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An Investigation into Multimedia Local Area Networks

A Thesis presented in partial fulfillment of the requirements for the degree of Master of Technology in Information Engineering at Massey University

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1997.

Acknowledgement

I express my deep sense of gratitude to my supervisor Dr. Jamil Y Khan, for all his help and guidance throughout this study. I also express my deep gratitude to my co-supervisor Prof. R.M. Hodgson for all his help and encouragement.

I also thank Prof. Jagan P Agrawal and Dr. Upkar Varshney for their prompt clarifications on their paper entitled "Performance Evaluation of a Multimedia Local ATM Network".

I also thank my family members for their unstinted support and encouragement during this study.

Finally, I thank all those who are directly or indirectly helped me in this study.

Abstract

In this thesis the performance of the Multimedia Local Asynchronous Transfer Mode Network (MLAN) protocol is evaluated by a computer simulation method using voice and data source models. SIMSCRIPT II.5, a discrete event simulation language is used for the simulation. In addition, Fiber Distributed Data Interface (FDDI) and Fast Ethernet networks were simulated for data traffic and their performance is evaluated using COMNET III, a communication network simulation package. The main aim of this work is to evaluate the performance of the MLAN and to analyse the suitability of MLAN for Multimedia Traffic. The work is further extended by comparing the performance of MLAN with FDDI and Fast Ethernet LANs. Simulation results show that MLAN protocol has some potential to operate as a Multimedia LAN. However, analysis shows that some modification of the protocol is required to increase the bandwidth utilisation.

Acronyms

AAL	ATM Adaptation Layer
ABR	Available Bit Rate
ARP	Address Resolution Protocol
ATM	Asynchronous Transfer Mode
B-ISDN	Broadband ISDN
BT	Burst Tolerance
CAC	Connection Admission Control
CBR	Constant Bit Rate
CDV	Cell Delay Variation
CDVT	Cell Delay Variation Tolerance
CLR	Cell Loss Ratio
CSMA/CD	Carrier Sense Multiple Access with Collision Detection
DLPI	Data Link Provider Interface
DVI	Digital Video Interactive
ELAN	Emulated LAN
FDDI	Fiber Distributed Data Interface
GFC	Generic Flow Control
HDTV	High-Definition TV
IEEE	Institute of Electrical and Electronic Engineers
IETF	Internet Engineering Task Force
IP	Internet Protocol
IPX	Internet Packet Exchange
ISO	International Organization for Standardization
ITU-T	International Telecommunications Union – Telecommunications
JPEG	Joint Photographic Expert Group
LANE	LAN Emulation
LAN	Local-Area Network
LEC	LAN Emulation Client

LECS	LAN Emulation Configuration Server
LES	LAN Emulation Server
LE_ARP	LAN Emulation ARP
LLC	Logical Link Control
LUNI	LAN Emulation User to Network Interface
MAC	Medium Access Control
MCDV	Maximum Cell Delay Variation
MCLR	Maximum Cell Loss Ratio
MCR	Minimum Cell Rate
MCTD	Maximum Cell Transfer Delay
MII	Media-Independent Interface
MPEG	Motion Pictures Expert Group
NDIS	Network Driver Interface Specification
NetBIOS	Network Basic I/O System
NNI	Network Node Interface
NTSC	National Television Standards Committee
ODI	Open Data Link Interface
P-NNI	Private NNI
P-UNI	Private UNI
PCI	Protocol Control Information
PCR	Peak Cell Rate
PHY	Physical layer
PMI	Physical Medium Independent
PMD	Physical Mmedium Dependent
PVC	Permanent Virtual Circuit
PVP	Permanent Virtual Path
QOS	Quality Of Service
SHD	Super High Definition TV
SVC	Switched Virtual Connection
TCP	Transmission Control Protocol
UBR	Unspecified Bit Rate

UNI	User-Network Interface
VBR	Variable Bit Rate
VCI	Virtual Channel Identifier
VC	Virtual Circuit
VF	Variance Factor
VPI	Virtual Path Identifier
VP	Virtual Path

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Introduction

1.0 General Introduction

Until mid 80's, network traffic was almost entirely comprised of voice and data traffic. With the advancement of computer technology, there is an increasing demand for multimedia traffic that comprises of audio, video, image, graphics, text and data. Generally, multimedia traffic requires medium to high data transfer rate or bandwidth. Different compression techniques are used to reduce transmission bandwidth requirements. For example, an MPEG-2 session requires a bandwidth between 4-10Mbps, to transmit audio and video signals, while the projected required bandwidth for HDTV is between 5-30Mbps [1]. The multimedia traffic requirements are low latency, low jitter, lower packet loss etc.

Different high speed network structures have been proposed to support multimedia traffic in Wide Area and Local Area Network environments. Broadband Integrated Services Digital Network (B-ISDN) [2] standard has been developed to integrate various types of traffic. It can offer very high data transmission rate using optical fiber link. B-ISDN is a logical extension of the narrow band ISDN. The B-ISDN will be able to integrate all existing network technologies. In addition to that it will be able to support all future teleservices like video-on –demand, video conferencing, high speed data transfer, videophony, home shopping etc. The need for a flexible network and advances in technology and systems led to the definition of the Asynchronous Transfer Mode (ATM) protocol. The ATM [3] protocol is the standard protocol for the B-ISDN and is standardised by the International Telecommunication Union-Telecommunications (ITU-T). ATM is also accepted as the technology to interconnect computers over ATM Local Area Networks by the computer industry in the ATM forum.

From the network architecture point of view, computer networks can be classified into three types of networks such as Local Area Networks (LANs), Metropolitan Area Networks (MANs), and Wide Area Networks (WANs). The classification is done depending upon the distance the corresponding network is designed to span. While LANs cover an area of a few Km, on the other hand MANs cover an area of up to several tens of Km. WANs are generally supported by public carrier services which link users separated by geographically wider distances.

Use of high-speed data and multimedia applications such as voice, video, graphics etc. have been increasing rapidly in local and wide area network environments. Such integration yields several benefits such as the economy realized by the shared usage of resources. Other benefits of integration are the ease of use of data resources such as file servers for voice applications; and the facilitation of added functionality in data applications, for example, voice annotation of text files and electronic mail [4]. As the services like voice, video, fax, graphics etc. are integrated on to the same LAN, the protocols designed primarily for data transmission may not be suitable to meet the requirements of the multimedia traffic [3]. Therefore, significant amount of research has been carried out [5,6,7,8] to integrate data, voice and video traffic on to a Local Area Network.. These new applications continue to make increasing demands on the performance of LANs. LANs are, therefore, required to provide not only high channel throughput, but also satisfy stringent delay requirements. To meet these increasing demands, it is essential that future LANs be capable of operating at much higher data rates achieving high channel efficiencies and lower delay. Operating at data rates ranging from several Mbps to several Gbps, an ATM network with its flexible traffic handling capacity and high data transmission rate could be able to support the multimedia services.

With the increasing demand for multimedia services, a shared media ATM LAN (a non-switch based) spanning relatively shorter distances of a few km and operating in native mode may be a better choice than a switch based ATM LAN for the reasons enumerated below.

In a switch based ATM LAN

i. Each terminal in the user group requires a direct full duplex link to the ATM switch
ii. All intra-campus traffic of an organization passes through the ATM switch, which need to relay the same message on all links resulting higher resource requirements for the ATM switch.

Hence, the cost of the ATM switch, its installation, operation, and maintenance need to be considered. Therefore, ATM LAN emulation is desirable as a backbone network connecting terminals/workstations and traditional local area networks. However, a shared media LAN based on ATM technology could be an alternative low cost choice for certain multimedia applications.

1.1 Aim of the Research

The main objective of this work is to simulate the Multimedia Local Asynchronous Transfer Mode Network (MLAN) protocol [9] for voice and data traffic in order to evaluate the performance of the protocol. Performance of the MLAN is evaluated by the computer simulation technique, using Data and Voice traffic models.

The performance of MLAN was compared with two existing high-speed LANs. The existing LANs used for this study are the Fiber Distributed Data Interface (FDDI) [10,11,12,13] and Fast Ethernet (100base-T) [7,14]. The FDDI and Fast Ethernet models were simulated using the COMNET III, a communication network simulation package. These high-speed LANs' performances were evaluated in terms of throughput, end-to-end delay, channel efficiency, and ability to integrate different services. Finally, the performance of MLAN was compared with that of FDDI and Fast Ethernet using the above parameters.

1.2 Thesis Structure

Chapter 2 presents an overview of ATM for multimedia communication. In this chapter, the multimedia traffic requirements are discussed. Some of the existing network technologies and protocols, which support multimedia applications, are also discussed. A brief introduction of the ATM protocol is given along with the capabilities of ATM to support multimedia applications in the context of a Local Area Network. In addition, traffic management issues in ATM networks are also discussed.

In chapter 3, the architecture of a generic LAN, along with the architectures of ATM LAN and MLAN are discussed. The requirements of ATM LAN in the context of multimedia communication are also briefly discussed. The MLAN protocol is discussed in detail and a comparison of the architecture of MLAN is made with that of ATM LAN, FDDI, and Fast Ethernet.

In chapter 4, the simulation models of the traffic generators (data and voice) along with the simulation model of MLAN protocol is discussed. The salient features of the discrete event simulation language are explained. An overview of FDDI and Fast Ethernet protocols is also given. The simulation results of FDDI and Fast Ethernet using COMNET III are discussed. Finally, a comparison of MLAN is made with that of FDDI and Fast Ethernet using the simulation results.

In chapter 5, the conclusions drawn from the simulation of MLAN is presented along with that of FDDI and Fast Ethernet. Finally, the scope for future work in MLAN is discussed.

Chapter 2

Overview of Asynchronous Transfer Mode (ATM) for Multimedia Communication

2.0 Introduction

In the present information age, telecommunications is playing a more vital role than ever before: demand for various services is growing and is becoming increasingly diverse. Future telecommunications would have to provide broadband multimedia services such as videophone [15], video communication services [16], HDTV [17], high-resolution image transmission, high-speed data (data requiring high bandwidth), speech services, audio services etc.

In this chapter, the multimedia applications, their characteristics and their network requirements will be discussed. Suitability of several existing and evolving LAN structures will be analysed for multimedia communication. Also, the Asynchronous Transfer Mode based network will be discussed.

2.1 Multimedia Communication Applications and their Characteristics

Multimedia communication refers to the processing of various delay sensitive signals such as voice and video and delay insensitive signals such as data and stored information (voice, video). Therefore, a multimedia network needs to satisfy the diverse requirements of different types of traffic.

2.1.1 Characteristics of Video/Image signal

Video signal in digital form is a time sequence of equidistantly spaced two-dimensional frames [18]. A frame consists of samples of an image/scene captured by a camera or generated by computer graphics. The structure of a video sequence is shown in figure 2.1.1a.

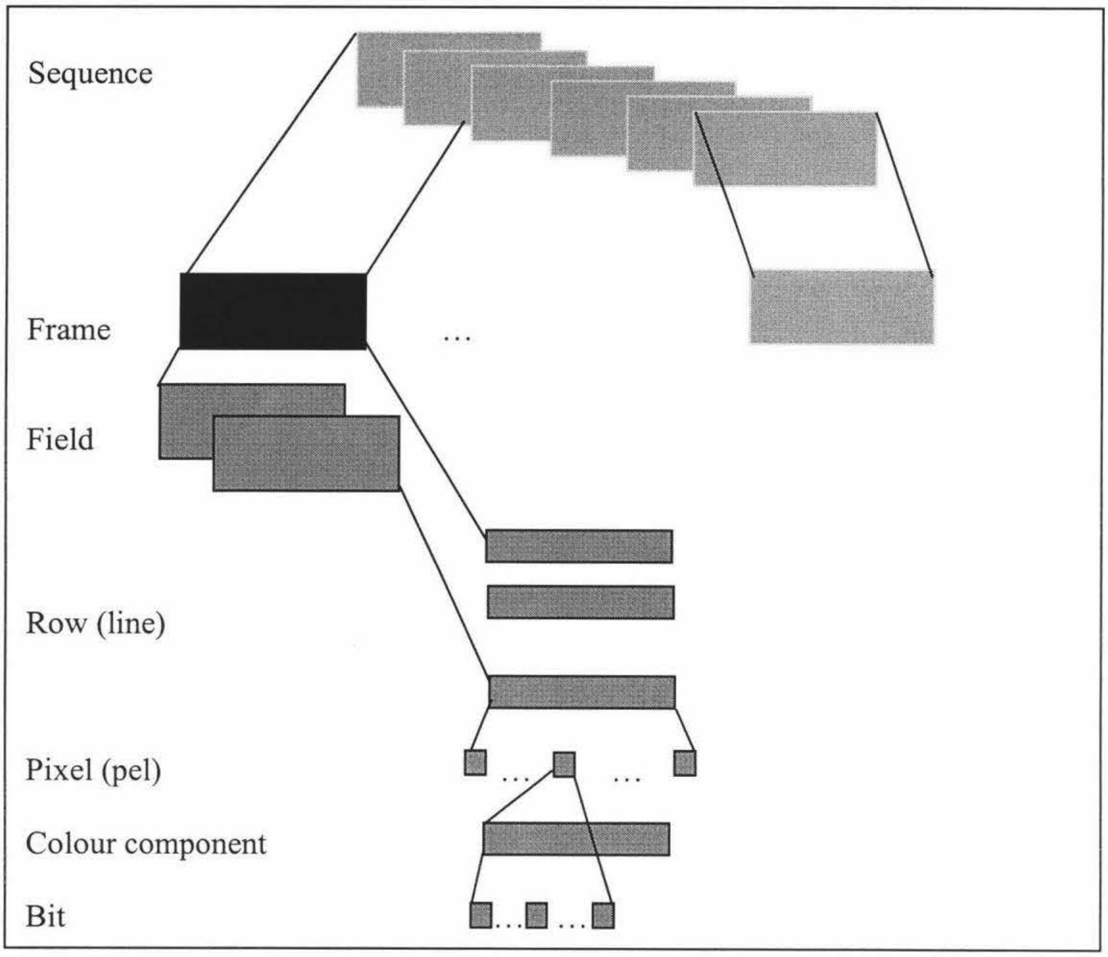


Figure 2.1.1a. Structure of a video signal.

As can be seen from the figure2.1.1a, the sequence consists of frames, which may be composed of fields. The fields are composed of rows and lines of pixels respectively, where each pixel is consists of three colour components (RGB) with a fixed number of bits. Digital image on the other hand, may be considered as a static one or still image, which is not in motion, that is not time dependent [19]. A video is also a digital image which contains the motion component (time dependent). The higher the quality of an image, the more pixels (and bits) are required to represent it, and consequently, the higher the data rate required to transmit the image.

The characteristics of the visual communications channel can be specified by the signal transmission rate and the transmission delay time [20]. The characteristics of digital image and video transmission services are shown organised by transmission rate (vertical axis) and transmission delay time (horizontal axis) in figure 2.1.1b. In this figure the digital image service's transmission time is considered to be essentially equal to the transmission delay time (i.e. the time required for the signal to transit the network), and the transfer rate is calculated to transmit the signal within that time. The multimedia signal bandwidth requirements are shown in table 2.1.1.

