots called cells. These cells have a 48 octet information field and a 5 octet header. The information field contains user data and the header field carries information for the identification of cells. The use of short cells in ATM coupled with the high transfer rates involved result in transfer delay variations which are sufficiently small to enable it to be applied to a wide range of services, including real-time services such as voice and video [30][31].

One of the main advantages of ATM-based networks is that the multiplexing and switching of cells is independent of the actual application [32]. This enables the same piece of equipment to handle low bit rate connections as well as high bit rate connections.

The disadvantages of ATM are cell loss and variable cell delays in the network. Also considerable processing is involved in converting user information into and from the ATM cell format, and in carrying cells at high rates through each switch [33]. Chapter 3 is dedicated to the concepts of ATM.

The ability to digitize images has lead to the prospect of reducing the physical space requirements, material costs, and manual labor of traditional film handling tasks in

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hospitals. The system which handles the acquisition, storage, and transmission of medical images is called a Picture Archiving and Communication System (PACS) \[34][35][36\]. A full fledged PACS often includes a teleradiology sub-system and supports long-term electronic storage of diagnostic images \[37][38][39\]. It is not just limited to a specific department but is used to manage the flow of diagnostic images throughout a health care system.

A teleradiology system acquires radiographic images from one location and transmits them to one or more distant sites where they are displayed and converted to hardcopy film recordings. It consists of two or more sites connected by a wide area network (WAN) or a metropolitan area network (MAN) communication system \[40][41][42\].

A teleradiology system provides for primary interpretation of radiological images for patients in under served areas, integrates radiological services for multi-hospital health care provider consortiums, improves emergency service and intensive care unit coverage, offers consulting-at-a-distance with sub-speciality radiologists, and provides radiologists immediate access to large academic centers for help in the interpretation of difficult and problematic cases.

The main challenge for the developers of PACS and teleradiology systems is the integration of Hospital Information Systems (HIS), Radiology Information Systems (RIS), and Picture Archiving and Communication Systems (PACS) \[43][44][45\].

A high speed, high-performance network which links medical clinics, hospitals, pharmacies, labs, universities, and medical archive sites should be able to support various types of information like plain text, medical images, digitized sound files, image sequences, real time animation, medical simulation, and multimedia applications like videoconferencing \[46][47][48][49\].
A complete PACS system will require some means of transmission of the large amounts of digital data. The transmission system will directly impact the speed of image transfer [49][50][51]. Today the most common transmission means used by acquisition and display station products is Ethernet [52]. However, when considering network media, it is important to consider what the long term needs will be. Ethernet may be easily available but sheer image data sizes will require most sites to move to higher speed network media [53][54][55][56]. FDDI may seem to be a good choice but considering the small incremental increase, most sites will consider moving directly to a much faster medium. This will reduce the need to reinvest in network upgrades after only a few years of use. Although ATM is a new standard, it is showing signs of becoming the next logical step to meet the needs of high speed networks [57][58][59][60].

In chapter 4, the various components which constitute a PACS, and the issues of integration of PACS with other digital systems are described in section 4.1. To facilitate easy transmission of medical images in a hospital (and among hospitals) which is (are) usually multi-modality, and multi-vendor environment(s), several medical imaging standards were developed, the most popular standards are described in section 4.2. With hospitals having to handle very large amounts of data, efficient image management is very important. The various aspects involved in medical image management, like database design, prefetching, and image compression are discussed in section 4.3. The different classes of PACS and the design concepts involved in the development of a PACS are described in sections 4.4 and 4.5 respectively. Reliability is a very important issue in a medical environment, section 4.6 deals with this issue. Section 4.7 lists some of the existing PACS and describes in brief, a few of the proprietary networks, some of these systems have used.
Chapter 5 presents the conclusions of this thesis in the form of observations and recommendations. Migration to ATM is complicated by the limitations of existing PACS, section 5.2 describes the differences between the existing LAN technologies and ATM, and discusses the two standard techniques for providing LAN services over ATM. Section 5.3 describes the various issues to be resolved for rapid deployment of ATM into the medical environment.
Chapter 2

Computer Networks

The proliferation of computer networks has necessitated the development of a common set of standards to govern the way in which they functioned. The International Standards Organization's (ISO's) Open System Interconnection (OSI) model is described in detail in section 2.2. A generic model for interconnecting two nodes on different networks is described in section 2.3. Section 2.4 describes the salient features of some of the most popular networking technologies. The increasing demand for larger bandwidths has lead to the development of a broadband network. Section 2.5 describes the requirements; and section 2.6 describes the various transfer modes considered for the development of a broadband network. Integrated Services Digital Network (ISDN) was developed to provide a single network for all kinds of data, section 2.7 describes its design. For applications like medical imaging, interconnection of LANS, etc., the bandwidth provided by ISDN soon proved to be insufficient, this lead to the development of Broadband Integrated Services Digital Network (B-ISDN). Section 2.8 describes the features of B-ISDN.
2.1 Introduction

The telephone industry has influenced the development of the present telecommunication networks in a profound manner [4][5][61]. The T1 and T3 links provided by the telephone companies form the foundation for most of the voice networks. In fact B-ISDN is not a computer network, it is a world wide digital telephone network. Although computer communication is possible over this network, the real intent of B-ISDN is to build a digital telephone network for voice transmission, High Definition Television (HDTV), Video Conference, and other multimedia applications [14].

The progress in technology and system concepts has led to the mushrooming of various networking technologies, each one completely different from the others in the way data is handled and transmitted. This necessitated the development of a common set of standards to govern the way in which these networking technologies functioned [13][17]. The International Telecommunication Union (ITU) and ISO are two such standards bodies which develop standards to ensure compatibility between the various networking technologies [18][19].

2.2 The OSI Model

In the network environment, the network components communicate with each other through established conventions called protocols. Several protocols are necessary to support the various functions provided by the computer systems, leading to a great deal of complexity.

In order to deal with this complexity, computer networks are designed according to the concept of layered architectures[62]. In the layered architecture, all the functions associated with information transfer and exchange are partitioned and assigned to different
layers.

The ISO approved the Open Systems Interconnection (OSI) reference model as a standard in 1983. The OSI provides a common basis for the coordination of standards development for the purpose of systems interconnection, and is aimed at removing the technical barriers from communication between systems of different origin. The OSI model is organized into seven layers, each layer contains several protocols that are invoked based on the specific needs of the user. The layered architecture logically decomposes a complex network into smaller, more manageable layers, making the development and maintenance of networks a lot easier [1][2][3].

In the OSI model, each layer is divided into smaller operational entities[6][63], which are invoked by the end user to obtain the required services.

2.2.1 OSI Layers

In the OSI model, the functions that have to be carried out in order to establish and maintain information exchange between two or more systems are partitioned into seven layers as shown in figure 2.1. In the OSI model, the functions of computer communications and networking are subdivided into two functional groups; network-oriented layers and the application oriented layers. The functions of the network-oriented layers (layers 1 through 4) are: (a) to establish the connection between the two systems that wish to communicate through the underlying networks being used and (b) to control the transfer of messages across this connection without errors, losses or replication[63]. Collectively, they provide the higher, application oriented layers with a network-independent, reliable message interchange service.

The application-oriented layers (5 through 7) are concerned with the representation
File transfer, access and management, document and message interchange, job transfer and manipulation.

Syntax independent message interchange service.

Transfer syntax negotiation
Data representation transformations.

Dialogue and synchronisation control for application entities.

Network independent message interchange service.

End-to-end message transfer, (connection management, error control, fragmentation, flow control).

Network routing, addressing, call set-up and clearing.

Data link control (framing, data transparency, error control)

Mechanical and electrical network interface definitions

Physical connection to network termination equipment

Data communications network.

Application Layer

Presentation Layer

Session Layer

Transport Layer

Network Layer

Link Layer

Physical Layer

Figure 2.1: The OSI Model
of the data being exchanged. They provide user application processes with necessary support functions to perform a range of distributed processing tasks.

Each layer participates in data transfer by following its own protocol, communicating with only its peer layer in the corresponding system. The only direct link between two communicating systems is the actual physical network connection. Within each system, there is a well defined interface associated with each layer that allows to accept, process, and pass on data during a transfer. In this way, a layer communicates with its peer layer using the services provided by the layer immediately below. The message initiated at the application layer is passed down from layer to layer, each layer adding its own Protocol Control Information (PCI) if required, and behaving in accordance with its own protocol. At the physical layer, the fully prepared message is transmitted over the underlying networks to the receiving nodes. In the receiving node this procedure is reversed. On its upward path, each item of PCI is removed by the appropriate layer and the remaining message contents are passed up to the layers above.

2.2.2 Layer Functions and Standards

Network-Oriented Layers

- Physical Layer

This layer establishes the physical connection between the computer and network termination equipment and is specifically concerned with the electrical and mechanical properties of this connection. The physical layer standards include the type of signals used, the size of the plug or the socket, the number of pins and their meaning, and so on.
• Data Link Control Layer

This layer provides the means of transmitting data over the underlying physical connection. The functions of this layer include bit, byte, frame synchronisation, and error detection. The data link layer is sub divided into Medium Access Control (MAC) and Logical Link Control (LLC) sublayers. The MAC sublayer is responsible for functions such as collision detection or token passing, which allocate the transmission channel to different nodes. The LLC sublayer is responsible for error recovery and recovery of lost frames.

• Network Layer

The functions of this layer include establishing and clearing network connections and routing messages between two transport layer protocol entities. The network layer also enables two systems to interconnect across one or more subnetworks[Lin 92], thus providing a uniform end-to-end service. Two types of services can be supported; connection oriented in which a permanent connection, either physically or logically, is setup for the duration of the communication and connection less, in which a permanent connection is not required and a best try approach is adopted.

• Transport Layer

The transport layer provides the application-oriented layers above it with a reliable message transfer facility that is independent of the underlying network being used[65]. Its functions include flow control, error control, and message fragmentation[66]. Here again the connection oriented and connectionless operational modes are supported. In the connection oriented mode, the transport protocol provides full duplex transfer of information between two systems and enhances the quality of the underlying network
service to meet the application needs. In the connection less mode, the transport protocol offers a best try approach to transfer the offered data and is therefore intended for use with high quality networks. In order to internetwork with underlying networks of differing quality, the connection oriented standard defines five classes of protocols, class 0, class 1, class 2, class 3, and class 4, each intended for a different underlying network type[65].

Application-Oriented Layers

Session Layer
It provides the means whereby two cooperating application processes running on different network nodes can organize, synchronize, and regulate the orderly exchange of data. The session protocol consists of a set of functional units. The units required are negotiated when a session connection is set up.

Presentation Layer
This layer is concerned with the syntax of data that is to be exchanged between two communicating processes. When a connection is being established, the presentation protocol allows the two communicating processes to establish a common syntax for the data to be exchanged. The presentation layer has five facilities:

- Connection establishment facility
- Connection termination facility
- Context management facility
- Information transfer facility
• Dialog control facility

Application Layer

It forms the user interface to the protocol stack and the various protocol entities. The application layer provides the support for application programs to perform a range of distributed information processing functions in an open way. A number of different entities have been defined to serve different application requirements like file transfer, remote access, job transfer, e-mail, network management etc[67][68].

2.2.3 Entities and Services

In the OSI model, a layer is considered to be a service provider to the layer above it. The layer N at one end communicates with the layer N of another node using the services provided by the lower layer, the layer N-1.

The layer N obtains the services provided by layer N-1 by sending a primitive to layer N-1. Four primitives have been defined to establish a connection between, exchange information, and close the connections between two entities. They are request, indication, response and confirm. This interaction between the N and the N-1 layers is done via the service access points (SAPs) present between the N and N-1 layers[6].

The unit of data exchanged between two peer entities is called a Protocol Data Unit (PDU). The PDU consists of the Protocol Control Information (PCI) which is the information exchanged between the peer entities at different sites on the network and the Service Data Unit (SDU) which is the actual user data and the control information created at the upper layers. so the (N+1) PDU will become the (N) SDU, the (N) SDU plus the (N) PCI will form the (N) PDU and so on.
2.3 Internetworking of an OSI Network

Internetworking is basically connecting two nodes which belong to different networks. Figure 2.2 shows an OSI layered model of internetworking. The dashed arrows show the virtual peer-peer communication, and the solid arrows the actual communication path. An internetwork or 'internet', made in this fashion provides end-to-end connectivity at the transport/network layer boundary[69][70]. The transport layer connection is transparently constructed from one or more data link layer segments. The conceptual separation of layered functions enables each link layer segment to utilise different technologies, if desired, without affecting the semantics and functions offered by the network layer to the transport layer.

Routers and bridges are used for interconnecting two or more networks. The management of routers and bridges is an important part of an internetwork design. Routers have the responsibility of utilising the underlying link layer connections efficiently, and exchanging information with other routers regarding the destinations they can reach.
2.4 Some Existing Models and Protocols

Several networking technologies have been developed since the evolution of the first computer network, each differing in its use of physical media and access techniques. There are four characteristics that are important in describing a particular form of LAN. They are:

- **Transmission Medium**: The transmission medium is the cable or other physical circuit that is used to interconnect systems.

- **Transmission Technique**: The transmission technique refers to the type of signals that are exchanged over the physical transmission medium. The most common techniques used with LAN data links are called baseband and broadband transmission.

- **Network Topology**: The network topology identifies the logical shape that device interconnections take. Common LAN data link topologies are the bus, the ring, and the star.

- **Access Control Method**: The access control method describes the method by which communicating systems control their access to the transmission medium. Commonly used access control methods are contention, tokenpassing, and circuit switching.

2.4.1 Transmission Media

The four general types of transmission media that are used most often in constructing local area networks are twisted-wire pairs, coaxial cable, fiber optics links, and wireless transmission.

Twisted pair cabling is the cheapest and easiest to install. However it also poses the greatest technological challenge to designers. Obtaining speeds greater than several hundred
kbps requires specially constructed and terminated twisted pair installations. Commercial networks, such as LocalTalk (at 230 kbps) and 10baseT [71] (10Mbps Ethernet), have been offered over carefully designed twisted pair cables. Coaxial cable provides a more rigidly defined transmission medium, allowing signals of significantly higher bandwidth to be carried without distortion. It also provides higher levels of protection from external electromagnetic interference (EMI) corrupting data, or from LAN signals adding to external EMI. Ethernet, and the IEEE802.3 variant, are significant examples of LANs initially based on coaxial cabling. The drawback of coaxial cables is its higher costs of production and installation compared to twisted pair cables.

Optical fibres promise, and deliver, vastly more bandwidth than is achievable with coaxial cable. They offer immunity to EMI, and relatively low signal attenuation thus allowing very long distances to be covered without repeaters. The bandwidth bottle neck with current fibre links is the need to terminate and switch the data electronically. Ongoing research in photonic switching fabrics [14][72][73] and wavelength division multiplexing [74][75] are offering solutions to these problems.

Wireless transmission can be used in a number of different ways. It is often easier to use microwave or infrared transmission links where it is difficult to physically interconnect individual LAN cable segments. Wireless transmission has disadvantages as well. Interference is high and line-of-sight transmission is necessary to mention a few.

2.4.2 Transmission Techniques

There are two general techniques that can be used for transmitting signals over a physical communication medium, baseband and broadband. With baseband transmission, the entire bandwidth is used to transmit a single digital signal, and data is carried on the transmission
medium in its original form. With broadband transmission, analog techniques are used in which the available bandwidth may be sliced up into a number of channels. Data is superimposed upon a carrier signal that is modulated.

2.4.3 Network Topologies

There are three principal topologies that are employed by LAN data link technology these are star, bus, and ring [3].

In a star configuration, there is a central point to which a group of systems is directly connected. All transmissions from one system to another pass through the central point, which may consist of a device that plays a role in managing and controlling communication.

In a bus topology each system is directly attached to a common communication channel. As each message passes along the channel, each system receives it and examines the destination address contained in it. If the message is addressed to it, the system accepts it else the message is ignored.

In the ring topology, the cabling forms a loop, with a simple point-to-point connection attaching each system to the next around the ring. Each system acts as a repeater for all signals it receives and retransmits them to the next system in the ring at their original signal strength. The operation is similar to that of bus topology. The system that originates a message is generally responsible for determining that a message has made its way all the way around the ring and then not repeating the message, thus removing it from the ring.

2.4.4 Access Control Methods

An access control scheme ensures intelligent information exchange in a network without constant interference from transmissions from other nodes. The access schemes can be
broadly classified into three categories[7]. They are:

- Fixed Assignment Schemes
- Demand Assignment Schemes
- Random Assignment Schemes

In fixed assignment schemes, a portion of the communication channel is dedicated to each attached node. Examples of fixed assignment scheme include time division multiple access (TDMA), and frequency division multiple access (FDMA). But these schemes are not very efficient when the traffic is bursty. Lack of flexibility in allocating bandwidth to nodes and inefficient handling of bursty traffic are the major deficiencies of these access schemes for bursty traffic.

In demand assignment schemes, a station will have access to the entire bandwidth of the network when it accesses the network; that is it can transmit at the full data rate of the channel. The medium access control scheme ensures that each node has a fair opportunity of accessing the medium and that no bandwidth is allocated to nodes that do not have information to transmit. Examples of this scheme include polling and token passing[76]. In token passing, a token controls the right to access the communication channel. When a station receives the token, it is allowed to transmit certain number of packets before it transmits the token to the next node. If a node does not have any packets to send, it passes the token to the next node immediately. This way, the bandwidth of the channel is used by all the active stations of the network, and none of the bandwidth is dedicated to stations that do not demand it.
In random assignment schemes, the entire bandwidth of the channel is used by any of the stations at times determined by the stations themselves. Conflicts are resolved by rescheduling transmissions at random times. This technique is used in ETHERNET and IEEE 802.3 LANs.

Some of the most commonly used LAN data link technologies are Ethernet (IEEE802.3), ARCnet, LocalTalk.

Ethernet or IEEE802.3 is a LAN data link technology in which systems are attached to a common transmission facility, such as coaxial cable or twisted-pair cable. A system typically attempts to transmit whenever it has data to send. Ethernet is the most widely used form of LAN data link technology.

ARCnet is a relatively low-speed form of LAN in which all systems are attached to a common coaxial cable. Like the Token Bus form of LAN, a system transmits when it has a token.

And LocalTalk is a low-speed LAN data link technology which was developed by Apple computer. The systems are attached to a common cable. A given LocalTalk cable can support a maximum of 32 stations, spanning a maximum distance of 300 meters. It uses a Carrier Sense Multiple Access with Collision Avoidance (CSMA/CA), to ensure all nodes share the bus fairly [19].

All these are low speed networking technologies (in the less than 10Mbps range). Some high speed technologies have been developed and are also in use. Fibre Distributed Data Interface (FDDI), IEEE802.6 (DQDB) MAN, and 100 Mbps Ethernet Standards (100 Base-X and 100 Base-VG) are some of them.
In the 100 Base-X [77], the ethernet's present media-access-control (MAC) layer protocol that is the carrier sense multiple access with collision detection (CSMA/CD) has been retained in order to reduce the cost and risk of upgrading an existing system to fast Ethernet. In the CSMA/CD protocol, the nodes desiring network access first listen to the network to determine if it's in use. If the network is free, the node will begin to transmit but continues to listen to detect any collisions (simultaneous transmission by two or more nodes). If a collision occurs, the CSMA/CD protocol will ask the contending nodes to cease transmission and try again following a randomly timed delay. If several collisions occur in succession, the nodes double their delay interval and keep trying.

Although the CSMA/CD protocol eventually allows one node to gain access, the delay between access request and acquisition is both variable and unbounded. Also, a heavily loaded network spends considerable time resolving collisions, which is known as thrashing. Thrashing causes available network bandwidth to drop well below 10-Mbps this may lead to as much as 40% loss of system bandwidth in some cases.

The 100 Base-VG which is developed by Hewlett-Packard, IBM, and AT&T, seeks to avoid thrashing and to offer guaranteed network access by operating with a demand-priority protocol at the MAC layer. This alternative Fast Ethernet can also handle token-ring frames, thus providing an upgrade path for both Ethernet and token-ring users.

In the demand-priority protocol, the network hub arbitrates network access requests based on round-robin polling. This results in full utilization of bandwidth. The hub polls for port access requests creates a requester list, then grants access to ports in the list order. Each requesting port gets a turn before the poll is taken again.
The demand-priority protocol also provides high priority access, allowing a port to receive access ahead of its normal turn. Round-robin polling also determines high-priority request handling and requires that all high priority users be serviced before normal priority users gain access. Normal priority users cannot be locked out however. If a normal priority access request has been pending for more than 250 msec, the hub automatically upgrades that request's priority status so that it is serviced in the high-priority sequence.

Fiber Distributed Data Interface (FDDI) [77][78][79] is a set of standards developed by the ANSI X3T9.5 Task Group. Ethernet and Token-Ring support only asynchronous traffic. FDDI adds synchronous service. Synchronous traffic consists of delay sensitive traffic such as voice packets, which need to be transmitted within a certain time interval. Asynchronous traffic consists of data packets produced by various computer communication applications, such as file transfer and mail. These packets can sustain reasonable delay, but are generally throughput sensitive in that higher throughput is more important than the time for bits to travel over the network.

An important feature of FDDI is its distributed nature, all algorithms are distributed that is the control of rings is not centralized. When any component fails, other components can reorganize and continue to function.

Regarding higher layer protocols, FDDI is compatible with CSMA/CD, token rings, and token bus. Applications running over these LANs can easily work over FDDI without any significant changes to upper layer software.

FDDI-II provides support for isochronous service in addition to the asynchronous and synchronous service provided by FDDI (basic mode). If an application needs guaranteed transmission of \( n \) bytes every \( T \) micro seconds, or some integral multiples of \( T \) micro seconds,
the application is said to require isochronous service[79].

Like FDDI, FDDI-II runs at 100 Mb/s. FDDI-II nodes can run in the FDDI mode. If all stations on the ring are FDDI-II nodes then the ring can switch to the hybrid mode, which provides isochronous service in addition to basic mode services; but if even one node is not an FDDI-II node then the ring can not switch to the hybrid mode and continues in the basic mode.

To provide periodic isochronous services, FDDI-II uses a periodic transmission policy with transmission opportunities repeated every 125 micro seconds. This interval matches the basic system reference frequency clock used in public tele-communications networks in North America and Europe. At this interval, a special frame called a “cycle” is generated. At 100 Mb/s, 1562.5 bytes can be transmitted in 125 micro seconds. Of these, 1560 bytes are used for the cycle and 2.5 bytes are used as the intercycle preamble[79].

The 1560 bytes of the cycle are divided into 16 Wide Band Channels (WBCs) of 96 bytes each. Each WBC provides a bandwidth of 96 bytes per 125 micro seconds or 6.144 Mb/s, sufficient to support one television broadcast, four high-quality stereo programs, or 96 telephone conversations.

The bytes of the cycles are preallocated to various channels for communication between two or more stations on the rings. Some of the 16 WBCs may be allocated for packet mode transmission and the others for isochronous mode transmission. Allocation is made using station management protocols.

Distributed Queue Dual Bus (DQDB, IEEE802.6) [80][81][82] was designed as a MAN technology that can support asynchronous and isochronous services - allowing bursty traffic to share the medium with traffic requiring guaranteed bandwidth. It is based on
two unidirectional fibre buses with a cell based structure. Cells are 53 octets of which 5 octets are for the header and 48 octets for the payload. The buses may be implemented at a variety of bit rates, utilising whatever fibre technology is available. Each bus carries frames every 125 microseconds, with each frame being subdivided into as many cell slots as the medium bitrate and physical layer overhead can support. The MAC protocol ensures that guaranteed number of cell slots are available to carry services requiring fixed bandwidth connections, and that the remaining slots are shared equally between attached nodes. It has been chosen to be the access network technology for the Switched Multimegabit Digital Service (SMDS).

HIPPI (High Performance Parallel Interface) [83][84] is a simplex, packet oriented, point-to-point interface for transferring data at peak rates of 800 Mbps or 1600 Mbps over distances upto 25 meters. A logical full duplex circuit is accomplished by using two HIPPI interfaces.

The HIPPI standard is divided into six sections:

- Physical interface (HIPPI-PH)
- Switch control (HIPPI-SC)
- Framing protocol (HIPPI-FP)
- Link encapsulation (HIPPI-LE)
- Memory interface (HIPPI-MI)
- Intelligent peripheral interface (HIPPI-IPI)

HIPPI-PH standard defines the physical layer. The specifications include use of one or two copper twisted-pair 100 pin cables for point to point connection over a distance of
25 meters. Data transfers are performed and flow controlled in increments of bursts, each burst has up to 256 words.

HIPPI-SC standard defines the control for the HIPPI physical layer switches.

HIPPI-FP standard defines data framing. Large block data transfers, with framing to split the data into smaller bursts of 256 words of 32 or 64 bits each are allowed. HIPPI-FP provides a connectionless data service and best effort delivery of data.

HIPPI-LE standard defines the protocol data unit (PDU) format and interface for transporting ISO 8802-2 (IEEE 802.2) logical link control (LLC) PDUs over HIPPI.

HIPPI-MI standard defines a protocol and formats by which an external device gains access to a memory sub-system of another device.

HIPPI-IPI standard supports information transfer between one or more hosts and peripheral systems.

HIPPI is a data channel, it is used to link super computers and work stations on giga-bit-per-second back planes. HIPPI-SC switches can connect a number of computer systems as a LAN.

HIPPI is very popular as a network interface because of its speed and connection oriented hardware protocol. It uses crossbar switches for interconnection. A HIPPI crossbar switch is an interconnection matrix with HIPPI interfaces. Each simplex HIPPI coming into the switch can be connected to one going out. The connections are made electrically and follow the HIPPI-SC protocol.

HIPPI-SC describes two modes of switch addressing. One is called "Source Routing" mode in which a 24-bit string of port numbers is provided to give the explicit route through a series of connected switches. The other mode, called "Logical Address" mode, requires
that each switch have look-up tables to map from 12-bit Logical Addresses to physical ports. This mode is preferred for networking.

2.5 Performance Requirements of a Broadband Network

The broadband network is designed to support a wide variety of services. These services are low speed, medium speed and very high speed. Low speed services include telemetry, telecontrol, telealarm, voice, telefax, low speed data transfer. Medium speed services include hifi fidelity sound, video telephony, high speed data transfer. Very high speed services include high quality video distribution, video library, video education, teleradiology and so on. A transfer mode for such a network must be very flexible in the sense that it must transport a wide range of natural bit rates [13][85] and it should be able to cope with services which have a fluctuating natural bit rate with respect to time.

A transfer mode is basically characterized by the switching technique used in the switching nodes of the network. The reliability of a network is measured in terms of semantic transparency and time transparency [86].

2.5.1 Semantic Transparency

Semantic transparency determines the capability of the network to transport information from the source to the destination with a limited number of errors. The types of errors introduced by the network may differ from one transfer mode to another. In a network, errors can be classified as transmission errors and switching/multiplexing errors.

Transmission errors occur due to imperfections in the transmission medium, since the transmission medium is only concerned with bits, transmission errors are only bit errors.
Switching and multiplexing operations occur at higher layers in the OSI model and the processing is done at the packet level. The switching/multiplexing errors comprise of both bit errors and packet errors.

- **Bit Error Rate**

  The bit error rate (BER) is defined as the number of bits which arrive erroneously divided by the total number of bits transmitted.

  \[
  BER = \frac{\text{Number of erroneous bits}}{\text{Total Number of bits sent}}
  \]  
  
  (2.1)

  Erroneous bit is a bit which arrives at the destination with a value other than that transmitted. The bit errors can occur either as isolated errors (singular errors) or group errors (burst errors). Isolated errors are mainly caused by noise and system imperfections and burst errors are caused by packet errors, maintenance actions and impulse noise.

- **Packet Error Rate**

  In packet oriented networks, bits are grouped in packets and the packet error rate (PER) is defined as the number of erroneous packets over the total number packets transmitted.

  \[
  \text{PER} = \frac{\text{Number of erroneous packets}}{\text{Total number of packets sent}}
  \]  
  
  (2.2)

  Packet errors are mainly caused by errors of the header. An incorrect value in the
header will either cause the switching system to discard the packet or to misinterpret
the header. The misinterpretation will cause a misroutting of the packet and this
results in a missing packet at the correct destination and an additional packet at an
incorrect destination. Taking into account the lost packets and inserted packets, PLR
(packet loss rate) and PIR (packet insertion rate) can be defined. PLR is defined as
the ratio of the number of lost packets to the total number of packets sent.

\[
PLR = \frac{Number\ of\ lost\ packets}{Total\ number\ of\ packets\ sent}\quad (2.3)
\]

PIR is defined as the ratio of the number of inserted packets to the total number of
packets sent.

\[
PIR = \frac{Number\ of\ inserted\ packets}{Total\ number\ of\ packets\ sent}\quad (2.4)
\]

The performance of a network can be enhanced by employing Forward Error Correc-
tion (FEC)[87] techniques or by retransmission if an error is detected using the Automatic
Repeat Request (ARQ) protocols. FEC techniques use complex coding schemes like Ham-
ing, Golay, BCH (Bose-Chaudhri-Hocquenghem) codes to add redundancy on the bit level.
ARQ techniques rely on the retransmission of information which was not received correctly.
The coding techniques mentioned above can be employed for the detection and correction
of the errors or just for the detection of errors.

The FEC and ARQ techniques should be judicially employed because the complex
error correcting codes lead to long processing delays, and frequent retransmission of infor-
mation will lead to increased traffic and congestion in the network[86], thus slowing down
the network.

2.5.2 Time Transparency

Time transparency determines the capability of the network to transport the information through the network from source to destination in a minimal amount of time.

Time transparency is dependent on two parameters; delay and delay jitter. Delay is defined as the time difference between the sending of the information at the source and the receiving of this information at the receiver. The delay $D$ can be different for every information block, and can have a minimum value of $D_m$ and a maximum value of $D_M$. The difference $D_M - D_m$ is called delay jitter.

The delay in a network is caused because of the time taken by the information to travel from the source to the destination, this is known as transfer delay ($D_t$). Delay can also occur because of the time taken for processing of the signalling and user information in the various network components, this is called processing delay ($D_p$). The total delay ($D$) is the sum of $D_t$ and $D_p$.

$$D = D_t + D_p$$

(2.5)

The time transparency of a network can be enhanced by improving the quality of the physical medium and reducing the processing overhead to a minimum.

2.6 Transfer Modes

The various transfer (figure 2.3) modes considered for the development of a single universal network are:
• Circuit Switching

• Multirate Circuit Switching

• Fast Circuit Switching

• Packet Switching

• Fast Packet Switching

2.6.1 Circuit Switching

In this transfer mode, a circuit is established for the complete duration of the connection. This is based on the TDM (time division multiplexing) principle to transport the information from one node to the other. This technique is also known as STM (synchronous transfer mode).

The information is transferred with a certain repetition frequency, the basic unit of this repetition frequency is called a time slot. Several connections are time multiplexed over one link by joining together several time slots in a frame, which is again repeated with a certain frequency. A connection will always use the same time slot in the frame during the complete duration of the session. Switching of circuits is controlled by a translation table which contains the relation of the incoming link and the slot number, to the outgoing link and the associated slot number.

Circuit switching is very inflexible, because once the duration of the time slot is fixed, the related bit rate is also fixed thus making it unsuitable for supporting a variety of services with varying bandwidth requirements. Also the time slots reserved for a particular circuit are unavailable for use by others even when no useful data is being exchanged. Hence
Transfer Modes

- Circuit Switching
  - Multimedia Circuit Switching
  - Fast Circuit Switching
- Packet Switching
  - Frame Relaying
  - Frame Switching
  - Fast Packet Switching
    - ATM
circuit switching has been considered unsuitable for a universal broadband network.

2.6.2 Multirate Circuit Switching:

To overcome the inflexibility of only a single bit rate, offered by circuit switching, a more enhanced version was developed called multirate circuit switching (MRCS) [86].

In this mode, like in simple circuit switching, time division multiplexing with a fixed basic channel rate is employed. But every connection can be built as a multiple of the basic channel rate hence one connection can allocate 'n' basic channels.

The main problem with this mode was the selection of a basic rate. If the basic rate is selected to be the minimal required channel rate, then a very large number of basic channels would be required for supporting high bit rate services like HDTV or medical image transmission. This will make the management and correlation of all these channels very complicated. On the other hand if the basic bit rate is selected to be large, then the waste of the bandwidth becomes too large for low bit rate applications like voice and low speed data communications.

To overcome this problem of selecting a basic rate, another type of circuit switching called Multirate Circuit Switching with different basic channel rates was developed. Here channels which are multiples of the basic rate are employed.

Both the above mentioned modes suffer from their inability to cope efficiently with sources generating data of fluctuating and bursty nature. The network resources are occupied even when the sending terminal is idle. Hence multirate circuit switching was considered unsuitable for a universal broadband network.
2.6.3 Fast Circuit Switching:

In order to extend the concepts of circuit switching to sources with fluctuating and bursty nature, fast circuit switching (FCS) was developed [86]. The resources in the FCS network are only allocated when information is sent, and released when no information is sent.

At call set-up, users request a connection with a bandwidth equal to some integer multiple of the basic rate. The system will not allocate the resources immediately, it will store in the switch, the information on the required bandwidth and the selected destination, and allocates a header in the signalling channel, identifying that connection. When the source starts sending information, the header will indicate that the source has information requiring from the switch to allocate the necessary resources immediately.

In order to build a system which can support different information rates, FCS and MRCS were combined, but the complexity of designing and controlling such a system lead to its rejection by the ITU-T as a potential solution for a universal broadband network.

2.6.4 Packet Switching:

In packet switching networks, user information is encapsulated in packets which contain additional information in the header for routing, error correction, flow control etc in the network.

The packet switching networks like X.25 [88] were developed in the sixties when the quality of of the transmission links was poor. To offer an acceptable end-to-end performance, complex protocols for performing error and flow control were employed on each link. Also the packets were of variable length and this necessitated a complex buffer management [89] inside the network leading to a large delay inside the network.
The large delays inside the network because of complex protocols makes it very difficult to apply packet switching technique for real time services and high speed services.

2.6.5 Frame Relaying and Frame Switching:

These techniques evolved from packet switching[89], they have reduced functionalities than X.25 and hence higher throughput can be achieved. Frame switching can support a bit rate of about 4 to 8 Mbps and frame relaying can support a bit rate of about 140 Mbps.

2.6.6 Fast Packet Switching:

Fast packet switching [86] is nothing but packet switching with minimal functionalities in the network. This coupled with the improved quality of the physical transmission medium will allow the systems to operate at a much higher rate than usual packet switching. The development of fast packet switching led to the development of Asynchronous Time Division Multiplexing (ATD) [90] and this evolved into Asynchronous Transfer Mode (ATM). These schemes allow an asynchronous operation between the sender clock and the receiver clock.

The ATM network is capable of supporting any service irrespective of its characteristics such as the bit rate, its quality requirements or its bursty nature. A network based on such a service independent transfer technique is flexible and future safe in that any changes in the characteristics of the services can be supported without modifying the ATM network. The ATM network is efficient in the use of its available resources. Every available resource can be used by any service. Since only one network needs to be designed, controlled, manufactured and maintained, the overall costs of the system will be smaller [31]. These advantages of the ATM network lead to its approval by the ITU-T as the transfer mode for a universal broadband network.

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2.6.7 Performance of ATM:

The information field length of an ATM cell is fixed and relatively small (48 octets) so delay and delay jitter are small enough to support real time services. Also since the ATM header has limited functions, processing in the ATM nodes is simple and can be done at very high speeds (150 Mbps is easily attainable) [27].

ATM operates in a connection oriented mode, the various parameters like bandwidth, quality of service (QOS) are negotiated before the resources are allocated to the user for information transfer over the network. This guarantees a very low (between $10^{-8}$ and $10^{-12}$) packet loss ratio (PLR) thus unlike in other packet oriented transfer modes like X.25 making flow control on a link by link basis unnecessary. Also because of the high quality of the physical transmission medium, error protection on a link by link basis can be omitted [30][86].