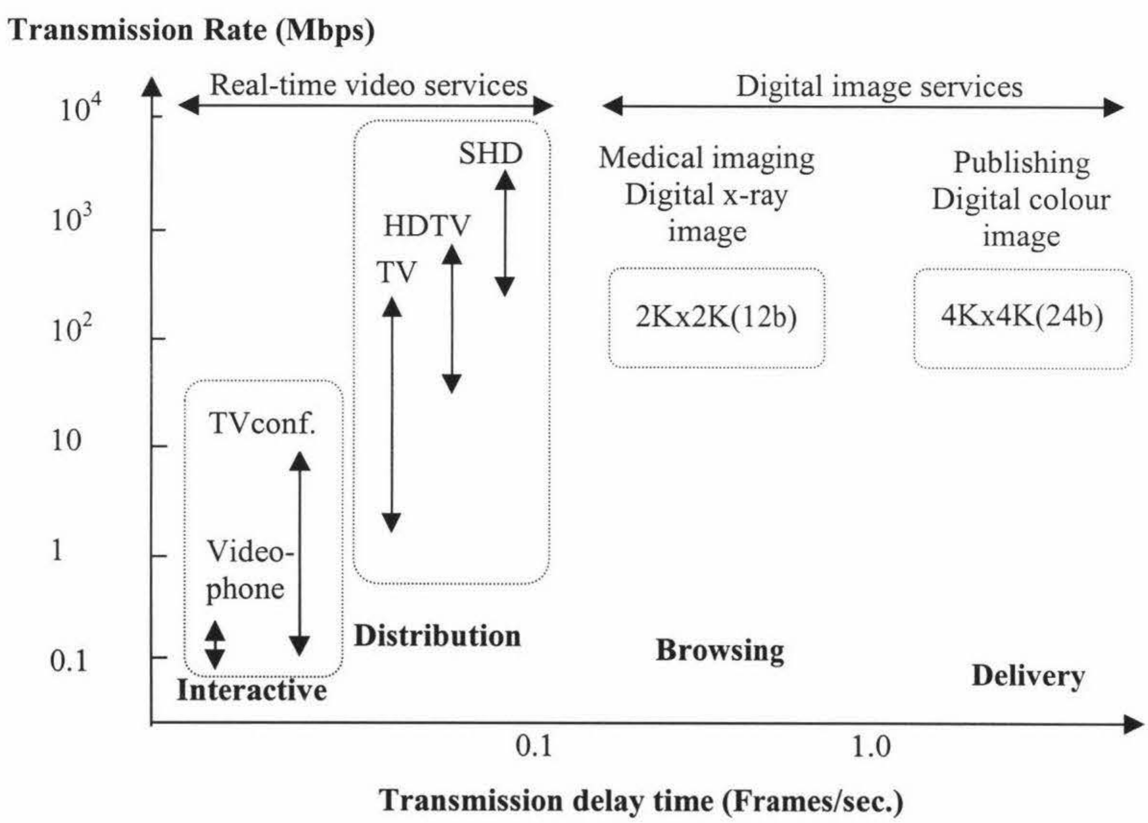


Figure 2.1.1b Digital images and video characterisation

Table 2.1.1 Multimedia bandwidth requirements ([21])

Media	Transaction Type	Format	Sampling dimensions Pixel, line, Frames/sec	Uncompressed bit rate	Compressed max. bit rate
Speech and music	Telephony		8 ksps x 8 bit/sample 16 ksps x 8 bit/sample	64Kbps	8 – 32Kbps
	Teleconferencing			128Kbps	48 – 64Kbps
	CD-audio			705.6Kbps	128Kbps
Image	Normal resolution image	SVGA	640 pixel x 480 line x 8 bit/pixel	2.458Mbps	24–245Kbps
		JPEG	720 pixel x 576 line x 16 bit/pixel	6.636Mbps	104-830Kbps
	Very high resolution image		1280 pixel x 1024 line x 24 bit/pixel	31.46Mbps	300 – 3Mbps
Business video	Videophone	QCIF (H.261)	176 pixel x 144 line x 12 bit/30 frames/s	9.115Mbps	P x 64Kbps
		MPEG-4 (H-320)	176 pixel x 144 line x 12 bit/10 frames/s	3.04Mbps	64Kbps
	Video conferencing	CIF (H.261)	352 pixel x 288 line x 12 bit/30 frames/s	36.45	m x 384Kbps (m=1,2,...,5)
		MPEG-1 (PAL)	352 pixel x 288 line x 12 bit/25 frames/s	30.4Mbps	1.15 – 3Mbps
		MPEG-1 (NTSC)	352 pixel x 240 line x 12 bit/30 frames/s	30.4Mbps	1.15 – 3Mbps
Entertainment video	VCR	CIF (MPEG-2)	352 pixel x 240 line x 12 bit/30 frames/s	30.4Mbps	4Mbps
	Broadcast television	MPEG-2 (PAL)	720 pixel x 576 line x 12 bit/25 frames/s	124.4Mbps	15Mbps
		MPEG-2 (NTSC)	720 pixel x 480 line x 12 bit/30 frames/s	124.3Mbps	15Mbps
	High quality television	HDTV	1920 pixel x 1080 line x 16 bit/30 frames/s	994.3Mbps	135Mbps
		MPEG-3	720 pixel x 576 line x 12 bit/25 frames/s	745.8Mbps	20 – 40Mbps

CIF : Common Intermediate Format

JPEG : Join Photographic Experts Group

MPEG: Moving Pictures Expert Group

QCIF : Quarter Common Intermediate Format

Real-time video transmission services may be classified as interactive and distributive. While interactive services are teleconferencing and videophone, broadcast services are normal television, Digital High Definition Television (HDTV), and Super High Definition (SHD) video. For all these services the transmission delay time must be shorter than the video frame period on average [20]. If this delay exceeds, the inter-frame gaps would be visible and the intelligibility of the audio signal drops down very sharply.

Applications that transmit still images include medical imaging, publishing, printing, and other image database search services. All of these communications can be categorised as distributive communications. As has been previously mentioned in this section the required transmission rates for digital image transmission services depends upon the image resolution and the required transmission time. For e.g., a medical imaging application, such as the rendering and transmission of a diagnostic X-ray, often involves 20 to 50 images, amounting to between 1 to 2.5Gb of information. A transmission time in the order of several seconds is desirable in medical imaging application [22]. In many cases real-time access is required to enable collaborative discussions to occur between physicians separated by long distances. Therefore, sheer volume of data must be transmitted in a very short time. The faster the response required by the service, the higher the transfer rate required.

If one considers the temporal variations in the content to be transmitted, it is apparent that the amount of information is essentially is variable. For example, in transmitting still digital images, during periods of transmission highest transmission bandwidth available should be offered to the terminal and during other periods, the connection can be used for some other terminal. Even when transmitting continuous video signals, if the encoding scheme removes redundancy by taking advantages of correlations either within a single image or between images, amount of information will change continuously according to the content of the video frame. With fixed rate transmission, transient quality degradations are not completely avoidable although small degradations may be avoided by applying rate control at the encoder. Further, constraints on the delay time also impose a limit on the buffer size that can be used to prevent degradation [20].

However, with variable rate transmission the situation changes. Since the video signal peak rate can be often predicted in advance from the characteristics of the video signal and the encoding algorithm, it would be possible to use variable rate transmission by selecting an appropriate upper limit for the variable-rate circuit. Therefore, rate control would be required less frequently than with the fixed rate transmission. Besides, with variable rate transmission, statistical multiplexing gain can be obtained. The statistical multiplexing gain will be discussed later.

An issue that must be addressed when using variable-rate transmission is delay-time jitter, which arises from contention or congestion that occurs in the packet multiplexer and switching equipment. For the transmission of real-time video and voice signals, delay times that exceeds a certain level are equivalent to a packet loss. Accordingly a correct estimation of network transmission delay distribution is critical for the variable-rate transmission of real-time video and voice signals. Further, delay jitter also gives rise to the problem of maintaining the transmission-side timing information (clock). In video transmission the frame frequency of the receiving terminal must match that of the transmitter. With these real-time services, frequency discrepancy of the video frame leads to data overflow or underflow. Therefore, the receiver extracts a clock component from the data and matches the phase using a technique such as phase-locked loop.

The influence of packet loss is not limited to the timing information maintenance problem, but also exerts a large influence on the image quality of the reproduced image or video. Therefore, recovery procedures are employed, which consist of first identifying packet loss and then performing some form of compensation. The simplest compensation technique is the insertion of previously received packets at the receiving side.

2.1.2 Characteristics of Voice/Speech signal

Voice traffic normally arises from two-or multiparty conversations. A voice signal consists of alternating segments of speech and silence, with each speaker spending about 40% of the time in talkspurts [23]. This characteristic of voice traffic may be exploited to achieve increased utilisation of channel bandwidth by silence suppression, i.e. by transmitting the voice signal only during talkspurts. The analog voice signal is encoded at a rate 'V' bits/s in a voice terminal. This rate may be fixed/constant or variable depending upon the application.

Some of the commonly used speech coding algorithms are [24]:

- 64Kbps pulse code modulation (PCM) using International Telecommunications Union (ITU) standard G.711 [25] to obtain a voice quality of 3.1 kHz
- 40/32/24/16Kbps adaptive differential pulse code modulation (ADPCM), using ITU standard G.726 [26]
- 16/12.8/9.6Kbps linear prediction using ITU standard G.728 [27]
- 64Kbps ADPCM, using ITU standards G.722 [28] and G.725 [29] to obtain a audio quality of 7 kHz – audio coding

Variable Bit Rate (VBR) coding employs ADPCM with silence detection. When speech activity drops, voice samples are not generated [24]. This takes account of pauses between phrases in human speech as well as silences while one person is listening to another speaking.

The principal requirement for voice traffic is bounded by delay. Delay can have two effects on connection performance. In the absence of noticeable echo, delay can interfere with the dynamics/activity of voice communication. In the presence of noticeable echo, increasing delay makes echo effects worse. At a delay of 50 ms, echo cancelers/suppressors are used [30].

2.1.3 Characteristics of Data Traffic

Data traffic is assumed to arise from computer communication applications. Data traffic may be interactive, which arises from applications such as remote logins and transaction processing, and non-interactive or bulk traffic arising from applications such as file transfers and electronic mail. Data traffic is typically bursty in nature, i.e. a station alternates between periods of high/low activity separated by relatively long periods during which it does not generate any packet. Data traffic requirements vary widely. Dta traffic is loss sensitive [31]. Variable delays on the order of several seconds may be acceptable in some cases such as bulk data transfers [31, 32].

These real-time multimedia services discussed previously have stringent delay constraints and demands not only high bandwidths, but also a predictable "Quality-of-Service (QoS)" [33]. Typical QoS parameters are latency, delay jitter, throughput, peak rate, loss sensitivity, reliability (mean time between failures) etc. [34].

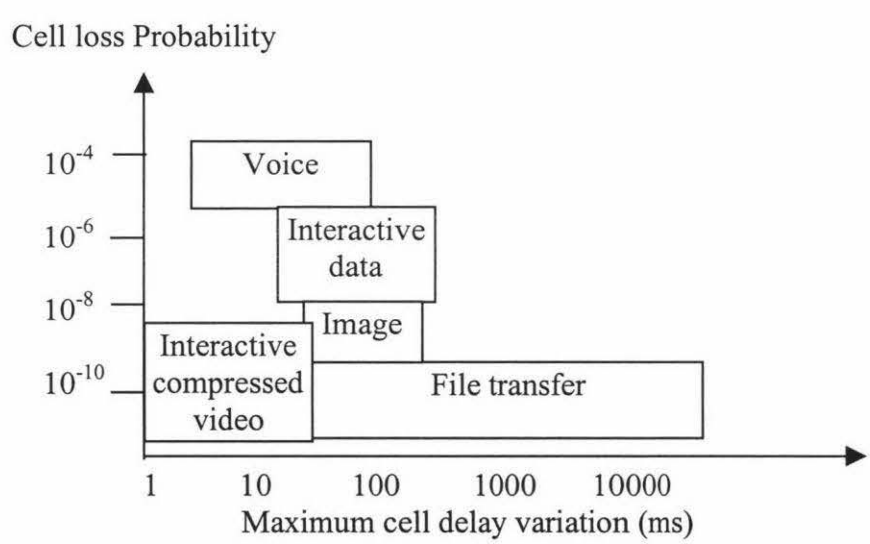


Figure 2.1.3 Multimedia traffic performance requirements

Figure 2.1.3 shows the delay and loss requirements for various multimedia applications. As can be seen from the figure 2.1.3, voice and video services are more sensitive to delay than data and image services. As mentioned previously, packet loss can result from buffer overflow at the destination or within the network, or from late packet arrivals at the destination. Packet loss can also occur due to congestion. Tolerable loss for video is generally much lower, but depends on the coding algorithm used and the algorithm used in reconstructing lost video cells at the receiver end. Packet loss upto 1% can be tolerated with much degradation in quality [33]. Loss tolerance can be higher if the source can designate particular packets for preferred dropping; this is termed as hierarchical coding.

2.1.4 Goals for real-time communication techniques

Before discussing the goals/performance metrics for real-time communication techniques, two of such important performance metrics such as latency and jitter are defined.

Latency is defined as the average end-to-end delay experienced by the packets [33] and Jitter is defined as the short term variations of the significant instants of a digital signal from their average positions in time [35].

The following are some of the desirable properties of real-time communication [33]:

i. Low jitter.
ii. Low latency that includes the response time and transmission time.
iii. Adaptable to dynamically changing network and traffic conditions by using rate-variability at the source encoder and congestion control schemes.
iv. High effective bandwidth utilisation by using statistical multiplexing.
v. Low overhead in header bits per packet/cell.

The different performance metrics mentioned above for real-time traffic suggests that network protocols and architectures developed earlier for data-oriented communication applications may not be well suited for supporting real-time and integrated real-time/non real-time applications [33]. Some of the local area network structures such as Switched Ethernet, Fast Ethernet, FDDI, and FDDI-II will be discussed in the following sections as these networks have the potential of carrying some form of multimedia traffic with existing set-up. Before discussing the above mentioned networks, architecture of a generic LAN will be discussed.

2.2 Architecture of a Generic LAN

IEEE project 802 LAN has defined a flexible architecture [36, 37] that is originated specifically to the standardization of local area network data link technology. The approach taken by IEEE in developing its LAN architecture is in conformance with the OSI model. However, IEEE project 802 LAN addresses only the lower two layers of the OSI model, the Physical and Data Link layers [37].

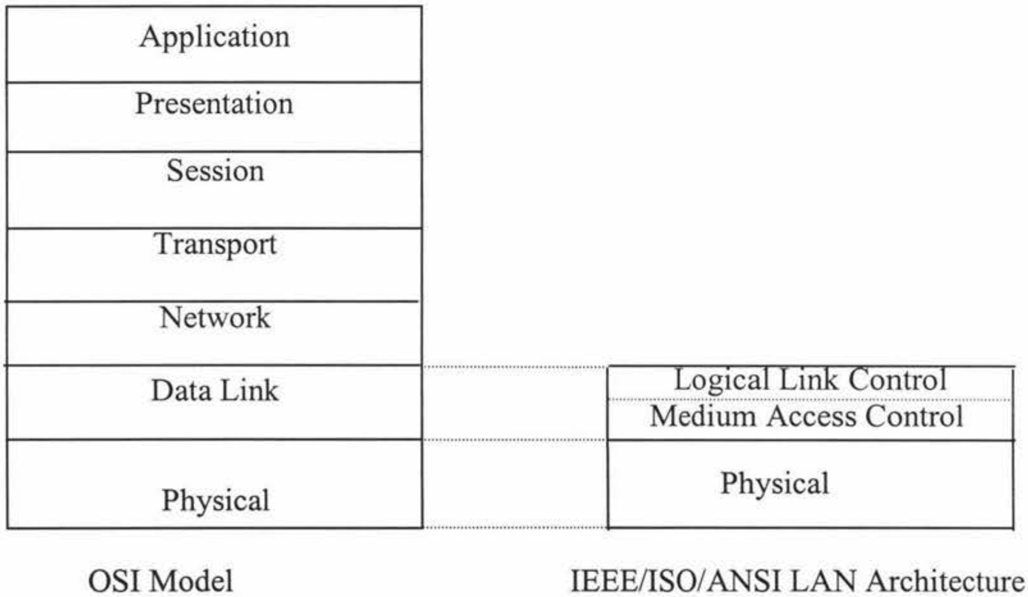


Figure 2.2 Comparing the layers of the OSI model with the layers and sub-layers of the IEEE/ISO/ANSI LAN architecture

As can be seen from the figure 2.2, in the IEEE/ISO/ANSI LAN architecture, the Data Link layer is divided into two sub-layers – the Logical Link Control (LLC) sublayer and the Medium Access Control (MAC) sub-layer.