The ATM transfer mode can provide very high transfer rates depending on the type of physical layer available, transfer rates as high as 600 Mbps are easily attainable[29].

2.7 Integrated Services Digital Network

Integrated Services Digital Network (ISDN) is basically a digital user interface to the digital telephone system built during the late 1970's and 1980's.

The end user device connects to an ISDN node through a User-Network Interface (UNI) protocol. The typical ISDN topology is as shown in figure 2.4 [91].

The ISDN model is based on functional groupings and reference points. TE1 and TE2 are end user terminals, the terminal adapter (TA) is a device that allows non ISDN terminals to operate over ISDN lines. The network terminator (NT1) is device which
connects the 4-wire subscriber wiring to the 2-wire local loop; it is responsible for the physical layer functions such as signaling synchronization and timing, and the network terminator (NT2) contains the layers 2 and 3 protocol functions of the OSI model [92].

The TE1 connects to the ISDN through a twisted pair 4-wire digital link. This link is time division multiplexed to provide three channels B, B, and D (2B+D). The B channels operate at a speed of 64 kbps and the D channel at 16 kbps. The 2B+D is called Basic Rate Interface (BRI). The B channels are used to carry the user payload (voice, video, data). The D channel is used for out of band signalling.

ISDN also supports another type of interface called the Primary Rate Interface (PRI). It is obtained by multiplexing multiple B and D channels onto a higher speed interface. The PRI can provide a bandwidth of 1.544 Mbps (used in North America and Japan) by multiplexing 23 B channels and a D channel and is called as 23B+D or H1 channel.
2.7.1 ISDN Layers:

ISDN covers the bottom three layers (Physical, Data Link, Network) of the OSI model as shown in figure 2.5 [91].

![Figure 2.5: The ISDN Layers](image)

Layer 1 (the physical layer) uses either the basic rate interface (BRI) or primary rate interface (PRI). Layer 2 (the data link layer) contains the Link Access procedure for D-channel (LAPD) protocol, and the Layer 3 (the network layer) messages are used to manage ISDN connections on the B-channels.

The ISDN layer 3 messages (Q.931 messages) are used to manage ISDN connections on the B channels. LAPD ensures that the Q.931 messages are transmitted across the link. The LAPD frame format is as shown in figure 2.6[REF]. The address field contains several control bits, a Service Access Point Identifier (SAPI), and a Terminal End Point Identifier (TEI). The SAPI and TEI fields are known collectively as the Data Link Control Identifier (DLCI) [92]

The SAPI identifies the entity where the data link layer services are provided to the layer above. A SAPI value of ‘0’ indicates that the frame is carrying signalling information.
Figure 2.6: LAPD Frame

<table>
<thead>
<tr>
<th>Flag</th>
<th>Address</th>
<th>Control</th>
<th>Information (ISDN layer 3)</th>
<th>FCS</th>
<th>Flag</th>
</tr>
</thead>
</table>

- **Link Control field**
  - (sequence numbers, ACKs, NAKs, flow control, link addresses, SAPs in the terminal)

- **Q.931 messages for the establishment and disestablishment of B channels**
a SAPI value of 16 indicates that it is carrying user traffic and a SAPI value of 63 indicates that management information is being carried in the frame.

The TEI identifies either a single terminal, or multiple terminals that are operating on the BRI link. The TEI is assigned automatically by a separate assignment procedure. A TEI value of all 1s identifies a broadcast connection.

![Figure 2.7: The Q.931 Message](image)

The Q.931 message format is as shown in figure 2.7. The protocol discriminator distinguishes between user-network call control messages other layer 3 protocol messages. The call reference identifies the specific ISDN call at the local UNI. The Message Type identifier identifies the message function. The contents of the Other Information Elements
field depend on the message type. Some of the Q.931 messages are shown in table 1 [91].

**ISDN Layer 3 Messages** The shortcomings of ISDN are its limited bandwidth, lack of

<table>
<thead>
<tr>
<th>Call Establishment Messages</th>
<th>Call Disestablishment Messages</th>
</tr>
</thead>
<tbody>
<tr>
<td>ALERTING</td>
<td>DETACH</td>
</tr>
<tr>
<td>CALL PROCEEDING</td>
<td>DETACH ACKNOWLEDGE</td>
</tr>
<tr>
<td>CONNECT</td>
<td>DISCONNECT</td>
</tr>
<tr>
<td>CONNECT ACKNOWLEDGE</td>
<td>RELEASE</td>
</tr>
<tr>
<td>SETUP</td>
<td>RELEASE COMPLETE</td>
</tr>
<tr>
<td>SETUP ACKNOWLEDGE</td>
<td></td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Call Information Phase Messages</th>
<th>Miscellaneous Messages</th>
</tr>
</thead>
<tbody>
<tr>
<td>RESUME</td>
<td>CANCEL</td>
</tr>
<tr>
<td>RESUME ACKNOWLEDGE</td>
<td>CANCEL ACKNOWLEDGE</td>
</tr>
<tr>
<td>RESUME REJECT</td>
<td>CANCEL REJECT</td>
</tr>
<tr>
<td>SUSPEND</td>
<td>CONGESTION CONTROL</td>
</tr>
<tr>
<td>SUSPEND ACKNOWLEDGE</td>
<td>FACILITY</td>
</tr>
<tr>
<td>SUSPEND REJECT</td>
<td>FACILITY ACKNOWLEDGE</td>
</tr>
<tr>
<td>USER INFORMATION</td>
<td>INFORMATION</td>
</tr>
<tr>
<td></td>
<td>REGISTER</td>
</tr>
<tr>
<td></td>
<td>REGISTER ACKNOWLEDGE</td>
</tr>
<tr>
<td></td>
<td>REGISTER REJECT</td>
</tr>
<tr>
<td></td>
<td>STATUS</td>
</tr>
<tr>
<td></td>
<td>STATUS ENQUIRY</td>
</tr>
</tbody>
</table>

Table 2.1: ISDN Layer 3 Messages

flow control in that ISDN preallocates bandwidth to users, this is not a good way of handling bursty traffic which computers generate, and there is no guarantee that a packet pushed into the network will make it to the destination. These shortcomings and the development of ATM, SDH/SONET and fiber optic technology are responsible for its failure.

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2.7.2 The need for B-ISDN

The highest bit-rate of 1.544 Mbps offered by ISDN is insufficient for the connection of local area networks, or transmission of moving images with good resolution. This demand for higher speed channels led to the development of Broadband ISDN (B-ISDN).

B-ISDN is designed to support different kinds of applications and customer categories. It will support services with both constant and variable bit rates, data, voice, still and moving picture transmission and multimedia services which combine data, voice and picture service components.

B-ISDN is designed to become a universal network, and Asynchronous Transfer Mode (ATM) is the transfer mode for implementing it [14][13].

2.8 B-ISDN Model:

The reference configuration and functional groupings of B-ISDN shown in figure 2.8 [91] are similar to those of NISDN. The reference points are R, S_B, T_B, U_B, these are as defined for NISDN. The functional groupings are labeled as B-NT1, B-NT2 (Broadband Network Termination 1 or 2), B-TE1, B-TE2 (Broadband Terminal Equipment 1 or 2) and B-TA (Broadband Terminal Adapter).

Reference points and functional groupings are just abstract concepts which are useful for understanding the B-ISDN architecture. The similarities between ISDN and B-ISDN are in concept only; it is not possible to upgrade an ISDN interface by simply supplementing it with B-ISDN functional groupings and reference points.
Figure 2.8: B-ISDN Reference Configuration
2.8.1 B-ISDN Reference Model:

The B-ISDN reference model is based on the OSI reference model. The functions of the layers and the relations of the layers with respect to each other are described in a protocol reference model (PRM) as shown in figure 2.9 [21][86].

![Diagram of the BISDN ATM Protocol Reference Model](image)

Figure 2.9: The BISDN ATM Protocol Reference Model

The B-ISDN model contains three planes. The user plane (U-Plane) is responsible for providing user information transfer, flow control, and recovery operations. The control plane (C-Plane) is responsible for setting up, releasing and managing network connections. The management plane (M-Plane) is responsible for coordination of all the planes (plane management) and for managing the entities in the layers, performing operation, administration and management services (layer management) [21].

For each plane a layered approach as in OSI is used with independence between layers. The physical layer is similar to the layer 1 of the OSI model, and mainly performs functions at the bit level. The ATM layer performs operations typically found in layers 2
and 3 of the OSI model. The ATM Adaptation layer (AAL) combines the features of layers 4, 5, and 7 of the OSI model. The B-ISDN layer functions are shown in figure 2.10, and are described in detail in the next chapter.

2.8.2 B-ISDN Services

The ITU-T has classified the services offered by B-ISDN into interactive services and distribution services. Interactive services comprise of conversational, messaging and retrieval services. Distribution services are further classified as distribution services without user-individual presentation control and distribution services with user-individual presentation control [86][91].

Conversational services are interactive dialogues with real-time operations. There is no store and forward operations occurring between the service user and service provider. Examples of such services are: Broadband videotelephony, videoconference, video surveillance, high-speed unrestricted digital information transmission (for LAN, MAN interconnection), high volume file transfer, high resolution image communication (for professional images, medical images).

Messaging services include user to user communications, such as video mail service (Electronic mailbox service for the transfer of moving pictures and sound) or document mail service (Electronic mailbox service for documents containing text, graphics, still and moving picture information and voice), which can be done on a conversational basis, or on demand.

Retrieval services fall into the store and forward category where a user can obtain information stored for public use. This information can be retrieved on an individual basis from the service provider to the service user. Video retrieval (for entertainment and

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<table>
<thead>
<tr>
<th>Layer functions</th>
<th>Names of layers</th>
</tr>
</thead>
<tbody>
<tr>
<td>Convergence</td>
<td>CS</td>
</tr>
<tr>
<td>Segmentation &amp; reassembly</td>
<td>AAL</td>
</tr>
<tr>
<td>Generic flow control</td>
<td></td>
</tr>
<tr>
<td>Cell header processing</td>
<td>SAR</td>
</tr>
<tr>
<td>VPI/VCI processing</td>
<td></td>
</tr>
<tr>
<td>Cell muxing &amp; demuxing</td>
<td>ATM</td>
</tr>
<tr>
<td>Cell rate decoupling</td>
<td></td>
</tr>
<tr>
<td>HEC header processing</td>
<td></td>
</tr>
<tr>
<td>Cell delineation</td>
<td></td>
</tr>
<tr>
<td>Transmission frame adaptation</td>
<td>TC</td>
</tr>
<tr>
<td>Transmission frame generation/recovery</td>
<td>PL</td>
</tr>
<tr>
<td>Bit timing</td>
<td></td>
</tr>
<tr>
<td>Physical medium</td>
<td>PM</td>
</tr>
</tbody>
</table>

CS        Convergence sublayer  
SAR       Segmentation and reassembly sublayer  
AAL       ATM adaption layer  
ATM       Asynchronous transfer mode  
TC        Transmission convergence sublayer  
PM        Physical medium sublayer  
PL        Physical layer

Figure 2.10: B-ISDN Layer Functions

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remote education), high-resolution image retrieval (for entertainment, remote education, professional image communication and medical image communications), document retrieval (from information centres, archives) and data retrieval services are some of the examples.

Distribution services without user individual presentation control include conventional broadcast services such as television and radio. This service provides continuous flow of information where users can obtain unlimited access to the information. High speed unrestricted digital information distribution, document distribution (for electronic newspaper, electronic publishing), video information distribution (for the distribution of video/audio signals), television programme distribution (of both existing quality TV and high-definition TV) services are some of the examples.

Distribution services with user-individual presentation control allows the central source to distribute the information to a large or small number of users based on some type of cyclical repetition for example full channel broadcast videography (for tele-advertising, news retrieval, remote education and training).
Chapter 3

Asynchronous Transfer Mode

The Asynchronous Transfer Mode (ATM) technology has been accepted by the standards bodies to be the transfer mode for a universal broadband network, the B-ISDN. Section 3.1 describes the typical topology of an ATM network. Section 3.2 describes the ATM connection identifiers, and section 3.3 describes the various ATM switch architectures. The feasibility of multicasting over ATM is discussed in section 3.4. Section 3.5 and 3.6 describe in detail the ATM and ATM Adaptation Layer (AAL) layers respectively. Different types of AAL have been defined for different classes of services, section 3.7 describes in detail the functioning of AAL 1, AAL 2, AAL 3/4, and AAL 5. A properly constructed ATM network must manage traffic fairly, and should be able to adapt to unforseen traffic patterns. The various methods proposed for congestion and traffic control are discussed in section 3.8.

3.1 ATM Topology:

ATM has been selected by ITU-T, ANSI, and the ATM Forum to be the transfer mode for B-ISDN [13][23]. It provides a high speed, low-delay multiplexing and switching network to support any type of user traffic, such as voice, data, or video.
ATM segments and multiplexes user traffic into small, fixed length units called cells. The cell is 53 octets, with 5 octets reserved for the cell header. Each cell is identified with virtual circuit identifiers contained in the cell header. An ATM network uses these identifiers to relay the traffic through the high speed switches from the sending customer premises equipment (CPE) to the receiving CPE.

An ATM network consists of ATM switches interconnected by point to point ATM links or interfaces. ATM switches support two kinds of interfaces, they are user-network interface (UNI) and network-node interface (NNI) [93][94]. UNI connects ATM end systems to an ATM switch. There are two forms of UNI, they are public UNI and private UNI. Public UNI defines the interface between a public service ATM network and a private ATM switch. A private UNI defines an ATM interface with an end user and a private ATM switch. The NNI can be defined as an interface connecting two ATM switches, precisely NNI is any physical or logical link across two ATM switches which exchange the NNI protocol.

The ATM interfaces and topology are organized around the ISDN model [REF]. A possible ATM based topology can be as shown in figure 3.1. The wide area network (WAN) can be a public network offered by a public telecommunications operator [91]. The ATM nodes at this interface use a public UNI to the CPE ATM nodes. The CPE ATM nodes connect with private UNIs to the various equipment on the user’s side.

3.2 ATM Connection Identifiers:

ATM networks are fundamentally connection oriented, a virtual connection has to be setup across the ATM network before any data transfer is done. An ATM circuit can be a virtual path connection (VPC) or a virtual channel connection (VCC) [93][94]. The VPC is
Figure 3.1: A Typical ATM Topology
identified by the virtual path identifier (VPI) and the VCC is identified by the virtual channel identifier (VCI). The VPI and the VCI are placed in the cell header and are assigned to the connection at the time of connection establishment. A virtual path is a bundle of virtual channels, all of which are switched transparently across the ATM network on the basis of the common VPI. The manner in which the VPIs or VCIs are processed in the network is not yet defined in the ATM standards. All VCIs and VPIs have only local significance across a particular link, and are remapped as appropriate at each switch.

3.3 ATM Switching:

The ATM switch is one of the most important components of a virtual connection. A switch fabric is necessary to establish a connection between an arbitrary pair of inputs and outputs within a switch node.

A switching element is the basic unit of a switch fabric [31]. At the input port the routing information of an incoming cell is analysed and the cell is then directed to the correct output port. In general, a switching element consists of an interconnection network, an input controller (IC) for each incoming link and an output controller (OC) for each outgoing link as shown in figure 3.2 [86]. To prevent excessive cell loss in the case of internal collisions, buffers have to be provided within the switching element.

Arriving cells will be synchronized to the internal clock by the IC. The OC transports cells which have been received from the interconnection network towards the destination. ICs and OCs are coupled by the interconnection network.

The interconnection network can be constructed by using a rectangular matrix of cross points (Matrix Type) as shown in figure 3.3 [86]. The buffers can be placed at any of
the input or output controllers or at the cross points [95], (the figure shows buffers placed at the output controllers) each design has its own advantages and disadvantages.

The Central Memory switching element shown in figure 3.4 has all the input and output controllers attached directly to a common memory which is accessible for all the input and output controllers [95].

Since all the switching element buffers share one common memory, a significant reduction of the total memory requirements can be achieved in comparison with physically separated buffers. On the other hand, a high degree of internal parallelism is necessary to keep the frequency of memory access within a realizable range.

In the Bus Type switching element, the interconnection network can be realized by a high speed time division multiplexed (TDM) bus as shown in figure 3.5[86]. The total capacity of the bus should at least be equal to the sum of the capacities of all input links for conflict free transmission. Buffers are only required at the output controllers because each input controller can transfer its cell to the destination before the next cell arrives whereas several cells may arrive at the same output controller[95].

Figure 3.2: Basic Switching Element

| IC | Input controller |
| OC | Output controller |

The Central Memory switching element shown in figure 3.4 has all the input and output controllers attached directly to a common memory which is accessible for all the input and output controllers [95]. Since all the switching element buffers share one common memory, a significant reduction of the total memory requirements can be achieved in comparison with physically separated buffers. On the other hand, a high degree of internal parallelism is necessary to keep the frequency of memory access within a realizable range.

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Figure 3.3: Matrix Switching Element With Output Buffers

Figure 3.4: Central Memory Switching Element

IC  Input controller
OC  Output controller
Figure 3.5: Bus Type Switching Element

IC    Input controller
OC    Output controller
TDM    Time division multiplexing

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In the Ring Type switching element shown in figure 3.6, all input and output controllers are interconnected via a ring network. The ring structure has the advantage over the bus structure in that a time slot can be used several times within one rotation.

![Diagram](image)

**Figure 3.6: Ring-Type Switching Element**

The switching elements can be combined to form single stage networks (Extended Switching Matrix Network, Funnel Type Network, Shuffle Exchange Network) and multi-stage networks (Banyan, Delta Networks) [86][95].

The basic operation of an ATM switch involves routing, queuing and header translation. This is done as shown in figure 3.7 [86], the incoming ATM cells are physically switched from an inlet $I_i$ to an outlet $O_j$, at the same time their header value is translated from an incoming value ‘a’ to an outgoing value ‘b’. The values of the headers are unique on each incoming and outgoing links individually, but identical headers are allowed on different links.

The translation table is used for translating the header values. From the table it can be seen that all cells which have a header value of ‘x’ on the incoming link $I_1$ are switched to outlet $O_1$ and should have a new header value of ‘k’. All cells with a header ‘x’ on link
Figure 3.7: ATM Switching Principle

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$I_n$ are also switched to outlet $O_1$, but their header should be translated to 'n'.

A buffer is required for the switch to avoid collisions. the type of buffer to be used and its placement is a widely discussed issue, ITU-T and ATM Forum have not recommended the use of any particular switch design. There are several ATM switches being used each of them different in their use of switching elements and buffer selections [33][95].

3.4 Multicasting:

Some data services (electronic mail) and distributive services (video library access, TV distribution etc.) are characterized by point-to-multipoint communication [96]. This capability can be supported by an ATM switch fabric. For this purpose, an ATM switching element must be able to transmit copies of an incoming cell to different outlets.

A multicast connection appears much like normal virtual connection from the sender’s perspective. Cells leave the originating ATM layer on a single VPI/VCI, and are then propagated and replicated throughout the network as necessary. Multicast switches replicate incoming cells, mapping different VPI/VCLs and port numbers to each copied cell [30][93].

ATM connections can be setup as permanent virtual connections (PVC) or switched virtual connections (SVC). A PVC is connection setup by some external mechanism, typically network management, in which a set of switches between an ATM source and destination are programmed with the appropriate VPI and VCI values. An SVC is a connection that is setup automatically through a signalling protocol [97][98]. Unlike PVCs, SVCs do not require any manual interaction and are easier to setup.
3.5 The Asynchronous Transfer Mode Layer:

The ATM layer's primary responsibility is the management of the sending and receiving of cells between the user node and the network node [32][33]. The ATM cell contains a 48 octet information field and a 5 octet header as shown in figure 3.8. The octets are sent in an increasing order, starting with octet 1 of the header. Within an octet, the bits are sent in a decreasing order, starting with bit 8 [86].

At the UNI, the header structure is as shown in figure 3.9 [86]. The first field contains four bits for the Generic Flow Control (GFC). The second field is the routing field, subdivided into a VCI field of 16 bits and a VPI field of 8 bits. A Payload Type Identifier (PTI) field is coded in three bits, it identifies the type of traffic residing in the cell. The Cell Loss Priority (CLP) field is a 1 bit value. If CLP is set to 1, the cell is subject to being discarded by the network. The cell has a higher priority if CLP is set to 0.
Figure 3.9: Header Structure at the UNI

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and is treated with more care than the cell in which the CLP bit is set to 1. The Header Error Control (HEC) field consists of 8 bits. It is an error check field, which can also correct single bit errors. The HEC is calculated on the 5 octet header. An adaptive error detection and correction method is employed by the ATM. The transmitter calculates the HEC value on the first four octets of the header using the polynomial generated by the header bits (excluding the HEC field) multiplied by 8 and dividing this polynomial by the generator polynomial \( x^8 + x^2 + x + 1 \). The coset value 01010101 is XORed with the 8 bit remainder and the result is placed in the last octet of the header. The complimentary calculation is performed at the receiver.

At the NNI, the header structure as shown in figure 3.10 [91] is identical to the one at the UNI except for the GFC field which is replaced by 4 additional VPI bits. This results in a VPI field of 12 bits at the NNI.

The VCI, VPI, and other parts of the first four octets of the cell can be coded in various formats to identify non-user payload cells. There several preassigned cell header values some of them are shown in table 2 [86].

3.6 ATM Adaptation Layer:

The AAL is an essential part of the ATM network, it conveys the user traffic to the cell based network. The AAL maps the user, control, management PDUs into the information field of one or more consecutive ATM cells of a virtual connection, and vice versa[33]. For the transfer of user traffic, the AAL operates at the end points of the virtual connection and does not operate within the ATM network. For control and management traffic, AAL is invoked at the network node of the UNI.
Figure 3.10: Header Structure at the NNI
Table 3.1: Pre-assigned Values of the Cell Header at the ATM Layer by ITU-T

<table>
<thead>
<tr>
<th>Cell type</th>
<th>VPI</th>
<th>VCI</th>
<th>PTI</th>
<th>CLP</th>
</tr>
</thead>
<tbody>
<tr>
<td>Unassigned cells</td>
<td>00000000</td>
<td>00000000</td>
<td>0</td>
<td>0</td>
</tr>
<tr>
<td>Meta-signalling cells</td>
<td>xxxxxxxx</td>
<td>00000000</td>
<td>0 A 0</td>
<td>B</td>
</tr>
<tr>
<td>General broadcast cells</td>
<td>xxxxxxxx</td>
<td>00000000</td>
<td>0 A A</td>
<td>B</td>
</tr>
<tr>
<td>Point-to-point signalling cells</td>
<td>xxxxxxxx</td>
<td>00000000</td>
<td>0 A A</td>
<td>B</td>
</tr>
<tr>
<td>Segment OAM flow F4 cells</td>
<td>yyyyyyyy</td>
<td>00000000</td>
<td>0 A 0</td>
<td>A</td>
</tr>
<tr>
<td>End-to-end OAM flow F4 cells</td>
<td>yyyyyyy</td>
<td>00000000</td>
<td>0 A 0</td>
<td>A</td>
</tr>
<tr>
<td>Segment OAM flow F5 cells</td>
<td>yyyyyyy</td>
<td>xxxxxxxx</td>
<td>100</td>
<td>A</td>
</tr>
<tr>
<td>End-to-end OAM flow F5 cells</td>
<td>yyyyyyy</td>
<td>xxxxxxxx</td>
<td>101</td>
<td>A</td>
</tr>
<tr>
<td>Resource management cells</td>
<td>yyyyyyy</td>
<td>xxxxxxxx</td>
<td>110</td>
<td>A</td>
</tr>
<tr>
<td>User information cells</td>
<td>yyyyyyy</td>
<td>vvvvvvvv</td>
<td>0 C U</td>
<td>L</td>
</tr>
</tbody>
</table>

A : Bit is available for use by the ATM layer.
B : Bit is set by originating entity, but network may change value.
C : Explicit Forward Congestion Indication bit (EFCI).
L : Cell Loss Priority bit.
U : ATM layer user to ATM layer user indication bit.
x : Any VPI value. For VPI = 0, the VCI value is valid for signalling with local exchange.
y : Any VPI value.
z : Any VCI value other than 0.
v : Any VCI value above 0031 H.

The AAL is designed to support different types of applications and different types of traffic, such as voice, video, and data. The services which will be transported over the ATM layer are classified into Class A, Class B, Class C and Class D as shown in figure 3.11[91].

In Class A, a time relation exists between source and destination. The bit rate is constant and the service is connection oriented. Class A is designed to support a Constant Bit Rate(CBR) requirement for applications like high quality video.

In Class B, again a time relation exists between source and destination. the bit rate is variable and the service is connection oriented. Class B is designed to support Variable Bit Rate(VBR) applications like VBR video and audio.

In Class C, there is no time relation between the source and destination, the bit
<table>
<thead>
<tr>
<th></th>
<th>Class A</th>
<th>Class B</th>
<th>Class C</th>
<th>Class D</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Timing between</strong></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td><strong>source and</strong></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td><strong>destination</strong></td>
<td>Required</td>
<td>Not required</td>
<td></td>
<td></td>
</tr>
<tr>
<td><strong>Bit rate</strong></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td></td>
<td>Constant</td>
<td>Variable</td>
<td></td>
<td></td>
</tr>
<tr>
<td><strong>Connection mode</strong></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td></td>
<td>Connection oriented</td>
<td>Connectionless</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

Figure 3.11: AAL Service Classes

rate is variable and the service is connection oriented. Examples of Class C services are connection oriented data transfer, and signalling.

In Class D, like in Class C, there is no time relation between the source and destination and the bit rate is variable but the service is connectionless. Class D services are typically connectionless data transport services like Switched Multi Megabit Data services (SMDS).

The AAL is subdivided into SAR (Segmentation and Reassembly) and CS (Convergence Sublayer) sublayers [REF]. The SAR sublayer processes the user PDUs that are different in size and format into ATM cells at the sending side and reassembles the cells into user formatted PDUs at the receiving side. The CS performs functions like message identification, time clock recovery etc. The function of CS sublayer depends upon the type of traffic being processed by the AAL. For some AAL types supporting data transport over ATM, the CS is further divided into Common Part Convergence Sublayer (CPCS) and Service Specific Convergence Sublayer (SSCS) [33][86][91].

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3.7 Types of ATM Adaptation Layers:

AAL 1 is defined for CBR services[86], AAL 2 is being defined (not yet standardised) for variable bit rate services for class B traffic, AAL 3/4 is defined for data services, and AAL 5 is defined for connectionless data services.

3.7.1 AAL Type 1:

For CBR services, the bit rate is constant and synchronized between the sender and receiver. ITU-T Recommendation I.363 specifies the AAL 1 services as- transfer of Service Data Units(SDUs) at a constant bit rate, transfer of timing information between the source and destination, transfer of data structure information, and indication of lost or errored information. The Type 1 AAL is divided into CS and SAR sublayers [86][91].

At the sending side, the SAR sublayer accepts a 47 octet block of data from the CS, and then adds a one octet SAR-PDU header to each block to form the SAR-PDU. At the receiving end, the SAR sublayer gets a 48 byte block from the ATM layer, and then separates the SAR-PDU header. The 47 octet block of SAR-PDU payload is then passed to the CS.

The SAR-PDU as shown in figure 3.12 is composed of a single byte header and a 47 byte SAR-PDU payload. The header carries a 3 bit Sequenc Number(SN) which can be used to detect lost or misinserted cells, a single bit CS layer indication(CSI) to indicate the existence of a CS layer, and a 4 bit header protection field called the Sequence Number Protection(SNP). The SNP is a 3 bit Cyclic Redundancy Check(CRC) of the SN field, with an additional parity bit to protect the CRC.

Various CS functions are described in recommendation I.363 for different services.
Figure 3.12: AAL1 SAR Structure

- CSI = CS Indication
- SN = Sequence Number
- SNP = Sequence Number Protection
- SDU = Service Data Unit

48 Bytes
The CS functions are expected to allow for synchronous service (where the source and destination are locked to a network derived clock and asynchronous service (where the data is clocked at a constant rate, but is not specifically locked to the clock of the network carrying the cells).

Error correction is applied to bit errors where possible, and padding is applied to fill in when cells are lost. The particular mechanisms depend on the service being carried, and are agreed at connection establishment time [99]. There is no mechanism for retransmission of lost cells.

3.7.2 AAL Type 2:

AAL 2 is employed for VBR services where a timing relation is required between the source and destination sides. Class B traffic such as VBR audio and video falls into this category. Since the source is generating a variable bit rate, it is possible that cells are not completely filled [91], and that the filling level varies from cell to cell, therefore more functions are required in the SAR. The standards bodies have not yet defined AAL 2 fully, but ITU-T has mentioned a possible way of implementing the SAR PDU as shown in figure 3.13 [86].

The SN field contains the sequence number to allow the recovery of lost or misrouted cells.

The IT (Information Type) field indicates the beginning of a message (BOM), continuation of a message (COM), end of a message (EOM), or that the cell transports timing or other information. BOM, COM or EOM indicate that the respective cell is the first, middle or last cell of a message. The LI (Length Indicator) field indicates the number of useful bytes in partially filled cells. The CRC field may allow the SAR to correct bit errors in the SAR-SDU. The coding and length of each field are for further study.

The CS sublayer will have to perform functions like clock recovery, handling of lost
Figure 3.13: AAI 2 SAR Structure

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or misdelivered cells, forward error correction. The CS functions and protocol are also still for further study[REF].

3.7.3 AAL Type 3/4:

For Class C and Class D services, initially two types of AAL (AAL 3, AAL 4) were proposed by the ITU-T [REF]. Realisation that the functional differences were almost non existent led to their merger into a single standard AAL 3/4 [33].

The Convergence sublayer of AAL 3/4 is further subdivided into Service Specific Convergence Sublayer (SSCS) and Common Part Convergence Sublayer (CPCS). The CPCS contains only the basic processing necessary to transport a variable sized block of data across a virtual connection. Additional functions such as flow control and the transmission of lost or corrupted data, are implemented within the SSCS.

The basic AAL 3/4 CPCS function provides a non-assured service[REF], where each user frame is contained in one AAL-SDU and transferred in one CPCS-PDU. In this mode the AAL does not guarantee the successful transfer of any given AAL-SDU from peer to peer.

Figure 3.14 shows a simplification of the transmission process on a single virtual connection [91]. An AAL-SDU (user PDU) is encapsulated to generate a CPCS-PDU. This is then segmented into a series of 44 byte SAR-PDU payloads, wrapped with 2 byte headers and trailers to create 48 byte SAR-PDUs, and inserted into the payload fields of cells for transmission.

The CPCS-PDU encapsulation contains information which enables the CPCS function to verify the correct reception of the entire CPCS-PDU, shown in Figure 3.15. The PAD (padding) field ensures the CPCS-PDU trailer is 32 bit aligned, and the AL (align-
Figure 3.14: Transmission Using AAL 3/4

- Higher Layer PDU
- AAL SDU
- Common Part convergence Sublayer
- CPCS_PDU HEADER
- CPCS_PDU PAYLOAD
- CPCS_PDU TRAILER
- Segmentation and Reassembly Sublayer
- AAL3/4

- CELL HEADER
- ATM CELL PAYLOAD
- To ATM cell transport layer
ment) field pads the trailer to 32 bits. The Common Part Indicator (CPI) specifies how the CPCS should interpret the remaining fields of the header and trailer. Presently a CPI of zero indicates that the BAsize field contains an estimate of the incoming CPCS-PDU's size in bytes, and the Length field contains the exact size of the CPCS-PDU payload in bytes.

Figure 3.15: CPCS-PDU Encapsulation

The BAsize field may be used by the reassembly machine to preallocate buffer space. The Length field aids in the detection of reassembly errors such as loss or gain of cells. Additional error detection is possible using the Btag and Etag fields. Both fields are set to the same value when the CPCS-PDU is transmitted, the actual value used being
unimportant so long as it changes for successive CPCS-PDUs.

Figure 3.16 shows a breakdown of the SAR-PDU, header, and trailer. The ST field indicates what type of information is carried within the SAR-PDU. Four types exist, they are: Beginning Of Message (BOM), Continuation Of Message (COM), End Of Message (EOM), and Single Segment Message (SSM). If a CPCS-PDU is less than 45 bytes long it will fit within the payload of a single SAR-PDU and will be sent as an SSM. Longer CPCS-PDUs are sent as a sequence of SAR-PDUs, beginning with a BOM, followed by zero or more COMs, and ending with an EOM.

While segmenting and sending a given CPCS-PDU, the SN field is incremented by one, modulo 16, for each SAR-PDU sent. The LI field indicates how many bytes of the payload are actually valid. It is always 44 for BOMs and COMs, but may be less for EOMs and SSMs corresponding to CPCS-PDUs that do not finish on a 44 byte boundary. The CRC field provides bit error detection across the entire SAR-PDU.

A distinctive feature of AAL 3/4 is that it can multiplex different streams of AAL-SDUs across a single virtual connection. Multiplexing is achieved through the 10 bit MID field. By associating each stream of AAL-SDUs, and hence CPCS-PDUs, with a distinct MID value, SAR-PDUs may be interleaved on the virtual connection and successfully extracted at the receiving end. When a user process is not multiplexing AAL-SDUs from distinct sources the MID field is set to zero.

At the receiving end of a virtual connection the SAR layer aims to recreate the CPCS-PDU. Once this is obtained, the original AAL-SDU may be extracted and passed up to the receiving user process. Each time a cell arrives it is considered to either represent a new CPCS-PDU (if it is BOM or SSM) or add to a CPCS-PDU already being assembled.
<table>
<thead>
<tr>
<th>2 byte Header</th>
<th>44 bytes</th>
<th>2 byte Trailer</th>
</tr>
</thead>
<tbody>
<tr>
<td>ST</td>
<td>SN</td>
<td>MID</td>
</tr>
<tr>
<td>2</td>
<td>4</td>
<td>10</td>
</tr>
</tbody>
</table>

ST: Segment Type
MID: Multiplexing Identification
CRC: Cyclic Redundancy Check
SN: Sequence Number
LI: Length Indication
Under normal operation the receiving AAL will, on a given virtual connection, see a repeating sequence of BOM-COMs-EOM....BOM-COMs-EOM.... cells (interspersed with SSMs if the traffic is a varied collection of short and long AAL-SDUs). The first BOM causes the AAL to note the MID and SN fields, and then look for following COMs which contain the same MID and have correctly incrementing SN fields. The payload is extracted from each SAR-PDU to form the CPCS-PDU. Finally when an EOM arrives, in sequence and with a matching MID value, the CPCS function is passed the final payload with an indication that the CPCS-PDU should now be complete.

Final error checking involves matching the Etag with the Btag, and ensuring the Length field matches the received data. AAL3/4 relies on the CRC in each SAR-PDU to ensure the CPCS-PDU’s components are error free. Handling multiplexed AAL-SDU traffic requires several instances of the SAR and CPCS entities to exist simultaneously. Each instance of the SAR/CPCS is associated with a valid MID, and a particular AAL service user.

COM and EOM SAR-PDUs that arrive with a MID value not corresponding to a current CPCS-PDU are ignored. Those that arrive with an out of sequence SN field are considered to indicate an error in the transmission process, and the CPCS must abort reassembly of the current CPCS-PDU. The arrival of a BOM with the MID of a CPCS-PDU that is still being reassembled also causes the current CPCS-PDU to be aborted.

3.7.4 AAL Type 5:

In AAL 5 unlike in AAL 3/4, the responsibility of multiplexing is shifted to a higher layer, error detection facilities are simplified, the BAsize facility to preallocate reassembly buffers is discarded, and only one type of CPCS-PDU format is allowed []. All SAR level encapsulation
is removed, enabling 48 bytes of CPCS-PDU to be carried within each cell. The SAR-PDU type is now indicated by the User-to-User (UU) parameter in the Payload Type Indication (PTI) field of the ATM cell header. Figure 3.17 shows a simplified AAL5 transmission of an AAL-SDU [86].

![Diagram of AAL5 transmission](image)

Only two SAR-PDU types are used under AAL5. The UU is set to one to mark the last (or only, in the case of a small CPCS-PDU) SAR-PDU of a CPCS-PDU. If the CPCS-PDU spans more than one cell, cells with the UU set to zero carry the beginning and continuation SAR-PDUs.