2.2.1 Logical Link Control sublayer

The Logical Link Control sublayer allows a LAN data link user to access the services of a local area network data link without having to be concerned with the form of medium access control or physical transmission medium that is used. The data unit that the LLC sublayer entities exchange is called the logical-link-control-protocol-data-unit. The user of a LAN data link requests data transmission services through a Service Access Point (SAP) into the LLC sublayer.

The functions of LLC sublayer are as follows [37]:

- Provide one or more SAPs, which acts as a logical interface between two adjacent layers.
- On transmission, assemble data into a frame with address and Cyclic Redundancy Check (CRC) fields.
- On reception, disassemble frame, perform address recognition and CRC validation.

2.2.2 Medium Access Control (MAC) sublayer

The MAC sublayer provides services to the LLC sublayer. The data unit that MAC sublayer entities exchange is called the medium-access-control-protocol-data-unit, which is often called a MAC frame. The purpose of the MAC frame is to carry the logical-link-control-protocol-data-unit across a specific type of physical transmission medium from one network device to another.

The MAC sub-layer performs a framing function that adds header and trailer information to each MAC frame that is sent. The header and trailer contain information necessary to identify the beginning and end of a frame, specify the source and destination of a message, synchronize the sender and the receiver, and provide for error correction [37].

2.3 Existing LAN Technologies for multimedia communication

Some of the existing LAN technologies for multimedia communication will be discussed below.

2.3.1 Switched Ethernet

Switched Ethernet is a modification of traditional Ethernet, in which stations are connected to a communication hub instead of conventional tapped connection. Each station is connected to the hub by means of a transmission line. The hub reads the destination address of the packets on an incoming line and switches them on the appropriate outgoing line. The hub also contains a buffer in case when the outgoing line is not vacant. This connection is equivalent of having a dedicated 10Mbps connection per station. Since the connections are switched, each station can communicate as if it were the only station on the network. In such a way, switched Ethernet provides the bandwidth for many multimedia applications such as DVI (Digital Video Interactive) and H.261, and MPEG-1. However, it lacks the bandwidth for some of the distributive multimedia applications such as MPEG-2, HDTV, etc.

2.3.2 Fast Ethernet (100Base-T)

100Base-T [7, 14] uses the same types of hub-and-spoke topologies used by 10Base-T. 100Base-T provides a non-disruptive, smooth evolution from current 10Base-T Ethernet to high-speed 100Mbps performance. Fast Ethernet is compatible with the Simple Network Management Protocol (SNMP). Fast Ethernet uses the same CSMA-CD access protocol and therefore, shares the same limitations with regard to access delay characteristics. Therefore, 100 Base-T can not provide delay guarantees, since any station on the network can disrupt the multimedia stream through heavy traffic [38].

2.3.3 Fiber Distributed Data Interface (FDDI)

FDDI [10-13], which will be discussed in detail, later in chapter 4 supports both synchronous and asynchronous modes of traffic. Synchronous traffic consists of delay sensitive traffic such as voice, which need to be transmitted at regular time intervals. Asynchronous traffic consists of data packets produced by various computer communication applications such as File Transfer Protocol and mail. In the synchronous mode, FDDI has low access latency and low jitter, but unfortunately configuring a low delay limit results in decreased bandwidth utilisation [12]. Further being dual ring type LAN the terminals require two active interfaces [9]. In addition, FDDI can not support isochronous traffic, which is highly desirable in interactive multimedia applications. Isochronous traffic allows fixed number of packets of data to be delivered in a fixed time interval. To support isochronous traffic, the basic FDDI is extended to FDDI-II.

FDDI II [12] designed to support constant bit-rate traffic, uses wideband channels, and bandwidth allocation is done in units of 64 kbps, which for bursty sources can be a waste of bandwidth [9].

Therefore, the challenge is to develop networks that meet the previously mentioned performance objectives for different traffic types, while making the best possible use of all the resources such as transmission bandwidths, buffers etc.

2.4 Evolving LAN Technologies

Two evolving LAN technologies such as Iso-Ethernet and 100Mbps Demand Priority LAN will be discussed.

2.4.1 Iso-Ethernet (ISLAN16-T)

Iso-Ethernet (ISLAN16-T) [39, 40, 41] is sponsored by the IEEE 802.9 committee. The intention behind the ISLAN16-T standard is to define a LAN transport capable of handling synchronous services such as voice and video while retaining compatibility with 10Base-T. ISLAN16-T is an enhancement of Ethernet (10 Base-T) that includes a 6.144 Mb/s isochronous data service in addition to the 10 Mb/s Ethernet packet service. ISLAN16-T has three modes of operation – 10Base-T compatibility mode at 10Mbps, multi-service mode at 16Mbps, and all-isochronous mode at 16Mbps determined dynamically at power-up [41]. Total bandwidth is 16.384Mbps. A fourth mode, under consideration by the IEEE 802.9 committee, permits ATM operation at 16Mbps [39].

ISLAN16-T allows end stations (such as multimedia PCs, intelligent phones, and video servers) that require circuit-switched isochronous capability upgrade without affecting the established Ethernet base. ISLAN16-T is also backward compatible with the existing 10Base-T Ethernet networks. If the end stations have only Ethernet capability, the hub port provides the correctly encoded information across the twisted pair connection. Each end station has a dedicated 6.144Mbps of isochronous bandwidth in each direction. The isochronous channel is completely independent of the Ethernet channel and therefore, one does not effect the other. Collisions and traffic congestion on the Ethernet will not effect the voice, video, or other information carried by the isochronous channel.

The ISLAN16-T frame is based on the 8kHz clock reference with frames occurring for every 125 microseconds. ISLAN16-T is a time division multiplexed frame that includes 10Mbps of IEEE 802.3 bandwidth, 6.144Mbps of isochronous bandwidth (96 C or bearer channels), 64Kbps for signalling or control (D-channel), and 96Kbps for maintenance (M-channel). The ISLAN16-T frame structures allocate bandwidth so that data rates for the Ethernet are compatible with 10Base-T Ethernet data rates. The ISLAN16-T frame contains independent data channels that are multiplexed and demultiplexed at the physical layer. The physical layer multiplexes the Ethernet, isochronous, and D and M-channel data into a frame and provides it to the physical media through special encoding. At the receiving end, the ISLAN16-T frame decodes and demultiplexes. The Ethernet, isochronous, and D and M-channel data are handed to the appropriate interface.

2.4.2 100Mbps Demand Priority LAN

100Mbps Demand Priority LAN [7, 42] also referred to as the IEEE 802.12 uses frame switching based on round-robin access control scheme that relies on the standard star-wired LAN structure. With demand priority a station issues a request to the hub, when it has a frame to transmit. The hub checks for requests from its attached stations and indicates to one station that it may transmit a frame. The access is granted on a round-robin basis. Thus this scheme can easily guarantee delay bounds for any maximum packet size. In terms of bandwidth, although Demand Priority suits multimedia better than 100 Base-T, it supports a relatively small number (up to 30) of stations [42]. The main purpose of 100Mbps Demand Priority LAN is increasing bandwidth while protecting existing wiring and interconnection investments. A more efficient coding scheme and four pairs instead of one pair of wires achieve a tenfold increase in channel bandwidth and at a cost similar to that of 10 Base-T technology [42].

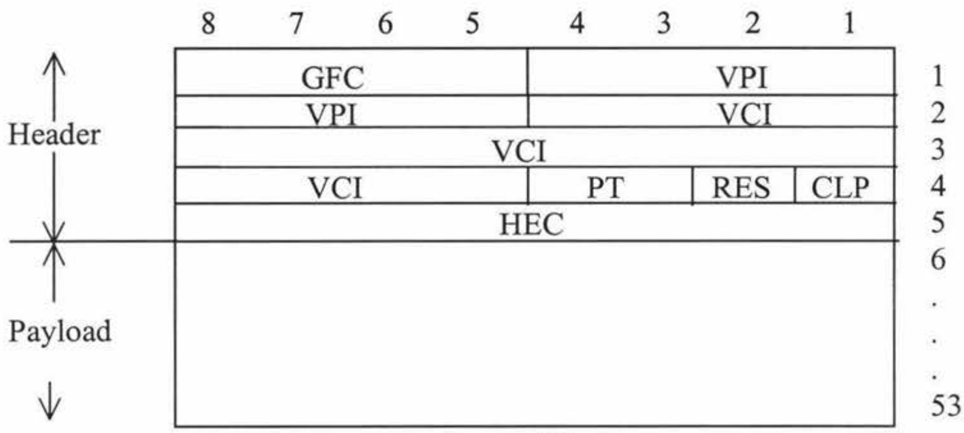
2.5 Asynchronous Transfer Mode (ATM)

ATM can support various types of networks including LANs. In this section ATM structure will be investigated in detail because most of the work done in this thesis is based on Multimedia Local Asynchronous Transfer Mode Network, which is an ATM based shared media LAN.

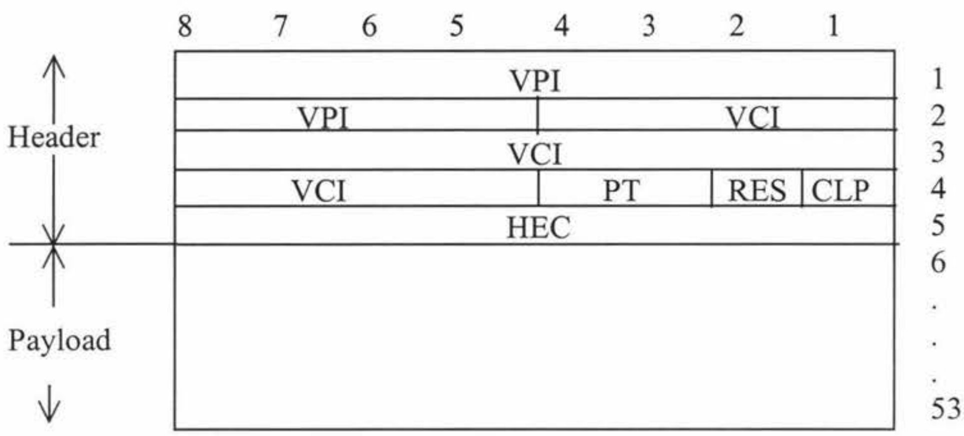
Cell relay (Asynchronous Transfer Mode) [3] has been designed to operate primarily at data rates of 10Mbps or above, using fixed size cells to transport information. Cell relay is more advantageous than frame relay because cells are fixed length, the processing requirements per cell is much lower. Thus, for a given level of processing power, one can achieve a higher number of data units per second with cells than with frames. Cells being small and fixed size are switched faster for a given level of switching power than frames, the transmission delay should therefore, be less [43]. Also cells being fixed size decrease delay variability. A long frame can occupy the transmission facilities for a long time, called "freeze-out". If all units of data are of same length, the delay variance may be decreased thereby allowing a fairer access to the facilities.

Asynchronous Transfer Mode (ATM) is a cell based multiplexing and switching technique and has the following features:

- Information is transmitted in short fixed length blocks called cells. Cells are transported at regular intervals; there is no space between cells, idle periods on the link carry unassigned cells.
- Overhead is minimised in order to maximise efficiency at the high bit-rates used (for example, flow control and error recovery, are performed on an end-to-end basis).



(a)



(b)

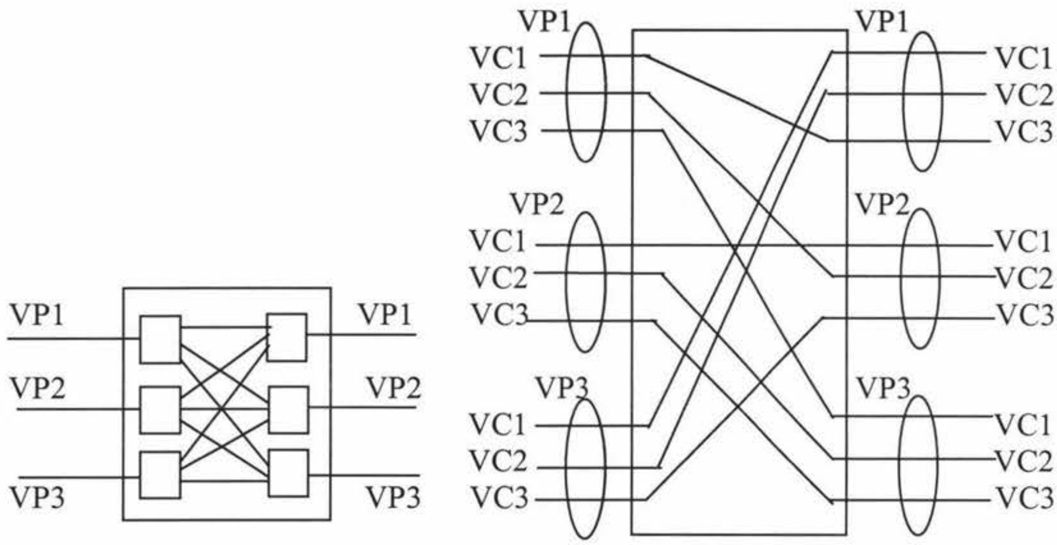
CLP: Cell Loss Priority; RES: Reserved; GFC: Generic Flow Control; PT: Payload Type; HEC: Header Error Control; VPI: Virtual Path Identifier; VCI: Virtual Channel Identifier.

Figure2.5.1ATM cell structure at (a) user-network interface; (b) network-node interface

Figure 2.5.1 illustrates the ATM cell structure at a user-network interface and at a network node interface. Each cell has a 5-byte header and 48-byte payload as shown in figure 2.5.1. The header consists of information necessary for routing, but does not contain a complete destination address. The cells are switched by the routing tables, which are set up when the network is initiates transmission. The header consists of several fields.

The Generic Flow Field (GFC) is used for congestion control at the User-Network-Interface (UNI) to avoid overloading. The Virtual Path Identifier (VPI) and Virtual Channel Identifier (VCI) fields contain the routing information of the cell. The Payload Type (PT) information represents the type of information carried by the cell. The CLP field indicates the cell loss priority, i.e. if a cell can be dropped or not in case of congestion. The Header Error Control (HEC) field is used to detect and correct the errors in the header. ATM does not have an error correction mechanism for the payload.

In an ATM network, a connection has to be established between two end points before data can be transmitted. An end terminal requests a connection to another end terminal by transmitting a signalling request across the UNI to the network. This request is passed across the network to the destination. If the destination agrees to form a connection, a virtual circuit is set up across the ATM network between these two end points. These connections are made of Virtual Channel (VC) and Virtual Path (VP) connections. These connections can be either point-to-point or point-to-multipoint. A VC is a logical connection between two switching points. A VP is a group of VCs with the same VPI value. The VP and VC switches are shown in figure 2.5.2.



(a) VP switch (b) VC switch

Figure 2.5.2 Virtual connections in ATM; (a) VP switch and (b) VC switch.

VCs that share the same VP have the same VPI. A VC switch must terminate VPs and can switch the VCs within a VP independently of each other including reassigning their VCIs as shown in the figure 2.5.2 (a). VPs and VCs are subject to switching within the ATM network. A VP switch can redirect a VP, perhaps reassigning the VPI, but keeps the VCs within the VP intact as shown in the figure 2.5.2 (b). VCs, which remain within a single virtual path throughout the connection, would have identical VCIs at both ends. Hence, cell sequence integrity is maintained throughout a Virtual Channel Connection.

2.6 ATM System Architecture

ATM is a layered architecture allowing multiple services like voice, data and video to be mixed over the network [44]. Three lower level layers have been defined to implement ATM's features. These are the ATM adaptation layer, ATM layer, and Physical layer. An ATM network can carry integrated traffic because it uses small fixed size cells, while traditional data networks use variable length packets, wherein a short packet gets delayed until the end of transmission of a long packet. Further, for ATM to support many kinds of services with different traffic characteristics and system requirements, it is necessary to adapt the different classes of applications to the ATM layer. The adaptation is done by the Convergence sub-layer and the segmentation and reassembly sub-layer, which will be discussed in detail later. Figure 2.6 shows ITU defined protocol reference model for B-ISDN. The characteristics of ATM adaptation layer, ATM layer, and Physical layer of the B-ISDN reference model shown in figure 2.6. Table 2.6 lists the activities of different layers.

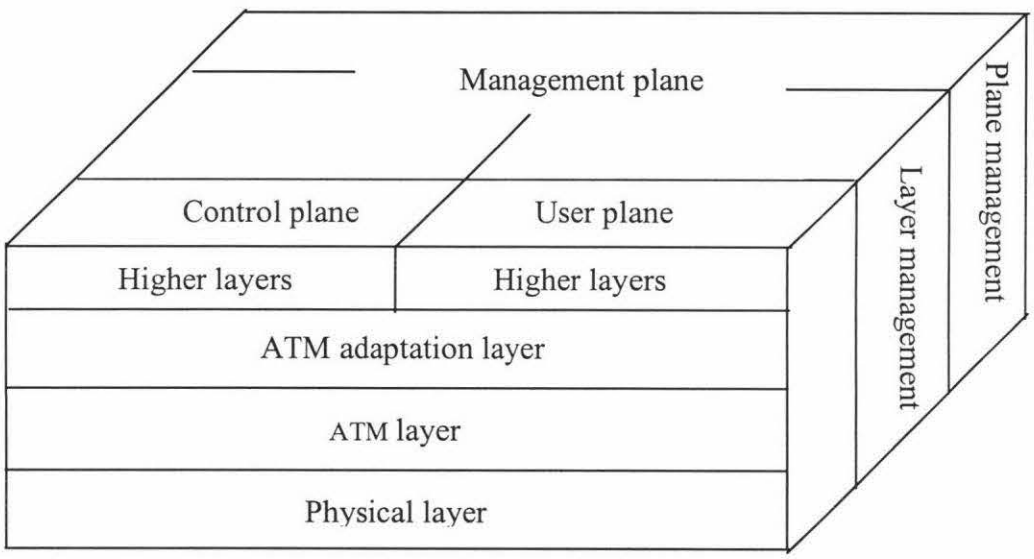


Figure 2.6. B-ISDN protocol reference model and the functions of the layer

2.6.1 The Physical layer

The physical layer [46] aspects of ATM can be broadly classified in to two categories: the Transmission Convergence (TC) sublayer and the Physical Medium Dependent (PMD) sublayer.

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

Table 2.6 ATM protocol model functions

Layer/ Sublayer	**Function**
ATM adaptation layer	
Convergence sublayer	Convergence
Segmentation and reassembly layer	Segmentation and reassembly
ATM layer	Generic flow control
	Cell header generation/extraction
	Cell VPI/VCI translation
	Cell multiplexing and demultiplexing
Physical layer	Cell rate decoupling
	HEC header generation/ Verification
Transmission convergence sublayer	Cell delineation
	Transmission frame adaptation
	Transmission frame generation and recovery
Physical medium sublayer	Bit timing
	Physical medium

The transmission convergence sub-layer performs all those functions necessary to transform a flow of cells into a flow of data units (e.g., bits) which can be transmitted and received over a physical medium. These two layers together specify how ATM cells would be transported across the physical medium and are referred to as the physical layer of a User network Interface (UNI). Two classes of UNIs have been defined: Public UNIs to be used to connect an ATM user to a public ATM network, and private UNIs to be used to connect an ATM user to an ATM switch that is part of a LAN as shown in figure 2.6.1. Private UNIs deal only with short distances while public UNIs connect users to central switches and must be capable of spanning long distances.

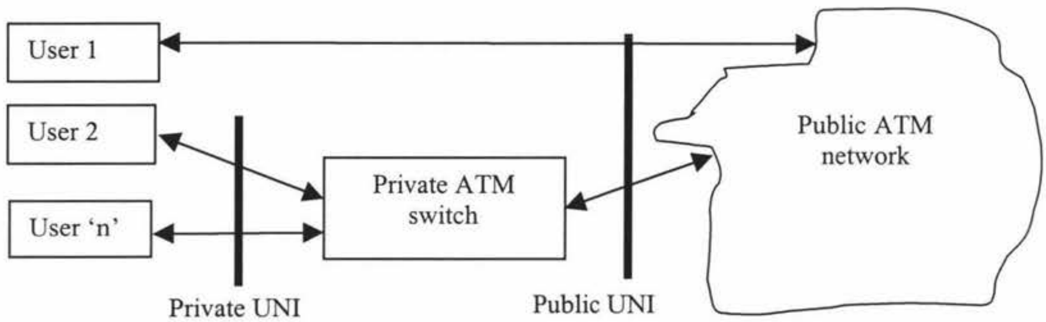


Figure 2.6.1 ATM User Network Interface (UNI).

2.6.2 ATM layer

ATM layer is the layer above the physical layer. It is a switching and multiplexing layer independent of the physical layer. It defines the structure of ATM cells. The functions of ATM layer are:

Cell multiplexing and demultiplexing - This function multiplexes cells from individual VPs and VCs into one resulting cell stream in the transmit direction. It divides the arriving cell stream into individual cell flows with respect to VC or VP in the receive direction.

VPI and VCI translation - This function is performed at the ATM switching and/or cross-connect nodes. As has been mentioned previously, at the VP switch, the value of the VPI field of each incoming cell is translated into a new VPI value of the outgoing cell. The values of VPI and VCI are translated into new values at a VC switch.

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

Generic Flow Control (GFC) - This function supports control of the ATM traffic flow in a customer network. This is defined at the B-ISDN User-to-network interface (UNI).

2.6.3 ATM Adaptation layers

In order for ATM to support many kinds of services with different traffic characteristics and system requirements, it is necessary to adapt the different classes of applications to the ATM layer. This function is performed by the AAL, which has two stages:

i. A service (traffic type)-dependent sub-layer called Convergence sub-layer(CS);
ii. A service dependent segmentation and re-assembly (SAR) sub-layer.

The CS assures the necessary error control and sequencing as well as the sizing of information. The SAR then chops the CS message into a 48-byte payload packet and attaches them to the 5-byte header. There are five types of adaptation services, designated AAL1, AAL2, …, AAL5 at the transmit node. AAL1 supports constant bit rate voice and video traffic, AAL2 supports variable bit rate voice and video traffic. AAL3 and AAL5 support connection oriented data and AAL4 support connectionless data (SMDS or

LAN-like) for cell relay switching. After formatting the message, the message is delivered to the segmentation layer where the cells are created and transmitted. At the receive side the cells go through the re-assembly layer and are passed to AAL1, AAL2, ..., AAL5 for the recreation of the original message.

AAL1 - It is intended for constant bit rate voice and video traffic, wherein cells are to be received in the exact sequence in which they were sent and they arrive at a constant rate. AAL1 assures sequence numbers and supports connection-oriented services that require constant bit rates and have specific timing and delay requirements.

AAL2 - Supports connection-oriented services that do not require constant bit rates. In other words, variable bit rate applications like some voice and video schemes.

AAL3/4 - This AAL is intended for both connectionless and connection oriented variable bit rate services. Originally two distinct adaptation layers AAL3 and 4, they have been merged into a single AAL which name is AAL3/4. AAL3 is designed for connection-oriented service and AAL4 for connectionless service such as datagrams. Some sort of indication has to be given to the receiver about the total length of the message so that an appropriate buffer size can be reserved for the message. AAL3 is very similar to AAL2 but AAL3 does not require synchronism between the receiver and the transmitter. AAL4 in addition must identify each cell as belonging to one datagram. So each cell is given a multiplex identifier (a 10-bit field) for this purpose.

AAL5 - Supports connection-oriented variable bit data services. It is a substantially lean AAL compared with AAL3/4 at the expense of error recovery and built in retransmission. This trade-off provides a smaller bandwidth overhead, simpler processing requirements, and reduced implementation complexity.

2.7 Capabilities of ATM

Some of the capabilities of ATM are statistical multiplexing, traffic integration, and network simplicity. These are briefly described.

2.7.1 Statistical multiplexing

In ATM networks terminals generate number of cells according to the source activity. The amount of network resources required by the user thus changes in proportion to the number of cells generated per unit time by the terminals. The network resources may then, be commonly shared by users, i.e. more appropriately, the resources are shared with either VC connections or Virtual Path (VP) connections established by user's requirements. Consider the situation shown in fig 2.7.1.

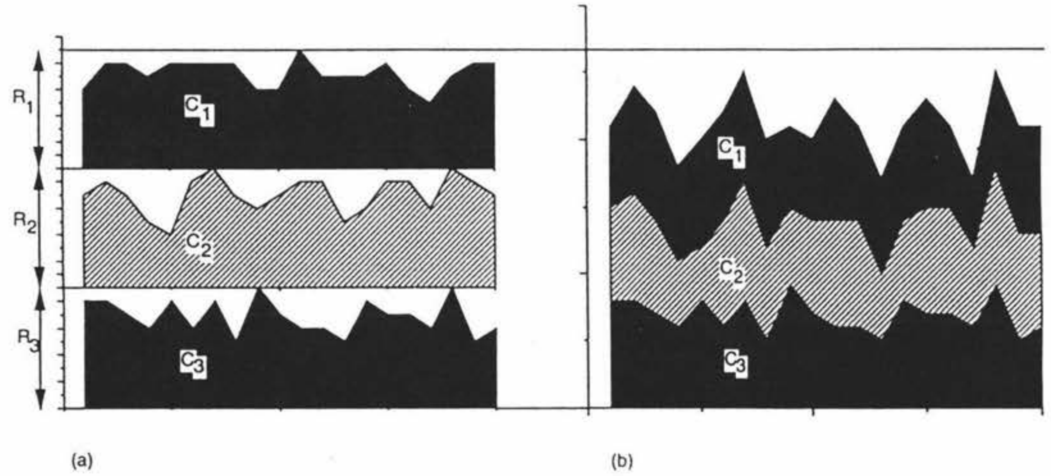


Figure 2.7.1 Statistical multiplexing gain: (a) no statistical multiplexing gain; (b) with statistical multiplexing gain. Adapted from [46].

If users 1, 2, ..., N require certain amounts of resources C_1, C_2, ..., C_N and if these resources do not take their peak values R_1, R_2, ..., R_N simultaneously, then for a fixed number of users the network can use less resources than would be required if resources were assigned according to the peak amounts required by each user, or it can increase the number of users accommodated by a fixed amount of resources. This is called statistical multiplexing gain [46], and is one of the main features of packet/cell based networks. For example, characteristics of VBR traffic may be represented by long term average and peak cell emission rates. Since not all VBR sources are expected to generate cells at their peak rates, the bandwidth less than the peak rate can be allocated to a VBR source. This allows for more sources to be admitted than the number of sources admitted by peak rates. At the same time, statistical variations in traffic load from individual sources can be smoothed out as many sources are multiplexed, resulting in better utilization of the shared resources [47]. In situations where the available bandwidth is less than the required one, certain buffering strategies such as output buffering, input buffering and internal buffering are used.

2.7.2 Traffic integration

With the standard cell format, data from different sources can be readily integrated in ATM networks. By dedicating resources for a brief cell time per source, data from sources appears to be transmitted concurrently. This is similar to time sharing systems where each job is given a fixed quantum of processor time. Further, being small and fixed size cells, a cell can be easily inserted into a cell stream from a large connection of data, whereas in data networks, a short packet gets delayed until the end of the transmission of a long packet. Further, true traffic integration takes place at the ATM layer as cells from different types of AAL and signalling are all mixed in the same cell format [47]. Also, the traffic integration helps improve the statistical multiplexing gain.

2.7.3 Network simplicity

Network simplicity is achieved in three ways [47], which are as follows:

- By taking advantage of very low bit error rates in optical fiber, transmission errors are not monitored at network nodes (except for header checking). Error handling is performed only at the network boundary nodes or at user-end devices. As the result of simplified node functions, ATM functions can be implemented in hardware, further improving on the processing speed. By reducing overhead at network nodes, ATM allows for very high-speed transmission with low levels of delays and delay jitters.
- Cells being small and fixed size, and also because cell sequence integrity is maintained, the boundary of a data frame is transparent to the receiver. This allows data to be transported in the same format over the entire network span regardless of data rates at intervening subnetworks.
- Routing of cells is made simple by pre-allocating routing labels for the entire fixed-path across the network thereby eliminating the need for reassembly and re-encapsulation in the network.

2.8 Disadvantages of ATM

The most significant disadvantages of ATM are cell delay variation (CDV) and cell assembly delay [35], which are explained below.

- Cell delay variation arises from the variable delays introduced in the network by the queues at switches and multiplexers and this lead to a change from what would be expected in the gap between different cells. The problem is most acute for a constant bit rate source where the difference in delay effects the quality of service perceived by the user. A typical example of such source is the speech.
- Cell assembly delay arises because information from a source is buffered until there is sufficient to fill a cell. The time the information waits in the buffer obviously depends upon the rate at which it is arriving and will be longer for low bit rate sources. Because the time spent in the buffer represents a delay it will have greatest impact on delay sensitive services. Voice telephony is again the most common

service to be affected. At 64Kbps transmission rate it takes 6ms to assemble 48 octets and this delay is significant when it comes to considering effects such as echo.

2.9 Traffic management in ATM Networks

A key issue for managing ATM connections in multimedia application environments is guaranteed Quality of Service (QoS). ATM traffic management control is responsible for this function. Traffic service classification, bandwidth reservation, and congestion control strategies for each class are the main problems that need to be solved in order to guarantee the multiple QoSs required over ATM [34]. A major issue of traffic management is congestion control, which would be discussed in some detail later.

ATM networks are connection-oriented in the sense that before two terminals on the network can communicate, they should inform all intermediate switches about their service requirements and traffic parameters [48]. Virtual circuits are used to support connection oriented services. Typically, a user declares key service requirements at the time of connection set up, declares the traffic parameters and may agree to control these parameters dynamically as demanded by the network.

2.9.1. Quality of Service (QoS) and traffic attributes

While setting up a connection on ATM networks, users can specify the following parameters related to the input traffic characteristics and the desired quality of service.

- **Peak Cell Rate (PCR):** The maximum instantaneous rate at which the user will transmit.
- **Sustained Cell Rate (SCR):** This is the average rate as measured over a long time interval.
- **Cell Loss Ratio (CLR):** The percentage of cells that are lost in the network due to error and congestion and are not delivered to the destination.

Each ATM cell has a "Cell Loss Priority (CLP)" bit in the header. During congestion, the network first drops cells that have their CLP bit set. Since the loss of CLP = 0 cell is more harmful to the operation of the application, CLR can be specified separately for cells with CLP = 1 and for those with CLP = 0.