The CPCS-PDU is shown in figure 3.18. The Padding field, which may be between 0 and 47 bytes long, ensures that the Trailer always occupies the last 8 bytes of the last SAR-PDU, and the overall CPCS-PDU is aligned on a 48 byte boundary. Primary error detection is provided by the 4 byte CRC field, which covers the entire CPCS-PDU (including the Length field, but excluding the CRC itself). The Length field specifies the payload size.
in bytes. The Length field also provides a backup protection against cell loss or gain, in the unlikely event it is not picked up by the CRC.

The Common Part Indicator (CPI) field is used to interpret subsequent fields for the CPCS functions in the CPCS header and trailer. The only legal encoding at the moment is all zeros, indicating that the CPCS contains user data [100]. Other uses are for further study.

Figure 3.18: CPCS-PDU Format for AAL 5

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Because AAL5 does not support the simultaneous multiplexing of CPCS-PDUs on a single virtual connection, all SAR-PDUs carry either the current CPCS-PDU or the next one. The first SAR-PDU to arrive after a SAR-PDU with UU set to 1 is assumed to be the beginning of the next CPCS-PDU, irrespective of whether it is a SAR-PDU with UU set to 1. A CPCS-PDU shorter than 41 bytes may be represented by a sole SAR-PDU with UU set to 1.

3.8 Traffic Management and Congestion Control:

A properly constructed ATM network must manage traffic fairly and provide effective allocation of the network capacity for different sorts of applications, such as voice, video, and data [99][101][102]. The ATM network must also provide cost effective operations relative to the quality of service (QOS) stipulated by the user, and it must be able to support the different delay requirements of the applications, an important support function known as Cell Delay Variation (CDV) management.

The ATM network must be able to adapt to unforeseen traffic patterns (Traffic Control) for example, unusual bursts of traffic from the various end user devices or applications. Also, the network must be able to shed traffic in certain conditions to prevent or react to congestion (Congestion Control).

To attain the goals of traffic and congestion control several schemes have been proposed. At the connection admission level, every new connection is evaluated for its likelyy impact on pre-existing conditions using a set of procedures called Connection Admission Control (CAC) procedures. A connection request for a given call is accepted only when sufficient resources are available to carry the new connection through the whole network at its
requested Quality of Service while maintaining the agreed QOS of already established connections in the network. During the connection establishment procedure, a traffic contract specification is negotiated between the user and network to enable CAC to make reliable connection acceptance or denial decisions.

The traffic contract contains information such as the peak cell rate (PCR), sustainable cell rate (SCR), burst tolerance, cell error ratio, cell loss ratio, cell misinsertion rate, cell delay variation (CDV) etc.

After a connection is granted, and the network has reserved resources for the connection, each user's session is monitored or "policed" by the network. This is done at the User-Network Interface (UNI) and is known as Usage Parameter Control (UPC) and at the Network-Node Interface (NNI) and is called Network Parameter Control (NPC). The main purpose of UPC and NPC is to ensure that a particular connection does not exceed the bounds agreed upon during the connection admission stage.

The 'Leaky Bucket' is the first basic policing function to be standardised by the ITU-T. Various forms of this technique have been proposed [103].

The original form [104] proposed a model where packets of data would arrive at a policing point and be passed through ('leaked') at a capped maximum rate. If packet arrival exceeded this rate the excess packets would be buffered (in a 'bucket'), with the regular leaking of packets acting to eventually empty the bucket. If imbalance between packet arrival and leak rates continued for too long, the bucket would be filled and subsequent packets simply discarded. If the arrival rate was less than the leak rate, and assuming an empty buffer, packets would be passed through straight away.

This scheme prevents packets from ever being passed faster than the leak rate.
limitation of this technique is that while large buffers minimise the packet loss from large bursts, they also introduce extra end to end delays.

Another scheme provides support for cell bursts in the form of a counter which is actually a token pool, with a finite number of tokens[103]. When a cell arrives it will be passed onwards only if a free token is available in the pool wherein the token is removed and the cell transmitted. As cells continue to arrive the token pool may be depleted to the point where the next cell to arrive cannot obtain a token. When no token is available the cell may be buffered until a token is free or discarded depending on the actual implementation. Acting alongside this process is a local clock regenerating tokens at a fixed rate, the pool is replenished at a rate equivalent to the desired leak rate. When the pool is considered to be full no more tokens are generated.

If the cell arrival rate is equal to the leak rate, the token pool will retain a constant number of free tokens. However, a burst of cells will be passed through without modification, provided there are enough tokens in the pool at the time the burst occurs. The combination of leak rate and token pool size must be chosen carefully.

Another scheme called the Virtual Leaky Bucket [104] utilises the CLP bit in the cell header to indicate cells in violation of the agreed traffic limit. It can be regarded as a form of prioritising policing unit, because it attempts to lower the cell loss priority of violating cells rather than immediately buffering or discarding them. If a cell with CLP=0 arrives and there is no token available it is passed on with CLP=1. Violating cells that already have CLP=1 may be discarded or buffered according to some local algorithm.

The virtual leaky bucket has better burst tolerance than the token pool mechanism. A burst will only suffer cell loss if it’s violation of the virtual leaky bucket limits coincides
with actual congestion further along the path, at which point the tagged cells will be discarded. If the network actually has the required bandwidth available at the time of the burst all cells will make it to their destination.

The fundamental shortcoming of all these variations of the leaky bucket is its inability to effectively deal with the wide range of possibly bursty traffic [105].

The ITU-T and ATM Forum have standardised an algorithm for policing traffic at the UNI, it is called the Generic Cell Rate Algorithm (GCRA) [106]. The GCRA is implemented as a 'continuous state' leaky bucket algorithm or a 'virtual scheduling' (VS) algorithm. The purpose of these two algorithms is to identify the conforming cells and non conforming cells. GCRA uses two real valued parameters I and L, denoted as Increment and Limit and is described as GCRA(I,L).

At the arrival of a cell, the VS algorithm calculates the Theoretically predicted Arrival Time (TAT) of the cell assuming equally spaced cells (the distance between 2 consecutive cells is denoted by I) when the source is active. If the actual arrival time $t_a$ of a cell is after $TAT - L$ (L is the limit and it represents a certain tolerance value), then the cell is conforming, otherwise the cell arrived too early and is considered as non conforming.

The continuous state leaky bucket algorithm can be viewed as a finite capacity (capacity is L+1) bucket algorithm whose content leaks out at a continuous rate of 1 per time unit and whose content is increased by I for each conforming cell. If at a cell arrival the content of the bucket is less than L, then the cell is conforming, otherwise the cell is non conforming.

After the connection is granted and the cells are being transmitted through the network, controlling functions are necessary to minimize the duration and impact of the
transient overload conditions. These are called reactive controls [91][104]. The usefulness of reactive mechanisms are limited by the speed at which they function, they must be faster than the duration of the congestion condition that triggers them.

Cell discarding by policing units is a form of reactive control at the cell level. It is a rapid response to a localised instance of excess traffic, reducing the offered load to nodes beyond the policing unit. A bi-level discard policy is built into every virtual connection through the CLP bit in the cell header. Two possible uses have been proposed; the cell source may guide the policing units by setting CLP=1 on cells that it considers less important, or virtual policing units may act by changing CLP=0 to CLP=1 on cells that exceed the allowed traffic parameters. In either case, when a policing function anywhere in the network considers it necessary to shed load it will begin by dropping CLP=1 cells.

The ATM Forum is in the process of defining a new type of service, called the Available Bit Rate (ABR) [106]. ABR provides a mechanism for controlling traffic flow through LAN based workstations and the routers that service these workstations. Since LAN devices have no "contract" with a network, no easy method is available to relate the AAL CBR and VBR idea to LAN environment.

Several proposals are being considered for ABR one of them is the use of the Payload Type field in the cell header to carry congestion notification to the end user[104]. When informed of the congestion being experienced by traffic on that connection, the end user becomes responsible for slowing down its traffic. The complication is that the notification is initially carried to the receiving end of the connection, and must be passed back to the sender by some other means. This method is called Forward Explicit Congestion Notification (FECN).
An alternative is for the network nodes themselves to originate a congestion indication cell back to the source. This method is called Backward Explicit Congestion Notification (BECN) [104].

Another form of reactive admission control involves post-acceptance modification of a connection's parameters in order to cope with a new connection's requirements. Certain connection classes are considered to have higher priority than others, so low priority connections may have their allocated bandwidth reduced to make room for a subsequent higher priority connection request [104].

The end user equipment implement policing functions of their own in the form of 'traffic shaping' by making the traffic shape more acceptable to the network. Traffic shaping is basically altering the stream of cells (cell rate reduction, traffic discarding) emitted into a virtual channel or virtual path connection [107].
Chapter 4

Picture Archiving and Communication Systems

The ability to digitize images has lead to the prospect of reducing the physical space requirements, material costs, and manual labor of traditional film handling tasks in hospitals. The system which handles the acquisition, storage, and transmission of medical images is called a Picture Archiving and Communication System (PACS). The various components which constitute a PACS, and the issues of integration of PACS with other digital systems are described in section 4.1. To facilitate easy transmission of medical images in a hospital (and among hospitals) which is(are) usually multi-modality, and multi-vendor environments, several medical imaging standards were developed, the most popular standards are described in section 4.2. With hospitals having to handle very large amounts of data, efficient image management is very important. The various aspects involved in medical image management, like database design, prefetching, and image compression are discussed in section 4.3. The different classes of PACS and the design concepts involved in the development of a PACS are described in sections 4.4 and 4.5 respectively. Reliability is a very important issue in a medical environment. section 4.6 deals with this issue. Section 4.7 lists some of the existing PACS and describes in brief, a few of the proprietary networks, some of these systems have used.
4.1 General Concepts of Picture Archiving and Communication Systems (PACS):

Computerized images make up a substantial fraction of routine diagnostic radiology examinations. Images are directly produced in digital form in computed tomography, magnetic resonance imaging (MRI), ultrasound, nuclear medicine, computed radiography, and digital subtraction angiography [36][46][108][109]. The remaining radiologic images are in analog form. Conventional analog radiographs can be scanned by film digitizers for entry into a computer. Whether initially created by a digital device or converted by means of a film digitizer, any radiologic image can be translated into digital form with the help of present technology. Once in digital form, images can be enhanced, transferred, and stored by computers [110][111].

One of the major burdens of the radiology department is the storage and retrieval of the images. At present, in most of the hospitals all over the world, medical images are recorded and stored on film [112]. Even images such as CT and MRI scans, which are inherently digital, are typically transferred to film once the technologist has optimized them for viewing. Radiologists then place the filmed images on illuminated light boxes, where the films can be analyzed in batches and can be compared easily with previous and related studies. Occasionally, as in the case of ultrasound studies, the images are transferred to videotape for later review and interpretation [113].

Film storage requires a large amount of space in a radiology department. Typically, departments have the capacity to store films only for those patients who have had studies within the past 6 to 12 months. Older studies, usually retained for at least 7 years, are stored in a basement or warehouse [37].

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The ability to digitize images has lead to the prospect of reducing the physical space requirements, material cost, and manual labor of traditional film-handling tasks. This is possible through on-line digital archiving, rapid retrieval of images via querying of image databases, and high speed transmission of images over communication networks [34]. A system with such capabilities was visualised in the early 1980s and the term Picture Archiving and Communication Systems (PACS) was used to describe it. The issue of a totally filmless radiology department was raised at the First International Conference and Workshop on Picture Archiving And Communication Systems (PACS) for Medical Applications and has come a long way since then. Now the terms PACS and IMAC (Image Management and Communication Systems) are used interchangeably [36].

Picture Archiving and Communication Systems (PACS) now are considered to be multimedia systems specifically designed to handle the high speed imaging and transmission of medical data. The objectives of PACS are to digitize, display, and if necessary enhance, images captured by computed tomography, magnetic resonance, computed radiography and other techniques, and bring order and accessibility via wideband networks to the files of film in which they are stored [37][40][41][114].

4.1.1 Components of a PACS:

A Picture Archiving and Communication System is typically composed of the following components:

- Data Acquisition Devices: Computed Radiography (CR), Fluoroscopy, Angiography, Computed Tomography (CT), Magnetic Resonance Imaging (MRI), Nuclear Medicine (NM), and Ultrasound.
• Data Storage and Long-term Archive Subsystem: Receives acquired image data from the acquisition devices, transmits requested data from the storage device to the work stations, and archives images for long-term storage.

• Workstations: Generates the requests to data storage subsystem, receives images from the data storage subsystem, processes the images if required, and displays images on the monitors for viewing.

• Networking: Provides the channel to distribute and communicate between image databases and workstations, and between image databases and acquisition devices.

• System Operation: Involves the overall aspects of the operation such as data security, system reliability and maintenance, system interfacing to other hospital computer systems, e.g., Hospital Information System (HIS), and Radiology Information System (RIS).

A full fledged PACS often includes a teleradiology sub-system and supports long-term electronic storage of diagnostic images [39][41][49][115][116]. It is not just limited to a specific department but is used to manage the flow of diagnostic images throughout a health care system. The benefits of a PACS are:

• Substantial reduction of film and film-related expenses [35].

• More efficient storage, retrieval and availability of imaging records and reports.

• More efficient use of imaging equipment, resulting in reduced capital equipment cost [36].

• Faster turnaround for imaging diagnoses, resulting in fewer patient admissions and shorter hospital stays.
- Consolidation of imaging records with other medical records, eliminating labor intensive, redundant manual systems of archiving and retrieving medical records.

Teleradiology refers to the use of computers and electronic communication networks to transmit diagnostic images acquired at one location to another location for review and interpretation. It was first used for on-call review of imaging studies, but now is being used more frequently to provide primary diagnoses [116]. A teleradiology system provides the following benefits:

- Timely distribution of imaging data and patient reports to referring physicians and consultants.
- Teaching files of interesting cases available to a wider audience.
- Universal access to large databases of text and images for research and education [115].
- Improved integration of radiological services for multi-hospital /clinic health care provider consortia [40].
- Offers consulting-at-a-distance with sub-speciality radiologists [117].
- Provides radiologists in the community or in rural areas immediate access to large academic centers for help in the interpretation of difficult and problematic cases.

4.1.2 Integration of Hospital Information Systems and Radiology Information Systems:

The hospital information system (HIS) contains patient-specific data and administrative data [43][44]. It is used to manage the information that health professionals need to per-
form their jobs effectively and efficiently. The radiology information system (RIS) helps in scheduling equipment and examination rooms for maximum usage, assisting film library management, tracking the locations of films, and collecting and analyzing the data necessary for evaluation and planning decisions [118][119].

The PACS requires several different kinds of text data from the HIS/RIS in order to operate properly. The alphanumerical medical data of a patient such as reports, laboratory results, patient letters, medication and so on, which are commonly stored in an HIS/RIS, are needed for various tasks in the radiologic process, such as preparation of an exam, and reporting of an exam.

The HIS/RIS sends current patient identification and order entry information to the PACS so that it can register patients and process orders. All patient demographic changes are sent to the PACS to prevent patient mismatch and avoid duplicate patient registration. Examination progress information is sent to the PACS to trigger event processing, such as printing labels when the patient arrives. Radiology reports are entered on the HIS/RIS and are passed to the PACS so that they can be displayed on the PACS in conjunction with the images. Clinical scheduling information is also needed for prefetching images from the storage archive. Figure 4.1 [44] shows the event trigger mechanism for transferring information from HIS to RIS, and from RIS to the PACS. The PACS must supply the HIS with information about examinations that have been performed. Notification of completion of an examination must be sent to the HIS in order to update the order status and identify which images have been digitized. Research is going on to enable the entry of radiology reports on the PACS and forward them to the HIS/RIS[45].

Attempts to integrate HIS/RIS which are predominantly text-based with PACS
which is image-based were met with difficulties. The differences between the two systems were identified to be:

- The systems mostly run on different platforms, where various communication profiles are used [44][45].

- The syntax and contents of the messages that can be sent by one system do not match those of the messages that can be received by the other system.

- Often the definition of the data elements used in one system do not match those of the elements in the other system.

- Different code may be used to store the same information in different systems.

- The information exchange between the connected systems may follow various data transaction scenarios. For example, a PACS may request for patient and exam data when it receives images from the acquisition device, whereas the HIS/RIS may send the patient and exam data to the PACS as soon as the appointment for the exam is made. The acknowledgement of the receipt of the messages may be performed in various ways. In each scenario the information exchange is performed by a different set of messages.

- A HIS/RIS-PACS interface may not be a one-to-one interface, but an interface between multiple text-based systems and multiple imaging systems.

After considerable research, it was decided that creating a generic HIS/RIS-PACS interface so that any combination of HIS/RIS and PACS can be connected was the best solution [45][118].
4.2 Medical Imaging Standards:

Any digital image on a computer is basically an array of dots displayed in rows and columns. Each dot makes up one element of the image called a picture element (pixel). The computer keeps track of the precise location of each pixel by assigning each one an individual address in its display memory area. At every address, the computer allocates a certain number of bits to hold the intensity value of the corresponding pixel. The number of bits stored per pixel typically ranges from 8 to 16 in radiologic imaging devices and each bit can take a value of either one or zero. Thus if 8 bits are available, the number of unique combinations of ones and zeros is $2^8$. A total of 256 different shades of gray are possible in an 8-bit video display system.

The digitized images can be transmitted from one computer to the other over cable or other media. But the real challenge lies in making the receiving computer decode the incoming bursts of ones and zeros, and reconstruct the original image. There are as many electronic systems to send images as there are types of computers and hence the need for standardization [120].

In an effort to develop a standard means by which users of digital medical imaging equipment could interface display or other devices to these machines, the American College of Radiology (ACR) and the National Electrical Manufacturers Association (NEMA) formed a joint committee called the ACR-NEMA Digital Imaging and Communications Standards Committee in early 1983 [121][122][123].

The purposes of the initial standards activity were to:

- Promote communication of digital imaging information, regardless of source format or device manufacturer;
Facilitate the development and expansion of picture archiving and communication systems that can also interface with other systems of hospital information; and

Allow the creation of diagnostic information databases that can be interrogated by a wide variety of devices distributed geographically.

Version 1.0 of the ACR-NEMA standard was published in 1985. It was followed by version 2.0 in 1988 and the final version of version 3.0 also called DICOM 3.0 (Digital Imaging and Communications in Medicine) was published in 1993.

4.2.1 Version 1.0:

This version specifies data formatting and provides a data dictionary, a set of commands and hardware interface. It supports only point-to-point message transmission. Connection to a network requires additional hardware and software. It provides for a variety of functions such as data integrity checking, media access, flow control, fragmenting and reconstruction [121].

The data dictionary is a comprehensive table of rules for the encoding of information associated with images; such information accompanies the image pixel data in a file area called the header. They can contain such information as: patient name, patient identification number, study date, etc.

ACR-NEMA is patterned after the ISO/OSI Reference Model. However, it is a six-layered protocol since the Transport and the Network Layers were consolidated. The Transport/Network Layer, Data Link Layer, and Physical Layer all follow very specific protocols.
Application Layer:

An application can issue commands like SEND, GET, MOVE, FIND, DIALOG, ECHO, and CANCEL. Each command has a request form and a response form. A protocol for message exchanges is defined. The success or failure of a command request message is indicated by the value of a status element within the command response message.

Presentation Layer:

The protocols for the presentation and application layers were not as rigorously defined. The data dictionary is a part of the presentation layer. It defines the structure for packaging the various parts of data which cross the interface.

Session Layer:

This layer will establish and control an end to end connection. Additionally, it will provide its logical address as well as the network address.

Transport/Network Layer:

The Transport/Network layer specified by the ACR-NEMA standard is a simple X.25 style network layer. Its function is to support and maintain virtual channels for message packets and to provide flow control for those channels.

The ACR-NEMA standard does not specify an interconnection approach for multiple devices. The users are free to use any interface of their choice[118]. The network interface unit will provide a gateway to the network. Routing is provided by making the use of virtual channels and end-to-end session protocol. It will fragment images into 2K 16 bit packets, with a two-word packet header.
Data Link Layer:

This layer supports data flow control of two modes: datagram with no acknowledgements and stop-and-wait with an acknowledgement per-frame. Data from the transport/network layer is framed with control and error checking headers.

Physical Layer:

The physical layer specifies a 50-pin hardware interface for point-to-point communication. It specifies with 15 meters of cable length and can achieve a maximum of 8 megabytes/s of 16 bit data words. A minimum set of commands will initiate transactions over the interface. The standard gives complete details of the physical layer, it specifies signal timings, state diagrams, differential circuits, bidirectional data and parity lines, unidirectional control lines and a maximum error rate of $17 \times 10^9$ [34].

4.2.2 Version 2.0:

This version like Version 1.0 supports only point-to-point message exchange. The data dictionary rules are identical to those of Version 1.0 and it uses substantially the same hardware specifications. A number of errors and inconsistencies existing in Version 1.0 were removed and some new data elements were added. The layer specifications were the same as for version 1.0. It was concluded that both the versions 1.0 and 2.0 were not capable of providing a workable foundation for the development of cost-effective and efficient digital radiology interfaces, especially in a networked PACS [124].
4.2.3 Version 3.0

This version is known as digital imaging and communications in medicine (DICOM) and was published in 1992 [125][126]. The draft document of the standard is composed of nine parts. DICOM is a standard for the communication of medical images and associated information. It differs from ACR-NEMA Versions 1.0 and 2.0 in several major respects. The most important is that the basic design of the standard was changed. The earlier versions relied on an implicit model of the information that is used in radiology departments. The data elements were grouped based on the experience of the designers, and though the mapping was imperfect, the message structure allowed the necessary information to be transmitted. In contrast, DICOM relies on explicit and detailed models of how the various elements (patients, images, reports etc) involved in radiology operations are described and how they are related. These models are called entity-relationship (E-R) models and facilitate the development of data-structures used in DICOM.

The DICOM standard conforms fully to the ISO reference model for network communications. Radiologic imaging software applications interact with the top layer of the ISO protocol stack, the Application Layer. Part 7 of DICOM is the document that defines the requirements of an application software to interact with DICOM communications. In DICOM, a typical message consists of a command stream and a data stream. This message needs to be passed on to lower layers of the communication model for communication to take place.

The DICOM standard represents a major conceptual departure from the limited point-to-point communications goal of versions 1.0 and 2.0. While maintaining compatibility with the 50-pin physical connector (described in part 9 of the DICOM document)
and the signaling protocol of the original standard, the DICOM standard specifies means of alternative connections that simplify multipoint networking. Part 8 of the DICOM document defines the network support for exchanging the DICOM messages. Currently, TCP/IP and the ISO-OSI protocols are supported, but the nature of the upper layer service defined is such that it should be possible to expand to other protocols with relative ease [127][128][129][130]. Once out of the DICOM upper layer, the remainder of the communications protocol follows the existing standards. DICOM does not modify or customize anything in these standards. The DICOM model is shown in figure 4.2

![Figure 4.2: The Dicom Model](image)

Also the earlier versions of the ACR-NEMA standard specified only a minimum level of conformance leading to a great deal of ambiguity. The DICOM standard defines
principles that implementations claiming conformance must follow. Conformance require-
ments specify the general requirements which must be addressed by all implementations
claiming conformance. Conformance claims must be expressed in terms of specifically de-

defined service classes, the functional units of which are precisely described with predefined
terms[131][132][133].

4.2.4 American Society of Testing and Materials (ASTM) Medical Stan-
dards:

The American Society for Testing and Materials (ASTM) is composed of more than 132
technical standards writing committees. Together, they have published more than 9,100
standard specifications, tests, practices, guides, and definitions for materials, products,
systems, and services. ASTM’s committee E-31 was started in 1970 to write standards for
computer automation of laboratory instruments [108]. In 1975, a subcommittee to deal
with clinical laboratory systems was formed, to deal with medical computer systems.

The primary goal of the sub-committee on data exchange for clinical results E-31.11
is to define specifications for transferring clinical laboratory data messages between inde-
dependent computer systems. In relationship to networking protocols this ASTM standard
deals with the application layer of the ISO/OSI model. The lower layers can be any set
of reliable protocols. The messages are composed of a restricted ASCII character set. The
message can consist of an unlimited number of lines but each line cannot be longer than
220 characters.
4.2.5 Medical Information Bus Standard IEEE P1073

In 1984, the P1073 committee of the Institute of Electrical and Electronic Engineers (IEEE) Engineering in Medicine and Biology Society (EMBS) started developing the Medical Information Bus (MIB) standard [108]. The primary goal of this committee was to "provide an international standard for open systems communication in healthcare applications, primarily between bedside medical devices and clinical information systems, optimized for the acute care setting".

In relationship to networking protocols the MIB standard specifies a specialized local area network through a family of three network communication standards. These standards were modeled after the ISO/OSI reference model. The presentation layer is a subset of ISO 8822/23 standard, the session layer is a subset of ISO 8326 standard. The Transport layer provides simple transport services, but does not strictly comply with it as many parameters are defaulted. The network layer is a functionality inactive layer, while the data link layer offers a connection oriented, point-to-point data transfer. The physical layer uses the Electronic Industries Association (EIA) 485 specification with non return to zero inverted (NRZI) encoding at 375 Kbps.

4.2.6 Health Level 7 Standard

HL7 is a standard for electronic data exchange in healthcare environments. HL7's purpose is to facilitate communication in healthcare settings. The primary goal is to provide standards for the exchange of data among healthcare computer applications that eliminate or substantially reduce the custom interface programming and program maintenance that may otherwise be required. HL7 is accredited by the American National Standards Institute...
(ANSI) to write these standards [133].

The Standard currently addresses the interfaces among various systems that send or receive patient admissions/registration, discharge or transfer data, queries, orders, results, clinical observations, billing, and master file update information. The next version of the standard (2.3) will expand on the current coverage of these areas and will include new coverage for patient care, medical records, and automated instruments. Work is also underway to produce HL7 standards for recording immunizations and drug reactions.

The term “Level 7” refers to the highest level of the Open System Interconnection (OSI) model of the International Standards Organization (ISO). In the OSI conceptual model, the communications functions are separated into seven levels. Those developing the HL7 Standards are primarily focused on the issues that occur within the seventh, or application, level. These are the definitions of the application data to be exchanged, the timing of the exchanges, and the communication of certain application specific errors between the applications.

4.2.7 Medical Data Interchange Standard IEEE P1157:

The P1157 Working group of the IEEE EMBS started developing the Medical Data Interchange (MEDIX) standard with a primary goal of specifying and establishing a robust and flexible communication standard for the exchange of data between heterogeneous healthcare information systems. The MEDIX standard is based upon the ISO/OSI reference model [134]. Currently, it is based on the Government OSI profile (GOSIP).

A list of the standards and the type of information they deal with is given in table 3.
<table>
<thead>
<tr>
<th>Subject matter and kind of communication</th>
<th>Acronym/number</th>
<th>Responsible organization/name of standard</th>
</tr>
</thead>
<tbody>
<tr>
<td>Clinical alpha-numeric information within an institution (Linkages assumed to be “tight, synchronous”)</td>
<td>HL 7</td>
<td>Health Level 7</td>
</tr>
<tr>
<td>Clinical alpha-numeric information between institutions (Linkages assumed to be “loose, intermittent, non-synchronous”)</td>
<td>ASTM E1238</td>
<td>American Society for Testing and Materials Clinical Data Interchange Standard</td>
</tr>
<tr>
<td>Medical images in all contexts, between all picture archiving systems</td>
<td>ACR-NEMA DICOM 3.0</td>
<td>American College of Radiology-National Electrical Manufacturers' Association Digital imaging and communications in medicine</td>
</tr>
<tr>
<td>Information from laboratory instruments to computer systems</td>
<td>ASTM 1394 IEEE P1073</td>
<td>American Society for Testing and Materials Institute of Electrical and Electronics Engineers Medical information bus (MIB) standard</td>
</tr>
<tr>
<td>Electrocardiographic signals between ECG devices and computers</td>
<td>CEN TCPT 251</td>
<td>European Technical Committee for Normalization</td>
</tr>
<tr>
<td>Electroneurographic signals (EEG, EMG) between computers</td>
<td>ASTM 1467</td>
<td>American Society for Testing and Materials Standard specification for transferring digital neurophysiological data between computer systems</td>
</tr>
</tbody>
</table>

Table 4.1: Standards for Transferring Messages and Data.
Continuation of table 3

<table>
<thead>
<tr>
<th>Subject matter and kind of communication</th>
<th>Acronym/number</th>
<th>Responsible organization/name of standard</th>
</tr>
</thead>
<tbody>
<tr>
<td>Clinical medical logic, alerts, decision support prompts, guidelines</td>
<td>ASTM E1460</td>
<td>American Society for Testing and Materials Standard specification for defining and sharing modular health knowledge databases (Arden syntax)</td>
</tr>
<tr>
<td>Messages from application to bibliographic retrieval systems</td>
<td>ANSI Z39.50</td>
<td>American National Standards Institute</td>
</tr>
<tr>
<td>Billing and remittance transactions between care site and payer</td>
<td>ASC X12</td>
<td>Accredited Standards Committee X12 Subcommittee standards</td>
</tr>
<tr>
<td>Billing and eligibility information between pharmacies and payers</td>
<td>NCPDP</td>
<td>National Council for Prescription Drug Programs NCPDP standards</td>
</tr>
</tbody>
</table>

4.3 Medical Image Management:

A hospital is a very large environment, where objects (mostly images) have to travel around fast. Sometimes images even have to travel outside the hospital for inter-hospital communication or communication with a referring physician outside the hospital. Although high speed networks are available now, extra measures have to be taken to guarantee the best performance [135][136]. Management of voluminous medical images and related text/graphical data is still a key issue and involves such concepts as database design, image routing, image stacking, image aging, HIS-RIS/PACS interfacing, automatic file transfer and image prefetching.
4.3.1 Database Storage Issues:

The amount of storage required varies widely depending on what type of imaging modalities are available. The number of bytes required per study depends on the number of pixels per image array, the number of bits per pixel and the number of images per study. The image parameters of various imaging modalities are shown in table 4.

<table>
<thead>
<tr>
<th>Modality</th>
<th>Resolution (bits/image)</th>
<th>Typical study size</th>
</tr>
</thead>
<tbody>
<tr>
<td>Digitized x-ray film</td>
<td>4096X4096X12</td>
<td>6-34 Mbyte</td>
</tr>
<tr>
<td>CR</td>
<td>2048X2048X10-12</td>
<td>8 Mbyte/image; 2-3 images/study</td>
</tr>
<tr>
<td>CT</td>
<td>256X256X8 to 512X512X10</td>
<td>20-100 images = 10-100 Mbyte</td>
</tr>
<tr>
<td>MRI</td>
<td>256X256X16</td>
<td>2-50 Mbyte</td>
</tr>
<tr>
<td>Nuclear medicine</td>
<td>64X64X8 to 256X256X8</td>
<td>0.2-2 Mbyte</td>
</tr>
<tr>
<td>Ultrasound</td>
<td>512X512X6</td>
<td>36 still frames on 4 films=7 Mbyte</td>
</tr>
<tr>
<td>Digital angiography</td>
<td>1024X1024X8 to 2048X2048X12</td>
<td>80 images (10 second animation)=80 Mbyte</td>
</tr>
</tbody>
</table>

Table 4.2: Image Parameters

At present, storage is done mostly on two types of media: magnetic for short-term storage and optical for long-term storage [137][138]. For short-term storage magnetic disks are used because they offer speed, limited portability and a great deal of dependability that is not available in other storage media. The more commonly used magnetic disks are the large parallel disks and also the Redundant Array of Inexpensive Disks (RAIDs) technology used in MDIS systems. For long term storage, optical disks which are based on the WORM or Write Once Read Many times technology offer lower cost per byte storage than magnetic disks at very reasonable transfer rates. Also hundreds of optical disks can be stored in jukeboxes providing terabytes of storage.
4.3.2 Database Design:

Two different storage environments are possible: a central database and a distributed database [138][139]. In the central database design there is a single database which contains all the data generated by the various modalities. In this architecture, image access requests generated from remote locations are routed to and processed at the central site, where communication, disk-accessing and processing bottleneck often occur. This limitation of a centralized architecture results in slow system response time, giving rise to the consideration of an alternative distributed database design. Distributed databases are characterized by physical separation/replication as well as logical integration of database fragments. The principle of a distributed database design is to distribute the logical and physical components of a single database over a communication network while keeping the distribution hidden from the users. A distributed database design has been shown to better meet the various performance requirements of a PACS than a central database design in the clinical environment [140][141]. In a large scale PACS, the distributed architecture which allows parallel retrieval of images for a transaction. This will remove the communication bottlenecks encountered due to the differences in speed between the data storage systems and the data communication systems (high speed). A distributed design will allow multimedia data to be stored in different physical media to enable parallel data retrievals, thereby reducing retrieval response time [141]. In addition, a distributed design is also more dependable in that it can continue information service even if individual sites or parts of the communication links fail.
4.3.3 Automatic File Transfer and Image Prefetching:

As the PACS network grows and the number of archive and display nodes increases, special software is needed to efficiently manage automatic transfer of image files between different servers. Images must be sent from the acquisition/archive node to the display servers where they are needed. Also, images from previous examinations must be sent to the same servers for comparison. In the reporting activity, radiologists usually compare newly acquired images with images from previous examination of the same patient according to certain rules and needs. In a film-based system, the old images present in a film jacket are physically available to the reader. In a PACS, in order to minimize the traffic workload at peak hours, older images required for comparison should be sent in advance, possibly during night (overnight prefetch) time when the traffic is low. This technique is called "prefetching" [142] of comparative images and is possible only if a proper HIS-RIS/PACS interface is available. The simplest way to implement a prefetching system is to always send the full set of older images of a given patient to the display server considered. Such a scheme is relatively easy to implement but is not efficient. A more selective extraction of relevant images based on a set of rules designed by the clinical users depending on their different needs should be designed. Sophisticated prefetch algorithms have been designed and are being employed in most of the existing PACS Systems [143].

4.3.4 Image Compression Issues:

The large amount of information that must be stored for medical imaging purposes makes it necessary to employ compress the array representing the original image into an intermediate set of data having fewer bits than the original array. This reduces the amount of
storage required. Image compression algorithms are divided into lossy and lossless algorithms [144][145]. Lossy algorithms are capable of upto 20:1 or 30:1 ratios. Most lossless algorithms can only compress images upto 3:1 ratios. Most primary diagnosis is done using images reconstructed with lossless algorithms [146]. Lossy algorithms are only used for consultative or browsing (for quick identification of images for prefetch) purposes.

The images that are presented to the radiologists must preserve the information in the original film images in order for the soft-copy images to be accepted by the medical community. Radiological images were found to differ from other images in the following ways:

- The images are very low contrast images locally. Even the transition over the edge of a bone does not change the image intensity by a large amount. The dynamic range over the whole image may be quite large.

- Much of the information is in the local variation of the image intensity. The overall picture provides a context within which the information is embedded, but the pathology is visible as local variations.

- The images are subject to large changes in the individual pixel intensity as the radiologist views the image. The radiologist may change the contrast and the center of the contrast range in his efforts to see the details of the image. In the extreme the radiologist may invert the image changing white to black.

- The images are very large, typically more than $1k \times 1.2k$ pixels and up to $2k \times 2.5k$ pixels.

- The image dynamic range typically requires 12 bits.
• The image will be viewed at a number of magnifications from 4:1 minimized to 8:1 zoomed.

All of the images in medicine can be usefully compressed to reduce the storage and communication capacity required. The interchange of compressed images requires that the receiver be able to accept an image in the compressed format and restore the image to an approximation of the original image in a diagnostically useful form. Interoperability requires that the compressed images be in a standard format that the receiving system can interpret to restore the image.

JPEG (Joint Photographic Experts Group) compression [146] is the generally accepted compression for a wide variety of images in publishing, illustrations, and advertising. Since it has such broad acceptance, it has been included in the DICOM standard as an instance of a standard compression technique. For medical imaging the basic JPEG compression has drawbacks.