- **Cell Transfer Delay (CTD):** The delay experienced by a cell between network entry and exit points is called the cell transfer delay. It includes propagation delays, queuing delays at various intermediate switches, and service times at queuing points.
- **Cell Delay Variation (CDV):** This is a measure of variance of CTD. High variation implies larger buffering for delay sensitive traffic such as voice and video.
- **Cell Delay Variation Tolerance (CDVT) and Burst Tolerance (BT):** For sources transmitting at any given rate, a slight variation in the inter-cell time is allowed. For example, a source with a PCR of 10,000 cells per second should nominally transmits cells every $100\mu s$. A leaky bucket type algorithm called "Generalized Cell Rate Algorithm (GCRA)" is used to determine if the variation in the inter-cell times (inverse of the rate) is acceptable. This algorithm has two parameters. The first is the inter-cell time and the second parameter is the allowed variation in the inter-cell time. Thus, a GCRA ($100\mu s$, $10\mu s$), will allow cells to arrive no more than $10\mu s$ earlier than their nominal scheduled time. The second parameter of the GCRA used to enforce PCR is called Cell Delay Variation Tolerance (CDVT) and of that used to enforce SCR is called Burst Tolerance (BT).

- **Maximum Burst Size (MBS):** The maximum number of back-to-back cells that can be sent at the peak cell rate but without violating the sustained cell rate is called maximum burst size (MBS).

Since MBS is more intuitive than BT, signalling messages use MBS. This means that during connection set-up, a source is required to specify MBS. BT can be easily calculated from MBR, SCR, and PCR. Note that PCR, SCR, CDVT, BT, and MBS are input traffic characteristics and are enforced by the network at the network entry. CLR, CTD, and CDV are qualities of service provided by the network and are measured at the network exit point.

- **Minimum Cell Rate (MCR):** This is the minimum rate desired by a user.

2.9.2. Service categories

To provide a guaranteed QoS, a traffic contract is established during connection set-up, which contains a connection traffic descriptor and a conformance definition. However, it is not necessary for every ATM virtual connection to have a specified QoS, since if only specified QoS connections are supported by ATM, then a large chunk of network resources can be wasted as one or more connections may not be fully utilizing their QoS contracts. However, unspecified QoS contracts can be supported by an ATM network on best-effort basis as such best-effort services are sufficient for most of the data applications. In general, the traffic contract specifies four categories of service and they are briefly described below.

- **Constant Bit Rate (CBR):** This category is used for emulating circuit switching. The cell rate is constant. Cell loss ratio is specified for CLP = 0 cells and may or may not be specified for CLP = 1 cells. Examples of applications that can use CBR are, telephone calls, video conferencing, and television (entertainment video).

- **Variable Bit Rate (VBR):** This category allows users to send at a variable rate. Statistical multiplexing is used and so there may be a small nonzero random loss. Real-time VBR application is sensitive to cell delay variation. Hence, for real-time VBR, maximum delay and peak-to-peak CDV are specified. An example of real-time VBR is interactive compressed video.

- **Available Bit Rate (ABR):** This category is designed for normal data traffic such as file transfer and email. Although, the standard does not require the cell transfer delay and cell loss ratio to be guaranteed or minimized, it is desirable for switches to minimize the delay and loss as much as possible. Depending upon the state of the congestion in the network, the source is required to control its rate. The users are allowed to declare a minimum cell rate, which is guaranteed.

- **Unspecified Bit Rate (UBR)**: This category is designed for those data applications that want to use any left-over capacity and are not sensitive to cell loss or delay. Such connections are not rejected on the basis of bandwidth shortage (no connection admission control) and not policed for their usage behaviour. During congestion, the cells are lost but the sources are not expected to reduce their cell rate. Instead, these applications may have their own higher-level cell loss recovery and retransmission mechanisms. Examples of applications that can use this service are email, file transfer, etc. Of course, these same applications can use the ABR service, if desired. Note that only ABR traffic responds to congestion feedback from the network.

ABR and UBR are usually specified in the traffic contract when the ATM network is providing a best-effort service. Thus, these two classes of traffic are sometimes referred to as best-effort service.

2.10 Congestion schemes

Congestion control requires considerable exchange of information among the nodes in the network. The exchange of information occurs through the headers of data cells, and some through control cells. Regarding the control policies for ATM LANs, the preventive congestion control schemes such as Connection Admission Control (CAC) and Usage Parameter Control (UPC), which have been investigated for public ATM networks, are not preferred [49]. Traffic control in ATM LANs should be accomplished by individual terminals in a reactive manner, rather than in a preventive manner. Examples of such reactive congestion control schemes proposed are credit-based and rate-based schemes [49].

2.11 Credit-based schemes

The Credit-based approach [50, 51] is based on window flow control, which consists of per-link and per-VC. Each link consists of a sender node (which can be a source end system or a switch) and a receiver node (which can be a switch or destination end system). Each node maintains a separate queue for each VC. The receiver monitors queue lengths of each VC and determines the number of cells that the sender can transmit on that VC. This number is called credit. The sender transmits only as many cells as allowed by the credit.

However, the approach has two drawbacks; if the credits are lost, the sender would not know it and each VC needs to reserve the entire round trip worth of buffers even though the link is shared by many VCs. This approach, therefore, requires link-by-link flow control and a separate buffer for each VC resulting considerable hardware complexity. Hence, these are considered expensive and inflexible.

2.12 Rate-based Approach

Rate based schemes [52] use feedback information from the network to specify the maximum rate at which each source can emit cells into the network on every VC. Three types of rate-based schemes have been proposed: Explicit rate control, Forward explicit congestion notification (FECN), and Backward explicit congestion notification (BECN). With explicit rate control, the network periodically determines at what rate each source should be transmitting and sends a message to each source informing of them of the new rate.

2.12.1 Forward Explicit Congestion Notification (FECN) Rate Control Scheme

Forward explicit congestion notification (FECN) is one of the proposed rate schemes that are based on end-to-end control, where computational complexity resides mostly in the end systems. The feedback mechanism makes use of the explicit forward congestion indication (EFCI) state carried in the payload type identifier (PTI) field to convey congestion information in the forward direction. When a switch becomes congested, it will mark on each VC the EFCI state of all cells being forwarded to the destination. Upon receiving marked cells, the destination returns congestion notification cells to the source of congested VCs to inform the source of the congestion status. The source uses this feedback information to increase or decrease the cell transmission rate accordingly on each VC.

2.12.2 Backward Explicit Congestion Notification (BECN) Rate Control Scheme

BECN uses similar mechanisms except that the congestion notification is returned from the point of congestion to the source. FECN requires very little additional hardware in the switch, which makes it attractive to the public carriers, who envisage employing very large switches.

An obvious advantage of BECN over FECN is the faster response to congestion and allows connections in the local area faster access to the available bandwidth, which is important for LANs. Also, BECN is more robust against faulty or non-complaint end systems because the network itself generates the congestion notification, which is an important consideration for a WAN, offering service to a private LAN. On the other hand, BECN requires more hardware in the switches to not only generate the BECN resource management (RM) cells but also filter the congestion information. The filtering process is necessary in order to prevent excessive RM cells being generated.

The behaviour of a rate-controlled source is similar for both BECN and FECN, so the two rate-based schemes may be combined and is given in [53]. An alternative implementation for a combined FECN/BECN mechanism is to use positive control messages. In this case a control message is sent periodically, on each connection, from the destination to the source when the forward path is not congested. This scheme is more robust against loss of control cells, link failure, and misbehaving users, because the transmission rate decreases on the loss of control information. Also BECN may be implemented simply by deleting control cells. A problem with periodic transmission of control information occurs when a large number of connections, each wishing to transmit at a rate approaching that of the frequency of the control messages. In this situation the amount of control traffic can approach the amount of data traffic. To avoid this problem a rate-based scheme may be designed where the control traffic on each connection is proportional to the forward data traffic [54].

Chapter 3

Architectures of High Speed Local Area Networks for Multimedia Communication

3.0 Introduction

Communication networks can be classified into two categories such as switched communication networks (i.e. point-to-point) and broadcast communication networks. Traditional Local area networks (LANs) are essentially broadcast type networks.

Advances in fiberoptic communications and developments in advanced protocols and transmission techniques, has led the use of LAN for transmission of integrated voice, video and data traffic. Such integration yields several benefits such as the economy realized by the shared usage of resources. As the services like voice, video, fax, graphics etc., are integrated on to the same LAN, the protocols designed primarily for data transmission in some cases may not be suitable to meet the requirements of the multimedia traffic [33]. The multimedia traffic requirements were already discussed in chapter 2. These new multimedia applications continue to make increasing demands on the performance of LANs. Hence, LANs should provide high channel throughput and satisfy stringent delay requirements. To meet these increasing demands, future LANs must be capable of operating at high data rates with high channel efficiency and lower delay. An ATM network, operating at data rates ranging from 10Mbps to about 622Mbps, is one of the options to support the future multimedia traffic.

In the local area, ATM can be deployed [55] for the following purposes:

i. Workgroup ATM, for client-server computing with high-end workstations and servers.
ii. Backbone ATM, for connecting existing hubs, bridges and routers in a campus.
iii. Connectivity with wide-area ATM networks.

In this chapter, the architectures of ATM LAN and Multimedia Local ATM Network (MLAN) are discussed. ATM LAN emulation is also briefly discussed. At the end of the chapter, features of ATM LAN and Multimedia Local Asynchronous Transfer Mode Network (MLAN) are compared with the as Fiber Distributed Data Interface (FDDI) and Fast Ethernet (100Base-T) networks.

3.1 Architecture of ATM LAN

An ATM LAN is based on a network of switches and dedicated links to each host, so an aggregate bandwidth of an ATM network increases, as hosts are added [55]. Figure 3.1 shows the architecture of an ATM LAN.

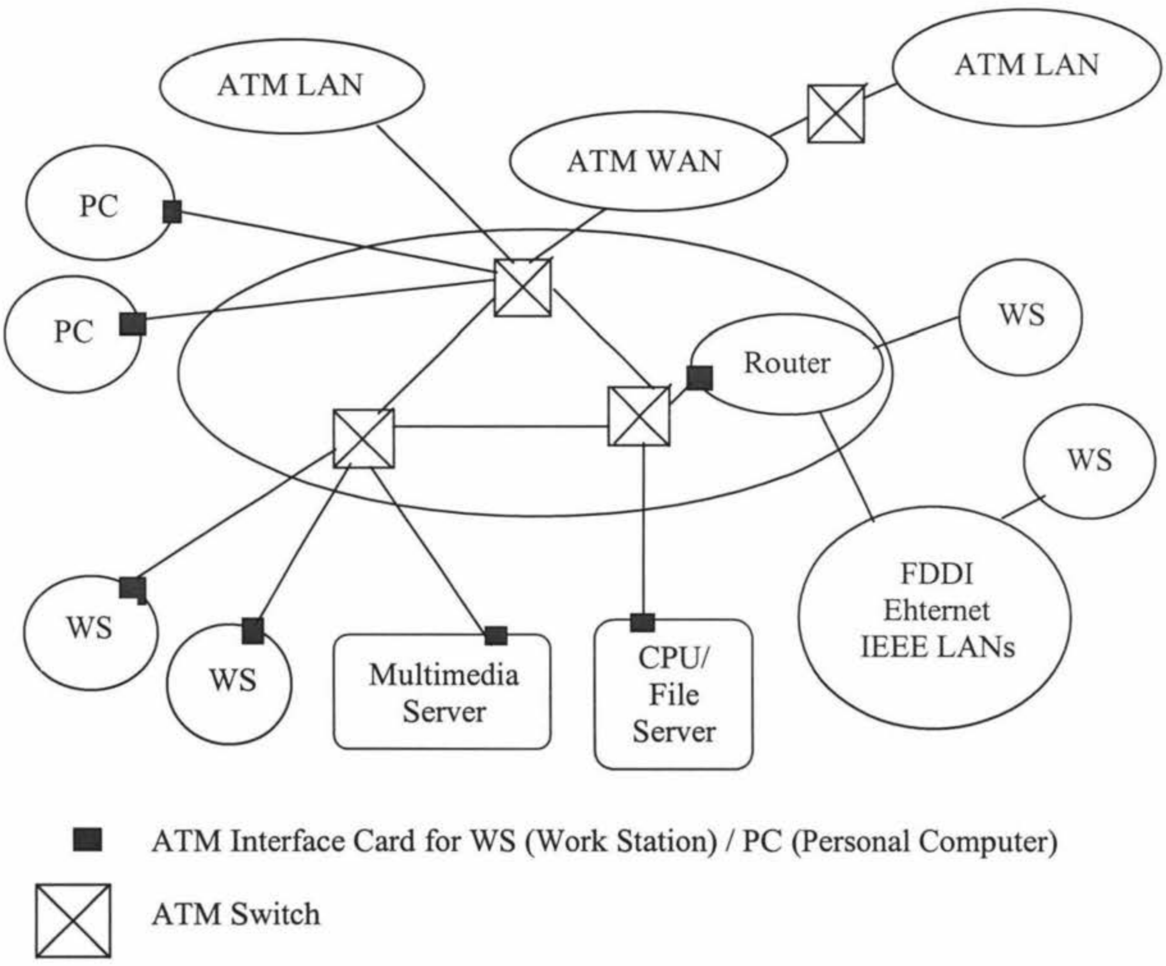


Figure 3.1. ATM LAN Architecture

There are several components of an ATM LAN [56] which are as follows:

* Hosts
* ATM switches
* Interworking devices such as routers and gateways
* Interfaces to the public ATM network.

A typical ATM LAN would use a mesh or star topology, high speed cell switching and standard ATM protocols. An ATM LAN may cover a small area by dimensioning the size of the ATM LAN switch (number of interfaces) and by using interfaces with short distances. A large campus network can be built by interconnecting multiple smaller ATM switches together in a distributed manner. Multimedia work stations and existing LANs such as Ethernet, Token Ring, FDDI, etc. can be connected directly to ATM switches. While ATM LANs will support Gbps speeds [56], the bandwidth of traditional LANs such as Ethernet, Token Ring, etc. is usually of the order of tens of megabits per second (Gigabit Ethernet being an exception). Generally, traditional LANs lack scalability. Key factors contributing to scalability are, switch based architecture and common cell structure. With switch based architecture, users can access ATM networks via a variety of physical connections irrespective of media types and applications. Within the limit of the physical link bandwidth, an arbitrary bit rate can be allocated to a user. The allocated bandwidth remains for the entire connection and the bandwidth can be provisioned in a scaled manner. As the network load increases and as more subnetworks need to be connected, more switch ports can be added in an incremental fashion. Present data communication networks re-encapsulate packets as network boundaries are crossed. In ATM, with common cell structure, packet re-encapsulation is avoided. An ATM VC connection is established on an end-to-end basis and cells cross network boundaries transparently. Further, it allows data to be transported over the entire network span regardless of data rates at intervening subnetworks [47].

ATM LAN emulation mode service allows a set of terminals, workstations and servers connect to the switch based network and to interact as if they were attached to the traditional LAN. The LAN Emulation service (LES), which will be discussed later, provides a bridged Ethernet/Token Ring/FDDI service that can be used to interconnect at full speed the conventional LANs. The access interface to such a service is the standard Ethernet/Token Ring/FDDI interface. It is the job of interworking function (IWF) to segment/reassemble the LAN frames and determines which ATM virtual connection should be used to reach the destination IWF. The LES can also bridge together conventional Ethernet/Token Ring/FDDI terminals and ATM terminals. The ATM LAN requirements may now be summarized.

3.1.1 ATM LAN requirements

The ATM LAN requirements [55] are as follows:

i. Support existing protocols transparently, especially the TCP/IP suit. For example, application binaries that use X Window system should continue to run unmodified over ATM.
ii. Support existing network management frame works.
iii. Be easily reconfigurable. No administrative intervention should be required for connecting or disconnecting an ATM host.
iv. Be reliable and robust. As long as a path remains between two systems they should be able to communicate.
v. Be efficient if there are multiple paths between two hosts, all should be used to capacity before new connection requests are denied.

If an ATM network satisfies these requirements we can claim it can be used as a plug and play replacement for any existing LAN. Performance generally improves not only due to the increased speed links, but also to the much larger aggregate bandwidth of the network.

In addition to the above requirements, an ATM LAN should also fulfill the following [55]:

- Support multiple guaranteed classes of service. Multimedia communication imposes two main requirements on the LAN [57], which are as follows:

i. It requires the support of connections with bandwidth guarantee. Such connections are used for example, to transfer audio and video streams, which implement the ATM adaptation layer, type 1 (AAL1) at both ends of the connection.
ii. Multiparty calls must be supported for conferencing, broadcasting, and multicasting services. Multiparty calls are considered as single entities consisting of several one-to-many ATM connections, i.e., they can have one or several destinations, but always have a single source. Many-to-many communication requires a group of one-to-many connections, all belonging to the same call. All connections can be routed via different paths, and cell sequence integrity is guaranteed within one connection. Multiparty calls are used not only for the distribution of broadcast information such as video, but also to support applications such as LAN emulation service, which would be discussed later. In shared-medium LAN technologies such as Ethernet and FDDI, multicast is essentially free. In switch-based networks, it represents an additional demand on the switch hardware and software architecture.

- Support standard ATM adaptation layers such as AAL3/4 and AAL5.

An ATM LAN satisfying these requirements allows applications that require high bandwidth, such as "video to the desktop", gets maximum performance (high throughput, and low response time) out of the network [57].

3.2 ATM LAN Emulation

The motivation for LAN emulation versus other possible solutions (e.g. "Classical IP over ATM", Internet Engineering Task Force RFC 1577) are as follows [58]:

i. The vast majority of currently available networks are mixed-protocol environments, using Internet Protocol (IP), IPX, NetBEUI, AppleTalk, DECNet, and so on; hence an IP-only solution is not generally usable in all environments. Very few of these protocols have defined mappings to ATM, which means that either a mapping must be defined for each such protocol, or a more common solution must be found. LAN emulation service offers the same MAC driver service primitives, thereby keeping the upper protocol layers unchanged. MAC device drivers include Network Driver Interface Specification

(NDIS), Open Data Link Interface (ODI), and Data Link Provider Interface (DLPI). They specify how to access a MAC driver. Each has its own primitives and parameter sets, but the essential services and functions are the same. LAN emulation provides these interfaces and services to the upper layers.

ii. ATM must be integrated with existing networks using bridges and routers so that it allows interoperability between software applications residing on ATM-attached end systems and on traditional LAN end systems.

iii. There is a huge base of existing applications that are not ATM-aware, and do not wish to be ATM-aware, but work with Ethernet or Token Ring. LAN emulation provides a general solution that allows any protocol defined to run over Ethernet or Token Ring to work transparently in an ATM environment

The main difference between ATM networks and the legacy LANs is, the size of the Protocol Data Units (PDUs) used to exchange information. While LANs use frames of variable sizes, ATM is a cell-based technology, which uses a fixed size of 53 bytes. Therefore, a LAN frame does not fit into one ATM cell and has to be segmented. Another obvious difference between today's LANs and ATM networks is that LANs are connectionless, whereas ATM networks natively support only connection-oriented services. Thus the most important function for a LAN emulation service is the provision of a connectionless service. An overview of various methods such as server based methods and methods without an overlay network, which provide connectionless service in ATM environment, is given in [59, 60].

In an existing IEEE 802 LAN segment, all communication (unicast, multicast, and broadcast) is broadcast to all stations on the shared physical medium, and each station filters out the packets it wants to receive. A physical LAN segment can be emulated by connecting a group of end stations on the ATM network to an ATM multicast virtual connection, which emulates the broadcast physical medium of the IEEE 802 LAN. It then becomes the broadcast channel of the ATM LAN segment. Any station may broadcast to all others on the ATM LAN segment by transmitting on the shared ATM multicast virtual connection. Existing IEEE 802 LANs, the membership of an individual LAN segment is defined by physical connection to the physical shared medium. With LAN emulation, membership of an ATM LAN segment is defined by logical connection to the multicast ATM virtual connection that emulates the broadcast channel for that ATM LAN segment. This offers terminal mobility and increased flexibility in network management [61]. In addition, with most LAN traffic being unicast [61], which can be supported using point-to-point ATM virtual connections, resulting in greater security because unicast traffic appears only at the two communicating stations and is not broadcast to all stations on the LAN segment. An ATM LAN segment can then offer much higher aggregate bandwidth than if all traffic were transmitted on the same broadcast channel. Also, the use of point-to-point virtual connections for unicast traffic permits much greater control of the quality of service thereby increasing the performance of the network [61] such as high throughput, low delay, and low probability of cell loss, etc. which are essential requirements for multimedia communications. Even multicasting is provided by several point-to-point virtual connections as mentioned previously in section 3.1.1.

LAN emulation is designed to function at level 2 (data link layer) of the OSI model, which means that it is independent of upper-level protocols. By implementing a single ATM MAC sub-layer, all higher layer protocols supported. With a single common interface, compatibility with the existing installed base of "legacy networks" is achieved. The ATM switch does not concern itself with LAN emulation, but rather with establishing a virtual connection and actually switching the cells. Figure 3.2.1 shows the components defined by ATM Forum for the support of a LAN emulation service over an ATM network.

The major components of ATM LAN emulation are the LAN emulation client (LEC), the LAN emulation server (LES), the broadcast/unknown server (BUS), and the LAN emulation configuration server (LECS). All devices that communicate with devices attached to traditional LANs (servers, routers, etc.) qualify as LECs. The BUS has the task of broadcasting multicast and unknown frames to all attached LECs.

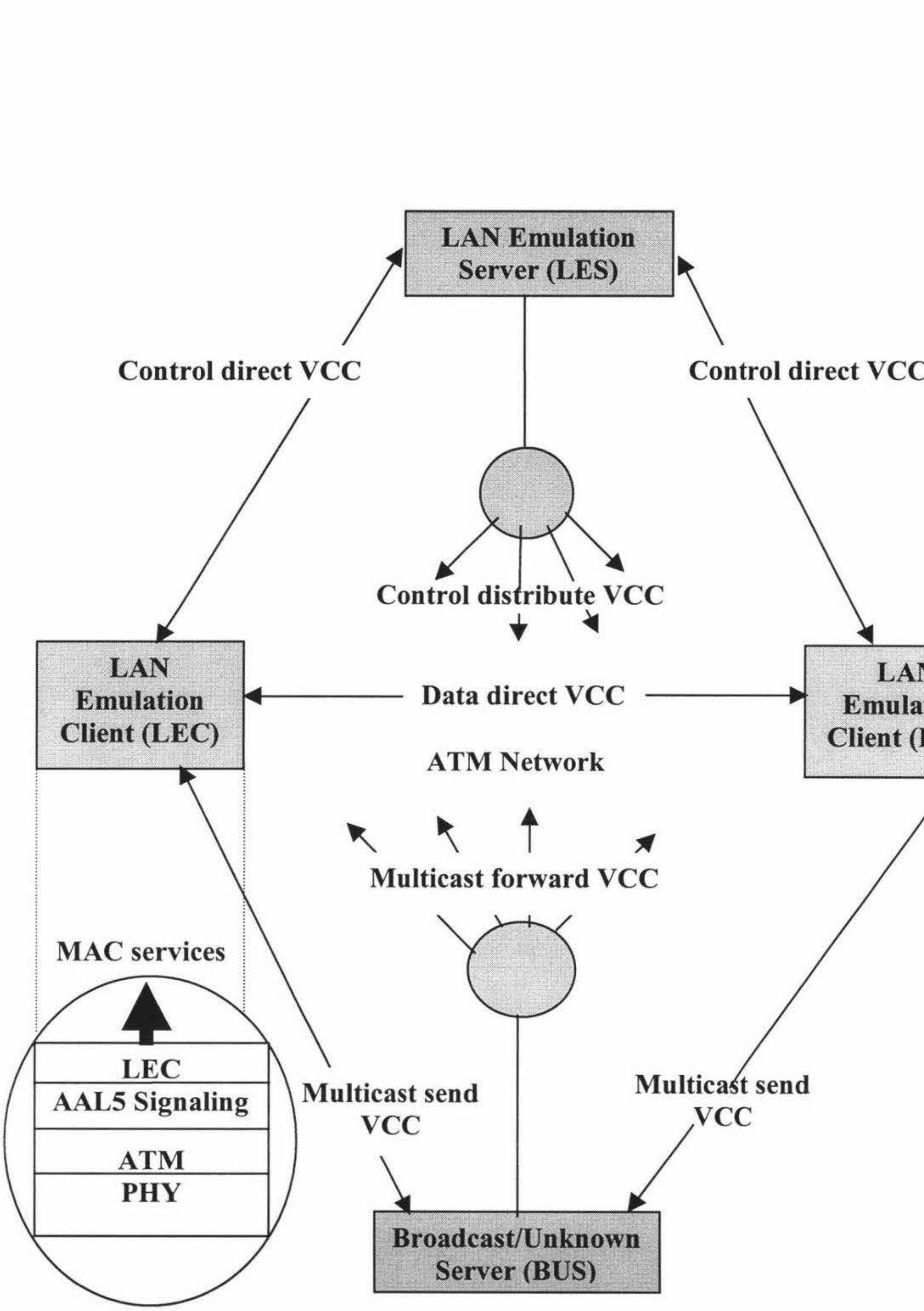


Figure 3.2.1. ATM Forum's LAN Emulation components.

The LEC acts as a proxy ATM end-station for LAN stations, and the LAN emulation server (LES) resolves MAC addresses to ATM addresses. LECs are assigned an ATM address for each attached LAN, and the MAC addresses of locally attached LAN stations are registered at a LES. When a LEC wants to forward a LAN frame over the ATM network to a target LAN station, it sends the LES a MAC ATM address resolution query containing the target station's MAC address. The LES responds with the ATM address of the LEC that is attached to the target station. The originating LEC then sets up an ATM-switched virtual circuit, converts the MAC frames to ATM cells, and transmits the cells over the network. At the receiving LEC, the ATM cells are converted to MAC frames and forwarded to the appropriate host.

The LECS enables LECs to configure themselves and join networks without requiring manual intervention. Its point-to-point connections to LECs are called "configure direct VCCs". The data direct, multicast send, and multicast forward VCCs connect the LECs and the BUS in a network of data plane connections. The configure direct, control direct, and control distribute VCCs connect the LECS, LES, and LECs in a network of control plane functions. ATM LAN emulation uses ATM adaptation layer type 5 (AAL5) frames, which converts packets in to cells, is shown in figure 3.2.2. A more detailed description of ATM LAN emulation is given in [29,62].

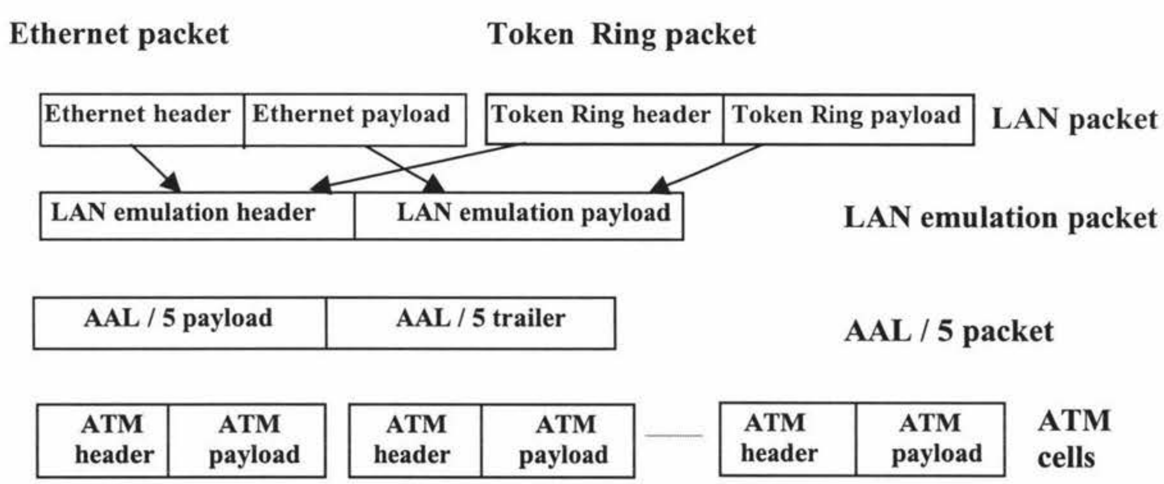


Figure 3.2.2. LAN emulation converts packets into cells.

In ATM LAN emulation over an ATM network

i. Each terminal in the user-group requires a direct full duplex link to the ATM switch,
ii. Each terminal in the user-group requires a dedicated interface to the ATM switch, and
iii. All intra-campus traffic of an organization (which is likely to be substantial) passes through the ATM switch, increasing its multicast traffic loading significantly.

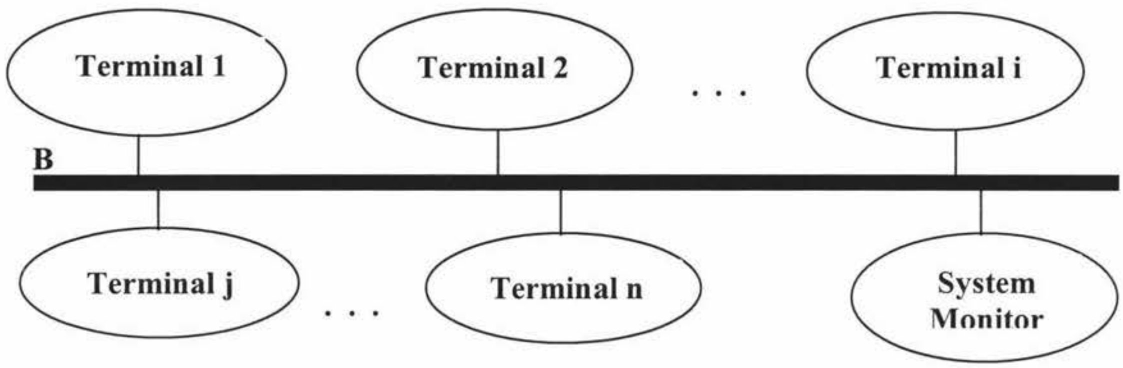
As has been previously mentioned in chapter 1, if a small ATM switch is dedicated to emulate an ATM LAN in an organization, the cost of the terminal-ATM switch interface as well as the cost of the ATM switch, its installation, operation, and maintenance must be considered. Hence, while ATM LAN emulation is desirable as a backbone network connecting terminals/workstations and traditional local area networks, a shared media ATM LAN that is cost effective and spanning relatively shorter distances could well be a better choice in other cases where cost is a criteria.

3.3 Architecture of Multimedia Local Asynchronous Transfer Mode Network

With the increasing demand for multimedia services, the demand for shared media local ATM networks spanning relatively shorter distances with the anticipated data rates, in the range of 100 to 150Mbps or higher, operating in native ATM mode is increasing. Data rates of 100-150Mbps have been chosen because some real-time services such as Digital TV and HDTV and digital image transmission services such as medical imaging, publishing, etc. which were discussed in chapter 2 require transmission rates ranging from 40Mbps to more than 150Mbps (Table 2.1 of chapter 2). Hence, there is a need for a shared media ATM LAN protocol designed to support multimedia traffic, operating in ATM mode to provide full connectivity to an ATM network, i.e. eliminating the need for ATM adaptation layer (AAL), and provide connectionless and as well as connection oriented services [9].