JPEG has a total of 29 variations ranging from lossless compression to lossy compression. Of these, the best known compression techniques are differential pulse code modulation (DPCM) processing with a Huffman coder for lossless compression and a block discrete cosine transform (DCT) with a Huffman coder for the lossy compression.

For lossless compression, JPEG can provide a compression ratio of 2:1 or 3:1. The drawbacks of the simple JPEG lossless compression are the Huffman coding table and the lack of a low entropy extension. While a fixed Huffman coding table may be used for many images, in general the Huffman coding table is constructed for a single monochrome image by examining each pixel of the entire image and building the table from the frequency of occurrence of pixel values. When there is only one type of image, a single Huffman coding
table can be built to do well on most of the images of that type. When there are several

types of images, a separate Huffman coder is required for each type or each image must be

examined separately to build the Huffman coder.

For lossy compression, the JPEG standard uses an $8 \times 8$ DCT which introduces

blocking artifacts that are unacceptable in most types of medical images. When the image

is viewed on a workstation, the image may be magnified for looking at the details, and the

blocking of the $8 \times 8$ DCT will be visible over the whole of the image, this is unacceptable.

For teleconferencing purposes it is necessary to compress full motion video. The

MPEG-1 and MPEG-2 [147] standards define methods for compressing high quality audio

and video. The MPEG algorithm is widely used for video compression, it relies on two basic

techniques; block-based motion compensation for the reduction of temporal redundancy;

and DCT based compression for the reduction of spatial redundancy. The image quality of

the MPEG-1 algorithm at rates of about 1.2 Mbps is limited to 360 samples per video line

and the video signal at the input of the source coder is 30 frames per second, progressive.

MPEG-2 supports compression of wide ranging quality video and audio beyond that of High

Definition Television (HDTV) into a bitstream up to 100 Mbps.

\section*{4.4 Classification of PACS:}

\subsection*{4.4.1 Modality Centered PACS:}

This is a small system with only a few (2-10) workstations. It is focused on one specific

modality, typically MRI, CT, NM, or US [148][149].
4.4.2 Departmental PACS:

This is used for image capture, diagnosis and reporting within a single department, such as radiology or Nuclear Medicine. Typically from 2 to 10 workstations are employed. Several departmental PACS are now commercially available. these can be integrated into an existing PACS.

4.4.3 Full Hospital PACS:

This can be considered as a Local PACS. It includes a network connecting multiple imaging modalities to display workstations throughout the hospital serving all departments which work with patient images and reports. The present generation Local PACS also includes a telemedicine system, which is a long distance network connecting healthcare facilities and/or practitioners, which includes teleradiology, telepathology, video-conferencing/consulting, and other information transfers electronically in real-time.

A Local PACS requires a relatively larger capital investment, but provides economies of scale and capabilities that go far beyond a simple enlargement of a modality centered PACS or a departmental PACS. With the smaller PACS versions, the hospital still has to rely on film. It must run both types of systems, and has to deal with the complexities of both approaches, therefore, not realizing the full benefits of PACS [150].

4.4.4 Global PACS:

A Global PACS consists of several Local PACS networks interconnected by a national or international backbone network, called the Internet [151]. Digital images and patient information are transferred between Global PACS sites using this backbone network. A
Global PACS system can interconnect many Local PACS across country, as shown in figure 4.3 [152].

Figure 4.3: A Global PACS Architecture

Each Local PACS has viewing workstations and database archive systems containing radiology modality image sets. The system software for the Global PACS is distributed among Imaging Equipment, Viewing Workstations, Database Archive System, and the net-
work interface nodes. A Global PACS environment provides new and beneficial operations between radiologists and physicians, when they are located in different geographical locations.

4.5 PACS Design Concepts:

Many technical problems make a total PACS solution impractical in many situations. These problems cover every aspect of PACS and include cost of equipment; inadequate or non-existent equipment interfaces; limited network bandwidth for image transfer; the need for large image archives with usable image management; and displays with acceptable resolution, speed, contrast range, and user interfaces.

Small clinically useful partial PACS ("miniPACS") have been developed[119] and these can provide experience for future development of more complete PACS. A partial PACS may have a component missing, but serves a limited clinical or functional role. A partial PACS is typically implemented (a) to solve a specific storage problem; (b) to allow archiving, communication and display of organ-specific systems, as in neuroradiology, which may include multiple modalities; (c) to provide PACS for a single modality system such as nuclear medicine image transmission, display, and processing. But without substantial work on the part of future users of the system, a partial PACS can be expensive and may not meet the basic requirements that would make the system clinically useful.

Planning for PACS and creating a supporting infrastructure are important and complex procedures [153][154]. Because of the complexity of a full scale PACS, simulating various hypothetical PACS scenarios before deciding on an overall architecture and implementing it is valuable. Modeling and simulation are used for capacity planning, for the
analysis of various tradeoffs and "what if" scenarios, and in the evaluation of alternative architectures [153][155]. One of the major steps in this process is the definition of system requirements and the establishment of measurable performance metrics associated with these requirements. PACS system requirements can be defined as follows [154,155]:

- **Usability**: Radiologists usually evaluate PACS system in binary terms: either the system is usable or is not usable. This is called subjective usability and can vary drastically among individual radiologists. Objective usability is a measure from the developers perspective and is used for system analysis and design and is associated with a measurable system metric which can be derived from other system metrics.

- **Reliability**: The reliability of a distributed system such as PACS depends on such factors as software quality, interdependency among components in the system, system monitoring facility, and fault resiliency mechanism. These metrics can be measured qualitatively.

- **Fault resiliency**: System fault resiliency can be implemented and measured in terms of the degree of component redundancy, automated failover and system degradation mechanisms, and automatic system reconfiguration upon failures.

- **Efficiency**: The performance and efficiency of a PACS system can be measured with these metrics: the rate to move data from optical storage media to magnetic media, the rate to move data from disk to memory, data network throughput between nodes, the speed for remote access of other medical data, the efficiency to search and sort various medical textual data, the speed of the image processing functions performed, and the speed to present the integrated information to radiologist.
• Image data locality and availability: The degree of image data locality is a measure that indicates the availability of image data on the local display workstations before a radiological diagnostic session. Intelligent image prefetching and optimal data replication can be implemented to provide the image data locality and functionality. General image data availability concerns the integrity and retrievability of the image data stored in long-term storage media such as optical and tape media. It can be measured as the success rate of retrieving a statistically significant sample of image data stored.

• Automation: Any system operation depends on human interaction is bound to errors. The degree of automation can be measured in terms of the percentage of the system that operates automatically without any human intervention.

• Operation optimization: Operation optimization of a PACS can be implemented by using such techniques as dynamic network load balancing, distributed system message delivery system for system-wide status collection and monitoring.

• Connectivity: Connectivity of a PACS indicates to what degree of remote services that the PACS provides. A PACS with LAN, MAN, and WAN capability has a higher degree of connectivity than that of a PACS which can only serve one or a few limited radiological services in a LAN environment. Likewise, higher degree of connectivity is provided if the PACS has the ability to access remote medical and other information sources.

• Scalability: High scalability of a PACS ensures that the addition of new imaging devices and/or new image display workstations to the PACS requires no or minimal
system level adjustment.

- Security: Implementing authorization and authentication mechanisms to control and monitor user access to and user utilization of the PACS system is an important factor of PACS system security.

- Conformance to industry standards: Conformance profiles of the PACS components should be used to evaluate the conformance to industry standards.

- Manageability: System management includes the management of software, hardware configuration, and other resources. The ease of management can be measured by the manpower required to maintain and manage a PACS

After the performance metrics have been defined and the measurements made, the simulation model can be created and executed using one of the many commercially available simulation tools. The results should then be analysed and the model verified before transforming the PACS model into practical implementation [156][157][158][159]. The design of the PACS architecture should be based on the ISO/OSI model to facilitate the interoperability with other systems. Table 5 gives the ISO/OSI communication protocol layers and the standards by which these layers can be implemented.

The feasibility of operation of DICOM over ATM has been demonstrated. A teleradiology model using DICOM and TCP/IP over ATM is shown in figure 4.4.

4.6 Reliability Issues in PACS:

Reliability is an increasing concern when PACS modules are moved from the controlled research environment to a busy clinical setting where radiologists and referring physicians

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<table>
<thead>
<tr>
<th>OSI layer</th>
<th>Event (ISO Standards)</th>
<th>Competing standards by which layers usually are implemented</th>
</tr>
</thead>
</table>
| 7         | Application           | American Society for Testing and Materials ASTM 1238  
|           |                       | Health level 7 (HL 7)  
|           |                       | Institute of Electrical and Electronics Engineers IEEE P1157  
|           |                       | Medical Data Interchange Standard (MEDIX) |
| 6         | Presentation (encoding) | Abstract Syntax Notation.1 (ISO ASN.1)  
|           |                       | Accredited Standards Committee X-12  
|           |                       | Electronic Data Interchange (EDI)  
|           |                       | Health Level 7 (HL 7)  
|           |                       | Standard Generalized Markup Language (SGML) |
| 5         | Session               | Berkeley Sockets |
| 4         | Transport             | NetBIOS, Sockets, SPX  
|           |                       | Transmission Control Protocol-Internet Protocol (TCP-IP) |
| 3         | Network               | Ethernet, Internet Protocol (IP), NetBEUI, X.25  
| 2         | Data link             | Ethernet, HDLC, LAPB, LAPD, LLC |
| 1         | Physical              | Ethernet, IEEE RS-232, IEEE 802.x, Token Ring, ATM |

Table 4.3: OSI Communication Protocol Layers.
will depend heavily on its operation [160][161][162]. Any system downtime may seriously affect patient care. From a user's point of view, reliability can be measured in terms of the speed at which the task is performed and the accuracy to which a task is carried out. A reliable system will consistently get images, reports, and necessary support information to the appropriate workstation within a specified period of time. This time constraint is critical since the longer it takes a radiologist to view and interpret a study the less chance the results of the study can be used for patient management decisions. A reliable system also accurately performs requested tasks [163][164].

It is difficult to attain a completely fault-tolerant PACS. Most PACS, in order to be cost-effective employ major pre-existing subsystems (workstations, Digital Signal Processors, microprocessors, displays, mini-PACS, etc.) that have no fault tolerance, or have it to a very limited extent only. The major points of failure in any PACS are:

- workstations,
- network links,
- image database systems,
- acquisition devices, and
- data interfaces.

Very high dependability can be attained by employing both fault-tolerance and fault-avoidance (testing, design verification, quality control, etc.) techniques. A fault-tolerant system can detect faults when they occur and restore correct computation by involving redundant hardware or software to re-establish error free operation.
Hardware redundancy can be introduced easily for all of the points of failure identified above. Most of the large scale PACS employ a multi-tier network architecture [50][57](for example ethernet, FDDI, and ATM). Failure of a particular network automatically triggers an attempt to send data over a different network circuit. The reconfiguration involves simply passing the communication routine a different network address.

The software is susceptible to failure too. To increase software reliability, techniques of recovery blocks and design diversity can be applied to protect critically important software elements. If software failures are detected, it is necessary to have backups that can be expected to work. This can be done by supplying less complex or well de-bugged earlier versions of programs that can be substituted if a program fails [160].

Providing mechanisms to detect and isolate errors and faults when they occur is essential in the design of a highly dependable system [71,72]. Once the errors have been detected, it is necessary to recover from them. Hence providing fault and error recovery mechanisms that can monitor the PACS system for proper operation, and upon detection of faults can properly take actions to prevent data loss, preserve data correctness and reassign functions to redundant hardware and software to allow the system to continue correct operation is necessary.

To ensure data integrity and security, techniques like key-based cryptography [165] and digital time stamping should be employed. Key-based cryptography associates the content of an image with the originator using one or two distinct keys and prevents alteration of the document by anyone other than the originator. A digital time stamping algorithm generates a characteristic “finger print” for the original document using a mathematical hash function, and checks that it has not been modified.
4.7 Some existing PACS:

Several medical communication networks have been developed at several university medical centers. These include University of California at Los Angeles, Bowman Gray School of Medicine at Wake Forest University, University of California San Francisco School of Medicine, Osaka University School of Medicine in Japan, University of Arizona, University of Miami School of Medicine, Brussels Free University Hospital in Belgium, Phillips University Hospital in Germany, Mallinckrodt Institute of Radiology at Washington University, University of Washington at Seattle, University of Alberta in Canada, Aachen University of Technology in Germany, Columbia University and several others. Also the U.S. military through the Medical Diagnostic Imaging Support (MDIS) system is installing teleradiology at multiple medical treatment facilities throughout the US and abroad.

Some of the medical centers mentioned above have used proprietary networks like UltraNet, ImNet and TeraNet. UltraNet was used in several medical centers including University of California, Los Angeles. It is a fiber optic-based LAN with a high throughput rate. It consists of two major components: a hub with a 1 Gbps signaling rate, and a host adapter with a signaling rate of up to 250 Mbps. The UltraNet network is a mesh star topology. Even though it can provide high transmission rates, because of UltraNet's medium access control (MAC) protocol being proprietary, it is not widely used [39].

ImNet/2 (the latest version of ImNet) is a network specialized for the image transport in a PACS. It has been developed by the PACS research lab at the Aachen University of Technology, the main principle here is the separation of the heterogeneous data flow into pure image data flows and management data flows. ImNet/2 is a fiber-optic-based network and has a data transport mechanism which permits asynchronous data transfer between two
nodes of image equipment without packeting the image data set into small frames. Data rates of upto 140 Mbps were achieved [166]. Again the technology is not standardized and is limited to a few hospitals.

TeraNet is an integrated network designed to provide user access rates as high as 1 Gbps. It is based on Wavelength Division Multiple Access (WDMA) and Sub-carrier Frequency Division Multiple Access (SFDMA). Each user is connected to a network access node, which has a small number of transmitters and receivers, each operating on a pre-assigned WDMA/SFDMA channel. TeraNet was developed at Center for Telecommunication Research, Columbia University [53] and some of concepts like negotiating the service requirements before connection setup, supporting multiple classes of traffic are similar to ATM.

4.7.1 Telemedicine Protocols:

A number of standards have been developed which can be used when design a telemedicine system. Each of these standards apply to one or more layers of the typical protocol stack as shown in figure 4.5. The H.320 family of international teleconferencing standards provides for simultaneous audio (G.700), video (H.261) and data transfer (T.120) using communication bit rates from 56 kbps to 1.92 Mbps. Compatibility with H.320 insures interoperability with the widest range of third-party teleconferencing systems. The MPEG-1 and MPEG-2 standards define methods for compressing high quality audio and video. The Internet family of protocols forms the common language for the global packet-switched network known as the Internet. IP is inherently connectionless-based. TCP supports error-correction, packet ordering and acknowledgements. User Datagram Protocol (UDP) provides minimal overhead but without any error-correction. Point-to-Point Protocol (PPP) supports the layering
Figure 4.5: Telemedicine Protocol Stack

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of IP and other packet-switched protocols on connection-oriented bitstreams.
Chapter 5

Conclusions

This chapter presents the conclusions of this survey. Section 5.1 lists the observations made, and describes the reasons for ATM to be used in developing a PACS. Migration to ATM is complicated by the limitations of existing PACS, section 5.2 describes the differences between the existing LAN technologies and ATM, and discusses the two standard techniques for providing LAN services over ATM. Section 5.3 describes the various issues to be resolved for rapid deployment of ATM into the medical environment.

5.1 Observations and Recommendations:

Upon deep perusal of the available literature, it has been observed that most of the existing medical communication systems have been designed to provide the basic communication needs for PACS in the least expensive way. Such an approach, though successful in meeting the immediate requirements, will be inadequate in satisfying the growing communication needs of increasingly image intensive medical applications. For example 3D therapy in cancer treatment generates an image data set containing 50 to 100 slices of 512 x 512 images, each of which has 12 bits of gray scale. Thus on an average each data set is approximately
236 Mbits. Also for interactive 3D volumetric rendering calculations, the physicians and physicists need to receive $512 \times 512 \times 24$ (64 Mbit) images at the frequency of at least 50 times per second. The bandwidth required is at least 300 Mbps, which is higher than even OC-3 rates.

In the past, digital video conferencing systems were designed to work over low bandwidth links. During the development of the North Carolina Information Highway (NCIH), it has been shown that for providing realistic tele-consulting and education environments, multiple cameras were required to cover even a single office. Similarly, mono-phonic audio was found to be inadequate to give a feel of the location of the audio source, and hence multiple microphone locations had to be considered [NCIH] [59][166][167]. Also for effective remote consultation and diagnosis, graphical annotations (like circles or boxes drawn by the radiologists/physician to mark areas of interest) of medical images have to be transmitted in real time [168][169][170]. Every node in the network should be able to support full-duplex traffic at rates higher than 100 Mbps. To support hundreds or even thousands of such nodes the aggregate network capacity must be in the order of tens of giga bytes. To support this magnitude of growth potential over a long term period the network must be scalable, modular and flexible.

Neuroradiology is a subspeciality in the field of radiology. Only about 1,600 senior members of the American Society of Neuroradiology are experts in interpreting CT and MR images. The scarcity of these experts hinders delivery of health care, most of them now use teleradiology to interpret CT and MR images on an emergency basis. But their success has been limited by the lack of high performance networks and associated PACS technologies [61].
With the increasing use, as well as the lowering cost, of networking and imaging technologies, a trend is emerging to integrate local PACS and textual databases across many hospital sections and departments for hospital wide service needs. Local area networks allow imaging and other equipment to be intraconnected within hospitals and clinics.

But cost control measures, such as managed health care and patient demands for convenient access to out-patient care is leading to an increasing shift away from single institution in-patient care towards out-patient care provided by consortia of health-care providers. Wide area networks allow hospitals, out-patient clinics, health maintenance organizations (HMOs), and physician offices located throughout urban, sub-urban, and rural areas to be interconnected. The lack of appropriate networking technology to integrate individual systems, and their services seamlessly is a major stumbling block.

A complete PACS requires LAN access with high bandwidth and low delay over long distances. To satisfy these needs, several high-speed networking techniques have been developed recently, offering data rates up to hundreds of Mbps, typical examples are FDDI, Frame Relay, Fast Ethernet, Ether Switch and, Asynchronous Transfer Mode (ATM). All of which are designed to satisfy connectivity requirements emerging from current LAN, as well as high-speed work stations in order to run powerful distributed multimedia applications.