3.3.1 Overview of Multimedia Local ATM Network (MLAN)

The MLAN protocol [9] uses a bi-directional broadcast bus and supports tree topology, which is shown in figure 3.3.1. The system monitor monitors the traffic on the bus, periodically updates and broadcasts the MLAN parameters. A sequential polling scheme is used by the terminals to obtain media access for the desired amount of bandwidth and priority. A dynamic bandwidth reservation scheme is designed to satisfy the requirements of real-time and high priority traffic. The real-time and high priority traffic requirements such as low latency, low jitter, etc. A fixed size transmission frame is used which is divided in to a fixed number of ATM transmission slots.



B: L – Km Bidirectional bus

Terminal n: Number of terminals

Figure 3.3.1. The MLAN Topology

In MLAN, only inactive terminals are polled, i.e. terminals, which are not currently transmitting, and for this reason the polling is referred to as "Polling Inactive Customers Only". Since all call durations in multimedia traffic such as voice, and video services are expected to last over many successive frames (few seconds to a few minutes or more), the terminals are given media access for the entire duration of the current talkspurt (in the case of voice) and burst (in the case of video), thus minimizing the polling overhead. In MLAN, transmission and polling are done sequentially. With the result, the elapsed time between any two successive responses from the terminals is only the propagation delay between two transmitting terminals instead of the end-to-end bus propagation delay (t_{pde}).

The ATM mode of operation in MLAN is achieved by requiring the terminals to prepare packets in ATM cell format for intra-MLAN (internal) communication as well as for MLAN-ATM network (external) communication. For internal communication on the MLAN, the cells should contain source and destination addresses. Therefore, the 24 bits in the ATM cell header normally allocated for virtual path identifier (VPI) and virtual circuit identifier (VCI) fields are used in a manner to indicate the source and destination addresses or internal communication. For external communication, the VPI/VCI fields are used in the conventional manner.

The attributes of MLAN protocol are as follows and these are discussed in detail subsequently.

* Determination and broadcast of MLAN parameters
* Polling and Bandwidth reservation
* Transmission of information

3.3.2. Determination and Broadcast of MLAN parameters

Each type of traffic is allocated a certain amount of minimum bandwidth according to the expected load it would generate by creating traffic boundaries in a frame. The boundaries are adjusted dynamically by the system monitor, which monitors all the activity (traffic conditions) on the bus, updates and adjusts the MLAN parameters (determines optimum bandwidth allocation for each type of traffic) accordingly. It then broadcasts a Control Cell (the Control cell distributes these parameters to all terminals) with updated information in the beginning of each frame. Each terminal uses the control cell information to schedule cell transmissions, channel access requests, and updates the queues until the next control cell arrives, and keeps track of their locations in the respective queues.

PR	ABM	CLASS 1	CLASS 2	CLASS 3	CLASS 4
64 bits	8 bits	A_1:S_1/Q_1	A_2:S_2 :Q_2	A_3:S_3:Q_3	A_4/S_4 /Q_4

352 bits

1 ATM cell

Figure 3.3.2 Control Cell Format

PR (Preamble) - Serves to delimit successive frames, enables terminals to resynchronize their clocks with system monitor's clock.

ABM (Activity bit map) - is used by terminals to determine the queue of inactive terminals, whose physical addresses are in ascending order. For example, a "1" in a location indicates the terminal is in an active queue and a "0" indicates the terminal is in an inactive queue.

i. $\mathbf{A_i}$ - Maximum bandwidth allocated in terms of ATM slots for each class of traffic (i = 1 to 4) i.e. the maximum number of ATM cells allotted to each class of traffic.
ii. $\mathbf{S_i}$ - Amount of bandwidth already reserved by terminals for each class of traffic in terms of ATM slots, i.e. the maximum number of ATM cells reserved by the terminals for each class of traffic.
iii. $\mathbf{Q_i}$ -Terminal queue length for each class of traffic.

3.3.3 Polling and Bandwidth Reservation

Terminals are polled to reserve the desired number of ATM cells/frame (say "m" ATM cells/frame). In the case of voice, bandwidth reservation is given to terminals for the duration of the call. The smaller the frame duration, the lower the media access delay. However, a smaller value of frame duration results in a larger polling overhead, supporting a smaller number of terminals. The frame duration is taken as 6ms. This is because voice packets are of 48 bytes and with 8kHz sampling rate the packet delay for voice should not exceed 6ms. Two queues (active and inactive) are maintained for each terminal for each class of traffic. An inactive queue is maintained for the terminals that are unable to receive bandwidth reservation because of the unavailability of free ATM slots in the frame. During polling and bandwidth reservation process, the terminals perform the following tasks:

i. Receive the control cell and using this information, update the active queues and parameters $\mathbf{A_i}$, $\mathbf{S_i}$, and $\mathbf{Q_i}$, and form the inactive queue, and
ii. Monitor poll responses and cell transmissions and update the queues accordingly.

A 4 – bit (b_1 b_2 b_3 b_4) polling field is used to indicate the class of traffic and the amount of bandwidth required in terms of ATM cells (m), and a 1 – bit (c_1) field to indicate whether a connectionless ($c_1 = 0$) or connection oriented ($c_1 = 1$) service is desired.

The bandwidth reservation process is depicted in figure 3.3.3.1. It has been assumed that there are 16 terminals T_1 to T_{16}, on the bus and the queue status at the beginning of the frame is as follows: Class1:{T_5, T_7 , T_9}, Class2:{T_8, T_{11}}, Class3:{T_4, T_{12}}, Class4:{T_3, T_{14}, T_{15}}, and the inactive queue is:{ T_1, T_2 , T_6, T_{10}, T_{13} , T_{16}}. The first terminal in the inactive queue T_1, responds as soon as it sees the last bit of the control cell by placing its 5 - bit poll response (b_1 b_2 b_3 b_4 c_1; c_1 = "0" for connectionless traffic and "1" for connection oriented; b_1 b_2 - indicate classes of traffic, i.e. 00 class 1; 01 - class 2; 10 - class 3; 11 - class 4 and b_3 b_4 - indicate amount of bandwidth requested in terms of ATM cells) on the bus. T_2 responds as it sees the bit 'c_1' of T_1, and so on.

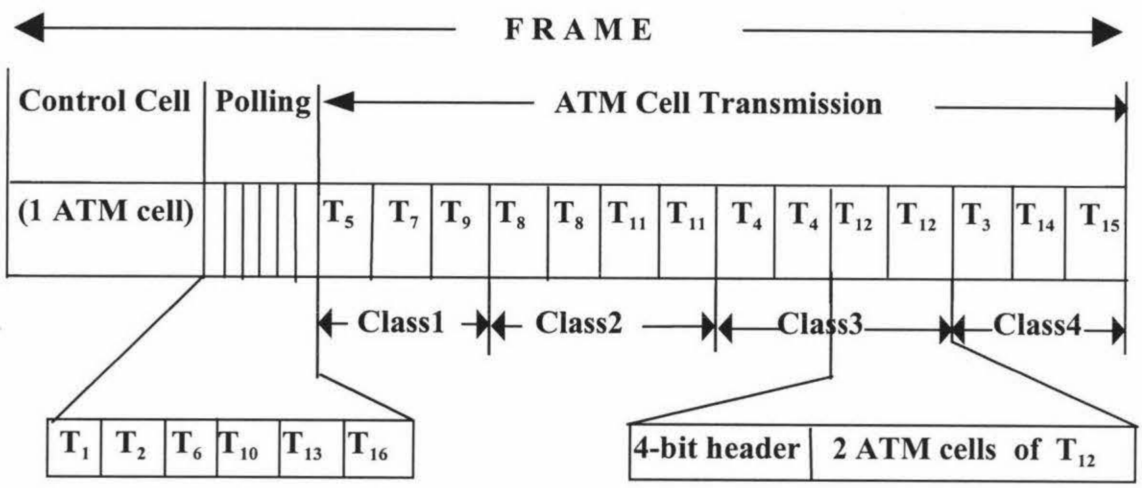


Figure 3.3.3.1. A typical MLAN transmission frame

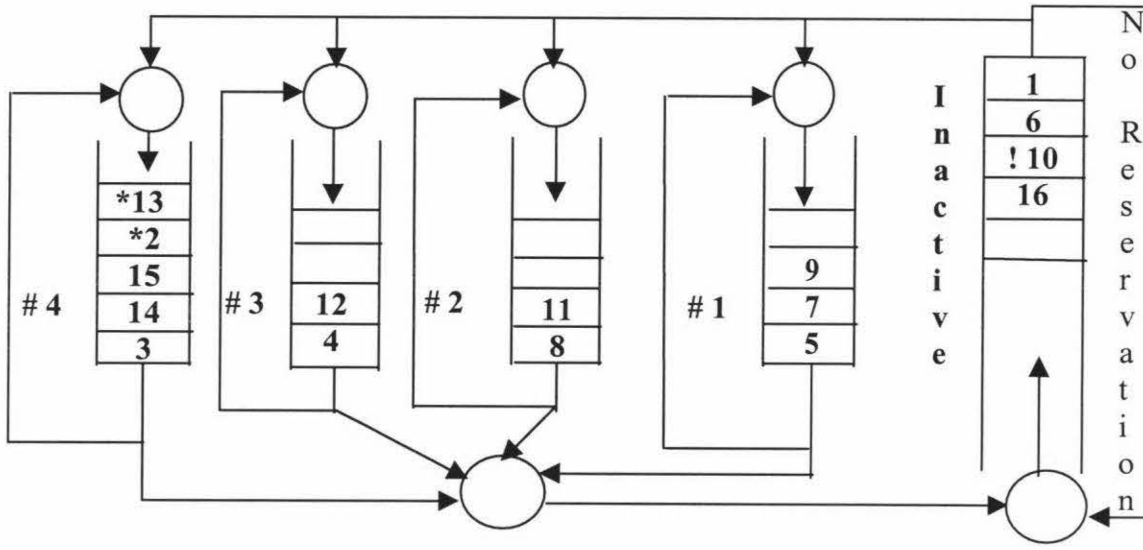
A terminal's request for class 1-3 traffic is accepted or rejected as follows:

- For connectionless service of class i traffic requiring m cells / frame, reservation is given, if at least m cells are available for class i traffic, i.e. $\mathbf{m < (A_i\text{-}S_i)}$
- For connection oriented service of class i traffic requiring m cells / frame, reservation is given, if at least 2m cells are available for class i traffic, i.e. $\mathbf{2m < (A_i\text{-}S_i)}$ since two reservations of m cells each are made for a connection oriented service.

Once these conditions are satisfied, terminals

i. update $\mathbf{S_i}$ to $\mathbf{S_i + m\,(C+1)}$
ii. update the inactive queue by removing the terminal from it and
iii. update the class i queue by entering the terminal in it.

In the case of class 4 traffic, the request is queued irrespective of whether bandwidth is available or not and cells are transmitted once their turn comes. Hence only queues are updated and the parameters are not updated. The poll responses in figure 3.3.3.1 shows new requests for connectionless service by the terminals T_2 and T_{13} for class 4 calls, and by T_{10} for a class 3 call. Assuming that T_{10}'s request is not entertained due to the lack of sufficient bandwidth, the status of the queues at the end of the polling cycle is as shown in figure 3.3.3.2.



***: Additions to the queue; !: Reservation not granted**

Figure 3.3.3.2. Queues after polling for the situation shown in figure 3.3.3.1.

3.3.4 Transmission of Information

Terminals would begin ATM cell transmissions immediately following the poll response of the last terminal in the inactive queue starting sequentially with the terminals in class 1 queue to class 4 queue. A 4-bit transmission header (a_1 a_2 b_3 b_4) is added to its first cell by each transmitting terminal in each frame. The bits (b_3 b_4) indicate the amount of bandwidth reservation and the bits (a_1 a_2) indicate call status as follows:

i. a_1 a_2 = 00 - call termination
ii. a_1 a_2 = 01 - no cell in this frame
iii. a_1 a_2 = 10 - no cell in the next frame and
iv. a_1 a_2 = 11 - call to continue.

The sub-headers a_1a_2 = 01 and 10 are used when the arrivals are bursty and / or irregular. When call termination is indicated by terminal, the corresponding "m" reserved slots are released; $\mathbf{S_i}$ is updated to $\mathbf{S_i - m}$ and the queues are updated accordingly.

3.4 A Comparison of ATM LAN, MLAN, FDDI, Fast Ethernet, Switched Ethernet and Iso-Ethernet

A comparison of ATM LAN, MLAN, FDDI, Fast Ethernet, Switched Ethernet and Iso-Ethernet is given in table 3.4.

Table 3.4. Illustrates a comparison of ATM LAN, MLAN, FDDI, Fast Ethernet, Switched Ethernet and Iso-Ethernet.

ATM LAN	MLAN	FDDI	Fast Ethernet	Switched Ethernet	Iso-Ethernet
Switch based architecture.	Bus architecture.	Dual ring architecture.	Bus architecture.	Hub based architecture	Bus architecture.
Each terminal in the user group requires a direct full duplex link and a dedicated interface to the ATM switch.	Requires only one interface per terminal to the bus. Operates in native ATM mode.	Each terminal requires two interfaces. Does not operate in native ATM mode.	Requires only one interface per terminal. Does not operate in native ATM mode.	Each terminal needs a direct full duplex link and a dedicated interface to the switch. Does not operate in native ATM mode.	Requires only one interface per terminal. Does not operate in native ATM mode.
Can be used as desktop. But generally used as backbone network.	Can be used as a LAN in a small campus environment [9].	Because of the higher cost, primarily used as a backbone network [47]. However, in some cases, high performance workstations are connected to FDDI LAN.	Can be used as a LAN in a small campus environment.	Can be used as a LAN for multimedia traffic	Can be used as a LAN for multimedia traffic
Dimensioning the size of the switch, support for number of terminals can be increased.	Can support up to 250 terminals at 100Mbps at a bus length of 3km under full load.	Supports about 500 terminals up to a distance of 100km.	Support for the number of terminals varies with segment length and traffic load.	Dimensioning the size of the switch, support for number of terminals can be increased.	Support for the number of terminals varies with segment length and traffic load.
Multicasting is not inherently available.	Multicasting is inherently available.	Multicasting is inherently available.	Multicasting is inherently available.	Multicasting is inherently available.	Multicasting is not inherently available.
Transmission delay is fixed and is <1ms. Suitability for multimedia is excellent.	Transmission delay is fixed and is < 1ms. Suitability for multimedia is excellent.	Transmission delay is configuration dependent. Suitability for multimedia is good.	Transmission delay is unpredictable. Suitability for multimedia is average.	Transmission delay is predictable. Suitability for multimedia is good.	Transmission delay is fixed and is <1ms. Suitability for multimedia is excellent.

Chapter 4

Simulation of Multimedia Local Area Network (MLAN) for Voice and Data Traffic

4.0 Introduction

The MLAN protocol [9] and its architecture have been discussed in chapter 3. The performance of MLAN is evaluated by computer simulation using data and voice source models. Two traffic generators are used to study the performance of the MLAN protocol. Traffic generators are validated before using them to obtain performance result of the MLAN protocol. Performance of the protocol such as throughput, queuing delay (access delay), and average cell delay are evaluated at different bus lengths, bus speeds, and terminal activity levels. In addition, the MLAN is also simulated with voice traffic generator and its performance is also evaluated for throughput, average cell delay and the speech cell loss.