ATM is superior compared to other networking technologies, as it offers high bandwidth, and it is scalable in the sense that the bandwidth capacity of an ATM system is not fundamentally limited to the technology itself. ATM can support multimedia traffic offering seamless integration with wide area ATM networks, both public and private. Multistar or mesh-connected switched networks based on B-ISDN/ATM technology is not only capable of providing large scalable bandwidth at reasonable cost, but will also facilitate easy and
smooth integration with public networks for universal access to various remote medical facilities. Since ATM is emerging as an international standard, eventually it will be widely available and it will be supported by the public telecommunication facilities. ATM LANs can support twenty times the peak bandwidth of FDDI. With the growing imaging modalities and increasing number of images per exam, FDDI, T1, Ethernet-based PACS will soon reach their maximum capacity. ATM/SONET interfaces currently available range from 45 Mbps to 622 Mbps, interfaces at 2.5 Gbps will soon be available. Although Frame Relay, SMDS provide capabilities similar to ATM, they are not widely accepted as international standards, and most phone companies do not provide these services. Thus, an ATM-based broadband network will be a cost-effective solution for providing scalable, modular and flexible high-speed LAN and MAN services. Several universities and hospitals have developed PACS based on ATM [57][59][60][61]. Though all of these are prototypes, the experience they have gained is valuable in developing future PACS based on ATM.

It is interesting to note that, although most of the places have used different networking environments and services, there are agreements in some basic communication requirements. During the design, development or trial phase of the medical communication systems mentioned above, several limitations were identified and important observations were made. These can be summarized as follows:

- To achieve the goal of a totally digital radiology system and other PACS objectives mentioned in the previous sections, and to overcome the drawbacks of the present communication systems, an Asynchronous Transfer Mode (ATM)/Synchronous Optical Network (SONET) based Broadband Integrated Services Digital Network (B-ISDN) compatible architectures should be considered since they can easily take advantage
of the evolving public high-speed telecommunication network services. By adapting
an international standard, smooth integration with other compatible systems will be
possible. This will also eliminate the need for any packet format or protocol conversion
which can prove to be a bottleneck in high-speed networking environments.

- Image compression board can also be a bottleneck since they are significantly slower
than the available transmission speeds. Moreover, lossy image compression is not
trusted in the medical community. Only about 3:1 compression ratio can be obtained
if lossless compression algorithms are used. Thus, focus should be on providing high
bandwidth networks rather than to image compressions for reducing the bandwidth
requirements.

- T1 lines connecting high-speed servers can be a bottleneck. More flexible and scalable
connections like switched, on-demand virtual circuits should be explored.

- Non-uniform design of PACS at different departments of a medical complex will dis-
allow smooth integration of departmental PACS into Global PACS. The design phase
should take into consideration all the departments that can potentially benefit from
integrated Global PACS.

- Delay between the display of two successive images should not exceed 0.5 seconds.
Adaptive routing methods and intelligent flow and admission control schemes are to
be developed to guarantee the desired grade of service.

- For better management, low response time and good image quality, a distributed
architecture for image storage databases should be considered.

- Display workstations that can handle ATM should be developed. A prototype de-
signed at the Washington University as part of their "Project Zeus" is shown in figure
5.1 [57].

5.1.1 Migration to ATM:

ATM technology can provide the high bandwidth and scalability required for PACS network
traffic; however, migration to ATM is complicated by the limitations of existing PACS.
The steps involved in developing effective migration strategies for incorporating ATM into
existing PACS are:

• Characterization of traffic flow.

• Development of network delay model.

• Design Optimization.

• Estimates of future traffic.

The present large-scale PACS are based on a distributed architecture, several com-
mercially available sub-systems and departmental "mini-PACS" are integrated into the
system. Multiple networks are used to support the flow of image data traffic within the
PACS. Generally, image flow in the network is from the imaging modality devices to the
display workstations and from the image archive to the display workstation. The delays in
the image flow were found to be because of:

• The network transfer delay from the scanner to the acquisition computer.

• The reformatting and decompression of the images at the acquisition computer.

• The network transfer delay from the acquisition computer to the image server.
Figure 5.1: A PACS Workstation.
• The network transfer delay from the image server to the display workstation.

• Queuing delays.

• The access delay in retrieving the images from the optical juke box.

It was shown that the most significant delays in the system were due to acquisition and archive retrieval processes [154]. It was found that the various imaging modalities and sub-systems were utilizing a very low percentage of the available ethernet bandwidth [60]. Hence simply using ATM to increase network bandwidth will not necessarily increase image throughput. Commercially available, intelligent ethernet and FDDI hubs should be used in combination with ATM switches.

The hubs can effectively segment existing networks into many segments and each segment can be connected to one port of an ethernet switch. Ethernet switches are available which provide an aggregate capacity of 100 Mbps for communication with the nodes in the same segment [171]. Some hubs are equipped with an ATM interface that can connect to ATM switches.

The design of the network should be optimized for providing the minimal delay for the transfer of images. The architecture should also be cost-optimal, provide increased capacity, and must be scalable and interoperable. One possible design is to use the ATM switches connected to each other in an arbitrary topology forming a backbone network. The stations requiring high bandwidth can be connected directly to the backbone network (for example an image server, display workstations) through a high speed (OC-3) connection. The ethernet switches can also be provided with OC-3 connections to the backbone network thereby providing connections to the computers in the rest of the network. A PACS architecture based on ATM communication backbone is shown in figure 5.2 [60].
5.1.2 ATM LANs:

It has been identified that the local area network is a major bottleneck in the realisation of a complete PACS [27][60][97][98]. ATM can be deployed to alleviate congested servers and backbones. But care should be taken before migrating into an ATM environment. One cannot simply move from the existing infrastructure, discarding or modifying all existing equipment and systems is impractical and will prove to be very expensive. Effective migration strategies should be developed in order to integrate the existing technologies, products and systems with ATM.

The infrastructure for present generation PACS is built around "legacy LANs" such as Ethernet, Token Ring, FDDI etc.. To be successful in such an environment, an ATM network has to provide services that permit the reuse of the vast base of existing LAN applications by "ATM stations" (stations attached directly to an ATM switch) and allow an easy interworking with those stations attached to the legacy LANs.

Legacy LANs and ATM differ in the following ways:

- Legacy LANs are connectionless, whereas ATM networks natively support only connection oriented services.

- Legacy LANs use broadcast and multicast media, while ATM uses a nonbroadcast multiple access (NBMA) medium [93][94][96]. In the broadcast model, every message sent is available for reception by every station on the network segment. In the NBMA model, messages are visible only to the sender and the recipient (for point-to-point connections) or to the sender and the recipient list (for point-to-multipoint connections).
• In legacy LANs, segments are connected together with bridges and routers. ATM networks are joined together using switches.

• Legacy LANs use MAC addressing. ATM uses a telephone-number-like addressing scheme [30].

• In legacy LANs information is exchanged in Protocol Data Units (PDUs), which are frames (or messages) with variable sizes of up to 18 kbytes. ATM is a cell based technology with a fixed cell size of 53 bytes.

Most LAN equipment conforms to the IEEE 802 family of protocols. In this architecture, the data link layer is split into the logical link control (LLC) and medium access control (MAC) sublayers. The LLC sublayer offers a common interface to the network layer, while each different MAC protocol is specific to a particular LAN, e.g., CSMA/CD, Token Ring, Token Bus, etc. It is possible to interface ATM directly to the transport layer or the network layer of the OSI model. This offers efficiency by avoiding the unnecessary complexity of the data link layer. However, there are many network layer protocols, and each one would have to be interfaced to ATM separately. To offer general compatibility with the installed base of networks and protocols, regardless of the network layer and upper layer protocol stack, and to support transparent MAC bridging, an interface at the MAC sublayer is required. This will permit the huge legacy of existing LAN applications to migrate to the ATM environment without major upheaval.

At present there are two schemes to support connectionless service over ATM, namely LAN Emulation proposed by the ATM Forum and Classical IP and ARP over ATM proposed by ITU-T's Internet Engineering Task Force (IETF).
LAN Emulation:

The LAN Emulation (LE) service [94][96][97] is exclusively designed to support three configuration scenarios: ATM-ATM interworking figure 5.3 [97], ATM-LAN interworking figure 5.4, and LAN-LAN interconnection figure 5.5. Under this scheme, a new ATM MAC sublayer is needed beneath the LLC sublayer and gives the appearance of a virtual shared medium such as IEEE 802.x LAN.

As standardized by the ATM Forum, a LAN Emulation service consists of the following three components:

- **LAN Emulation Server (LES).** The LES provides a facility for registration and resolving a MAC address into an ATM address.

- **Broadcast/Unknown Server (BUS).** The BUS forwards multicast/broadcast frames and delivers unicast frames for an unregistered or address-unresolvable LAN host.

- **LAN Emulation Configuration Server (LECS).** The LECS locates the LES and obtains configuration information for each ATM segment.

The three components of LE service can be implemented on a single physical entity or distributed on several physical entities. A device attached directly to an ATM switch contains a LAN Emulation Client (LEC). The protocol layers of an LEC is shown in figure 5.6. The main function of the LEC is to communicate with remote LE service components (LES, BUS, and LECS). The LEC is identified by two addresses: a unique 6-byte MAC address and a 20-byte ATM address. For unicast traffic, the source LEC first sends an LE-ARP request to LES, asking for the ATM address of the destination LEC. With the LE-ARP reply from LES, the LEC establishes a direct VC connection to that ATM address, and the
Figure 5.3: ATM-ATM Interworking
Figure 5.4: ATM-LAN Interworking
Figure 5.5: LAN-LAN Interworking
data frame transfer commences. If the LES does not get the address of the destination LEC in its address table, the source LEC sends the data frame to the BUS, which broadcasts to all stations on the ATM network. For multicast/broadcast traffic, the source LEC sends the data frame to the BUS for broadcast to all attached stations. Only those LECs whose MAC address are part of this group MAC address take the data frame.

Within ATM stations, the LE service is provided by the LE layer. The LE layer shields the higher layer protocol stacks from the characteristics of the ATM network and gives them the illusion of being directly attached to a traditional LAN. The LE service provides functions related to initialization, registration, address resolution, and forwarding of unicast or multicast frames.

LAN Emulation is well suited for small networks, due to its protocol independence and high speed. Multiple emulated LANs can be interconnected through traditional bridging or routing techniques. But it would be impractical to interconnect multiple emulated LANs through a wide area ATM network by means of bridging as it will lead to performance limitations because of increasing broadcast traffic [96][97]. The LE architecture is
based on using a single BUS for the distribution of all broadcast and multicast frames this
leads to a single point of failure. A failure in the operation of the BUS will lead to inter­
ruption of the service for all clients belonging to an emulated LAN. Future work needs to
investigate possibilities for introducing redundancy and hence improve reliability in BUS
implementations. Also, the emulated LANs hide much of ATM's functionality from the end
host's higher layer protocols. This is the main reason for the standards bodies to specify
mechanisms for running IP and other layer 3 protocols directly over ATM. There is also
a need for a method to provide connectivity among members of different emulated LANs
without the need of intermediate components like bridges and routers.

Classical IP over ATM:

Classical IP over ATM (RFC 1577 of IETF) is a mechanism to run IP directly over ATM. In
this model shown in figure 5.7, users attach nodes to the 'cloud' at UNI points. Signaling
services at the UNI [27][30] enable users to establish Switched Virtual Channels(SVCs)[98]
to nodes attached at other UNI points on the same cloud. The cloud may be as simple as
a small switch, or a WAN spanning hundreds or thousands of switches. With appropriate
internetwork signaling procedures (across the NNI) the cloud may extend its edges to en­
compass both public and private UNI points. Nodes attached at each UNI are identified by
one or more ATM addresses.

In the classical model, a logical IP subnetwork (LIS) consists of hosts and routers
having the same subnetwork address and netmask. Hosts connected to the same subnetwork
communicate directly. However, communication between two hosts on different LISs is only
possible through an IP router, regardless of whether direct ATM connectivity is possible
between these two hosts.
Figure 5.7: ATM networks linked to form wider ATM cloud.
Even though classical IP over ARP is very simple to implement and does not require any changes to existing systems, it is very limited, since communication between hosts on different subnets must occur through a router, even though it may be possible to open a direct VC between the two IP end stations over the ATM network. This is a significant limitation for an ATM based network. Currently, the Routing Over Larger Clouds (ROLC) working group of the IETF is investigating the possibility of setting up direct connections across ATM [99]. With this aim, the ROLC working group is working on a new protocol named “Next Hop Resolution Protocol” (NHRP) which relies on the use of “super” servers. The ultimate goal of the NHRP protocol is to enable a host to bypass some or all of the routers between source and destination hosts by establishing a direct connection through the ATM switches.

The classical IP over ATM has a number of weaknesses [98]. First, end-to-end quality of service (QOS) cannot be guaranteed as data traverses through several IP routers which do not allocate resources. Second, it does not scale well to the large size of the eventual ATM network due to a large number of routers. The NHRP protocol described by IETF’s ROLC working group uses a number of servers for setting up connections. This adds complexity for network management, and introduces more network failure points. Also, a large number of networks are mixed protocol environments, an IP only solution may not be easily acceptable in such a case.

5.2 Issues to be resolved:

It is necessary to develop protocols which expose ATM to the higher layers instead of hiding it. At the datalink layer, ATM offers a number of features, such as high bandwidth and
per-connection quality of service (QoS) guarantees, making it particularly attractive to multimedia applications. However, the existing protocols like LAN Emulation and IP over ARP in their effort to provide connection-less service over the connection oriented ATM employ techniques which make it a point to obscure ATM from the higher layers. Also, the higher layer protocols like TCP/IP with all their processing overheads fail to extend the benefits of ATM to their applications. There is a need to develop application specific "light-weight" protocols to improve the utilization of the features of ATM (It was shown that only twenty percent of the bandwidth offered by FDDI and Ultralan was utilized with TCP/IP [60]). Protocols should be designed which separate the control and data flows of the multimedia traffic. Such a design allows an application to control the data flow while delegating the data path to a more efficient device-to-device level, where devices (cameras, display workstations), rather than the processes are the real end points of communication.

At present ATM network service is limited to permanent virtual circuits (PVCs), which requires subscriber connections to be preconfigured. This approach is well suited for small networks, but it has been shown that an approach based on switched virtual circuits (SVCs) is necessary for providing a flexible large scale network. SVCs are not available yet [30][97], quicker standardization efforts are necessary to bring ATM to the radiologists' desktop.

ATM network services are available from the major local exchange carriers (LECs), interexchange carriers (IECs), and competitive access providers (CAPs). The LECs providing ATM services include GTE and the Bell operating companies (Ameritech, Bell Atlantic, BellSouth, Pacific Bell, US West). Major IECs offering ATM services include AT&T, MCI, and Sprint, while CAPs include LDDS WorldCom, MFS DataNet, and Teleport. Most
carriers do not provide access links below T-3 and then jump to OC-3. It is necessary to provide access links over T-1 to bring ATM to the desktop. Also, there is currently no inter-carrier linkage, which is necessary for ATM communication between users connected to different carriers. Such a facility is required in case of emergency review of medical cases, when the radiologist and physicians are attached to links provided by different carriers. There is also no standard set of applications supported by all of the ATM service providers. Each provider has its own specific type of services.

There is a need for faster standardization of protocols to ensure the development of a high bandwidth world wide internetwork. N-ISDN failed to proliferate because of slow standardization efforts. It has been decided that ATM will provide the transport service for B-ISDN, but slow standardization efforts are resulting in services like Frame Relay to gain popularity among business users. This might jeopardize the realization of the much touted B-ISDN. And already most hospitals are switching to ATM believing in the availability of a uniform public broad band network.

For remote consultation; tele-education; and access to medical information in facilities like National Library of Medicines's (NLM's) Unified Medical Library System (UMLS) [58][59], the medical professionals and students rely on the Internet. For faster service, the Internet should be based on higher speeds than the 45 Mbps which it now runs on.

The ATM Forum's Multiprotocol Over ATM is not yet standardized. The development of this specification is necessary to ease the transition of ATM network technology into the existing networks. Bridges and routers form the bottleneck of any internetwork, a method to provide connectivity among LANs without the need of intermediate bridges and routers should be developed.
In a large scale PACS, the optical juke box is still a single point failure [117][138]. Most reliability solutions do not provide redundancy here because of the high costs. Implementations of newly defined archive standards like CD-R (Compact Disk-Recordable) and HD-CD (High Density Compact Disk) should be considered [172].

Data compression is still a major issue. Although new algorithms using Wavelet compression provide for visibly lossless compression on standard X-ray images at compression rates of more than 30:1, they are still not used widely in the medical imaging field.

Security is of prime concern in any networking environment, more so in a medical network. High-speed encryption methods, high-performance key-agility techniques and methods for cryptographic system synchronization are being developed [165][173], their implementation in ATM based PACS should be considered.

Traffic control is one of the most challenging areas in the development of broadband ATM networks. Most of the solutions proposed suffer from serious shortcomings. Some are simple but include many approximations and assumptions that are hard to justify. Others include complicated mathematical solutions that may not be feasible for real-time implementation. The use of artificial neural networks (ANNs) should be considered for ATM traffic control [174]. It has been shown that traffic policing mechanisms based on neural networks are more effective than algorithmic ones such as leaky-bucket and others. Network interface units (NIUs) and Medical equipment interfaces to ATM networks are still expensive and not widely available, research should focus on developing faster interfaces for the various modalities. The radiologist display workstation is another area which merits a lot of attention. The American College of Radiology (ACR) currently recommends that small matrix images be digitized and displayed with a minimum of $500 \times 500 \text{pixels} \times 8$
bits per pixel, large matrix images be digitized with a minimum of $2000 \times 2000$ pixels $\times 12$
bits per pixel. Monitors that can handle this kind of resolution are commercially available.
But for images like mammographs with matrix sizes of $4k \times 5k$, display monitors are not
commercially available. Also, user-friendly PACS software which can take advantage of the
presently available display workstations needs to be developed.
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