In addition, two other high speed LANs such as FDDI and Fast Ethernet models have been simulated using the COMNET III, and their performance has been evaluated for throughput and packet delay. In this context, an overview of FDDI and the Fast Ethernet protocols have been briefly discussed and the simulation results of these two protocols are compared with that of MLAN protocol.

4.1 Data Traffic Generator

One measure of network capacity is the volume of traffic carried over a period of time. Two important parameters used to characterize traffic are the average arrival rate λ and the average burst size B. The average burst size is calculated as follows:

The sum of the number of messages generated in each burst divided by the number of bursts. If M_1 M_2, …, M_I, …, M_N are the number of messages generated in each burst, then the average burst size is given by

$$B = (M_1 + M_2 + \ldots + M_i + \ldots + M_N)/N \qquad \ldots \qquad [4.1.1]$$

The traffic intensity "A" is expressed as

$$A = \lambda B \qquad \ldots \qquad [4.1.2]$$

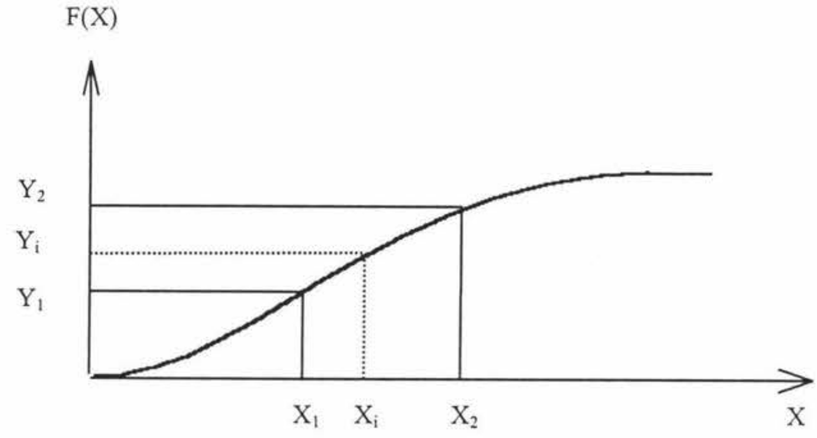
Let the time axis be divided in to a large number of small time segments of width Δt. Designate the average call arrival rate from a large number of independent sources as λ. Using the assumptions that only one call can occur in any sufficiently small time interval (segment) and the probability of a single terminal call arriving in a segment be proportional to the length of the segment, Δt, with a proportionality constant, λ, which represents the mean arrival rate. Then the probability of an arrival is $\lambda\Delta t$. This leads to the Poisson arrival distribution or Poisson distribution process, which is given by the following equation:

$$P_n(\lambda t) = \{(\lambda t)^n/n!\}\, e^{-\lambda t} \qquad \ldots \qquad [4.1.3]$$

Data traffic is often modeled using Poisson distribution process [63] with an exponential mean inter-arrival time of "λ" seconds. Data packets/cells are generated using a negative exponential distribution of different mean arrival rates.

4.2 Voice Traffic Generator

Human speech consists of speech and silence periods. In a normal telephone call without noise a speaker talks for about 44% of the time [64]. However, in practical situations, duration would be higher than 44% because of external noise. Since no information is transmitted during the silence period channel could be released during that period so that some other terminal can get access to transmit their packets. The talkspurts and silences were generated using probability transformation method as shown Fig 4.2.1. [64]. Brady's distribution shown in figure 4.2.2 has been adapted in this particular model [64]. Speech packets/cells from speech detector is strongly influenced by the choice of threshold levels [65]. In the case of a threshold level of -40 dBm, the mean talkspurt and silence periods were taken as 1.34 and 1.67 seconds respectively [64]. Any talkspurt less than 15 ms was considered as noise and silence less than 200 ms was taken as a talkspurt [66].



F(X): Probability density function; $X_1, X_2, \ldots, X_N$ = Measured abscissa; $Y_1, Y_2, \ldots, Y_N$ = Measured ordinate.

Figure 4.2.1 Transformation Techniques for Generating Talkspurt and Silence Length.

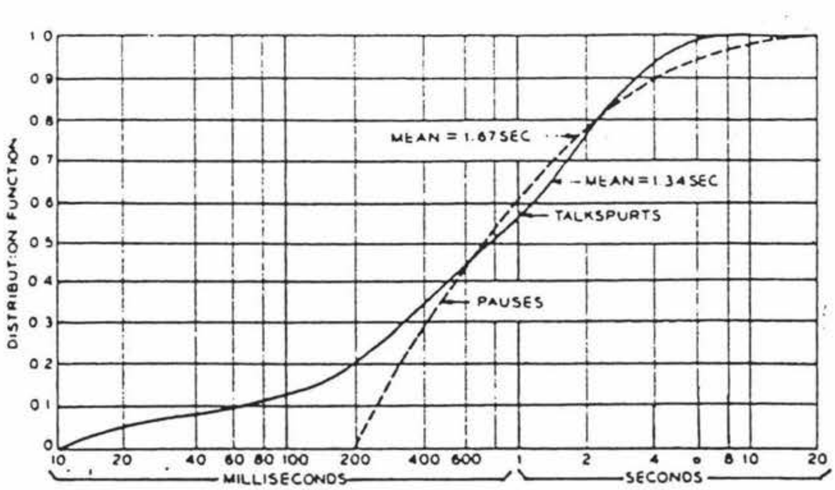


Figure 4.2.2 Brady's distribution

Let f (x) be the probability density function of some measured value of X_1, X_2, ..., X_N and Y_1, Y_2, ... Y_N which are corresponding measured abscissa and ordinate respectively. In order to generate tallkspurts and silences first a random variable is generated between 0 and 1 and would be assigned the value Y_i. This particular Y_i is then used to find corresponding Y_1, Y_2, X_1 and X_2. When all the necessary parameters are found equation 1 is used to generate talkspurt/silence duration X_i.

$$X_i = \frac{Y_i - Y_1}{Y_2 - Y_i}(X_2 - X_1) + X_1 \qquad \ldots \qquad [4.2.1]$$

If X_i is the duration of silence no packets are transmitted in that time. On the other hand if X_i is the duration of a talkspurt equation [64] is used to calculate the number of voice packets generated during that talkspurt.

Number of voice packets generated = (X_i / Frame Length) ... [4.2.2]

4.3 Discrete Event Simulation Model

A model is a representation of a system, which can be a replica, a prototype, or a smaller-scale system [67]. A model simplifies the system to a sufficiently detailed level to permit valid conclusions to be drawn about the system. Simulation is a computer representation of the real system under investigation. Simulation imitates the behavior of the system over time and provides data as if the real system were being observed.

Simulation as a modeling is attractive for the following reasons [68]:

i. It is the next best thing to observing a real system in operation.
ii. It enables the analysis of very complicated systems. A system can be so complex that its description by a mathematical model is beyond the capabilities of the analyst.
iii. It does not rely heavily on mathematical abstractions that require an expert to understand and apply.

iv. It is useful in experimenting with new or proposed designs prior to implementation. Once constructed, it may be used to analyze the system under different conditions. Simulation can also be used in assessing and improving an existing system.

In addition to handling varying levels of complexity, simulation models overcome other limitations of analytic techniques; namely, peak loads, and transient behavior can be investigated. Also, by varying design parameters, sources of performance fluctuations can be identified. Finally, the time scaling capability of simulation models allows designers to examine the performance of the modeled system for differing intervals of time.

There are two types of physical systems: continuous and discrete. In continuous systems the variables change continuously with time. The system is characterized by smooth changes in the system state. A discrete model is one in which variables change in discrete steps. Continuous simulation represents the system model by sets of algebraic or differential equations, which are solved numerically. Discrete event simulation describes the model of a real system in terms of logical relationships that cause changes of states at discrete points in time. The discrete simulation model generally tends to be both stochastic and dynamic in nature. In such an approach results are obtained by generating scheduled events at different points in time in the simulation model. In other words, the changes in the physical system are represented by a series of discrete changes or events at specific instants in time. In discrete event simulation, time and state relationships are represented in terms of event, activity, and process. These terms are defined as follows [64]:

Event: In the simulation model, a change in the state of an entity, occurring at an instant that initiates an activity. For example, initiation or termination of a call.
Activity: The state of an entity over an interval.
Process: Process is a succession of states of an entity over one or more contiguous intervals.

The dynamic behaviour of a system may be studied by tracing various system states as a function of time and then collecting the system statistics. Timing of different activities in the simulation program is maintained by an internal clock, which is incremented and maintained by the simulation program. The simulation time can be advanced in two ways, they are interval-oriented simulation and event-oriented simulation. The first method which is a uniform time increment method where the clock is advanced from t to t + δ, where δ is a uniform fixed time increment. The second method is based on variable time increment method, where the clock is incremented from time t_1 to the next event time t_2 whatever may be the value of t_2 i.e. the interval will depend on the activation of the next event. This method involves sorting of event activation time and maintaining current and future event lists. This method is mostly used in the discrete event simulation [64].

A discrete event simulation model based on event-oriented approach has been developed to study the different characteristics of the MLAN protocol in integrated traffic environment by using specific source models. Performance of the protocol was evaluated by measuring the parameters such as terminal access delay, cell end-to-end delay, and throughput in the case of data traffic, while in the case of voice traffic, parameters such as percentage of speech cell loss, terminal access delay, and throughput. These values were calculated for various bus speeds, bus lengths, terminal activity levels, and by giving priority for voice traffic. The simulation tool used is SIMSCRIPT II.5 programming language [69, 70, 71], a discrete event simulation language whose structure and features are briefly described below.

4.4 SIMSCRIPT II.5 Programming Language

SIMSCRIPT II.5 is a discrete event simulation language, which uses the process and resource concepts. The process and resource concepts are described below [71]:

4.4.1 Process Concept

A process represents an object and the sequence of actions it experiences throughout its existence in the model. There may be many instances (or copies) of a process in a simulation. There may also be many different processes in a model. A process object enters a model at an explicit simulated time, its "creation time". It becomes active either immediately or at a prescribed "activation time". From then on, the description of its activity is contained in the process routine. A process routine may be thought of as a sequence of interrelated events separated by lapses of time, either predetermined or indefinite. Predetermined lapses of time are used to model such phenomena as the service time (deterministic or stochastic), whereas indefinite delays arise because of competition between processes for limited resources. In this latter case processes will automatically be delayed until the resource is made available to it. At each activation or reactivation of the process routine, it may execute statements representing changes to the state of the system. The process routine may test for system conditions and take alternative courses of action. Processes interact either implicitly (for example, through resource competition) or explicitly (through executing statements to "activate", or "interrupt", or "resume" one another).

4.4.2 Resource Concept

Resources are the passive elements of a model. A resource is used to model an object, which is required by the process objects. If the resource is not available when required by the process objects. If the resource is not available when required, the process object is placed in queue or waiting line and made to wait until the resource becomes available. A resource becomes available when the process holding it "relinquishes" it. The first of the process objects in the queue is then given the resource and reactivated. If a resource is relinquished when no process object is waiting for it, it is merely made available to be allocated when requested.

4.5 Description of Simulation Model

The simulation model is shown in figure 4.5a, which consists of various modules is briefly described now. The module MAIN controls the simulation. At the start of the simulation, the necessary variables are read from the input and assigned to appropriate global variables by the routine READTALK.

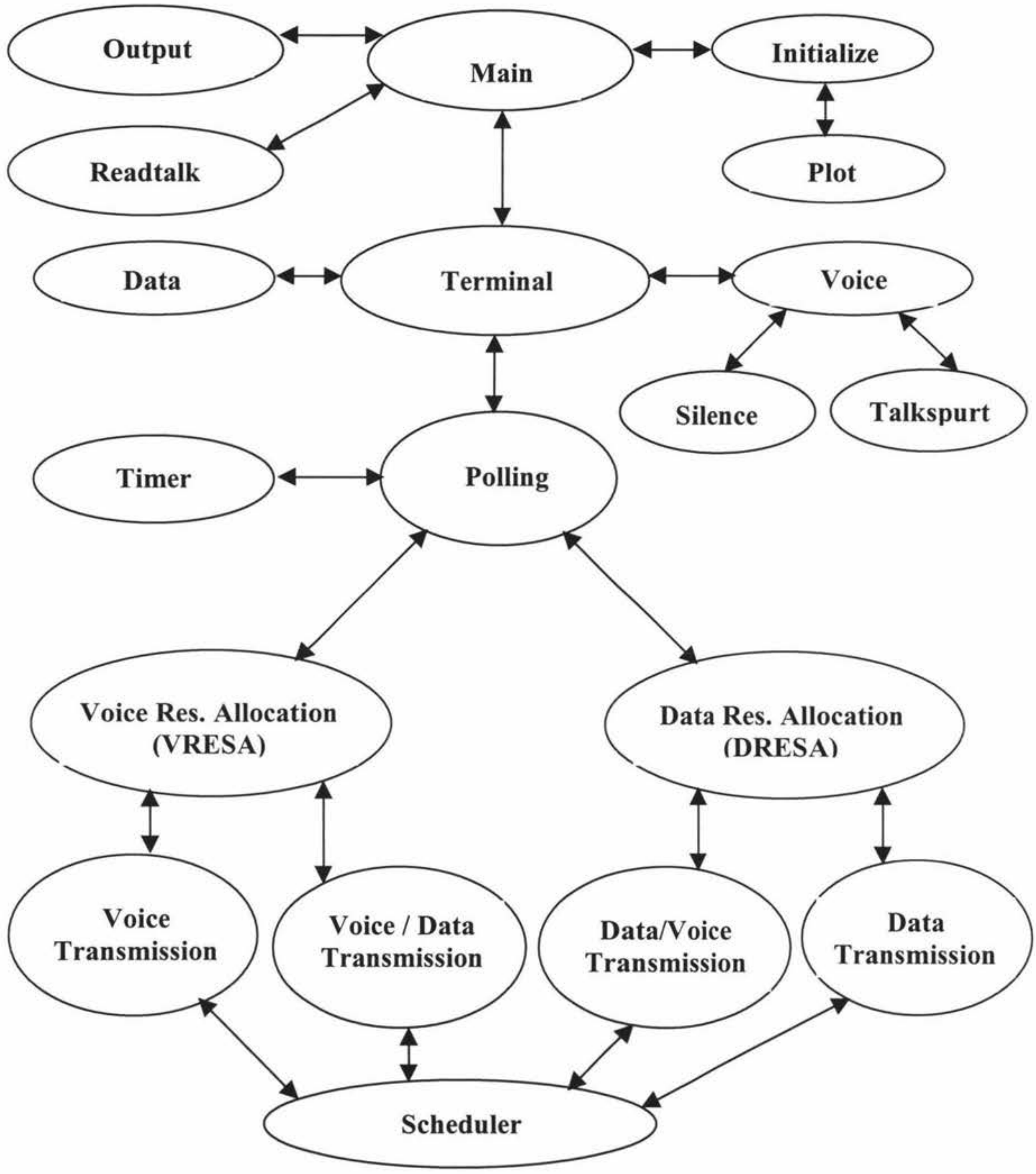


Figure 4.5a Block diagram of simulation model

The corresponding global variables are then initialized in the routine INITIALIZE. Variables such as number of slots per frame, number of polling slots per frame, number of transmission slots per frame etc. are also defined in this routine. Further, routine INITIALIZE also activates processes TERMINAL, TIMER, and PLOT. Process TIMER shown in figure 4.5b generates frame timing with 6ms duration. This process therefore, provides the current and next reservation slots at any given time during the simulation. This process also activates the process POLLING, which is performed during the start of the frame soon after the broadcast of the control cell information.

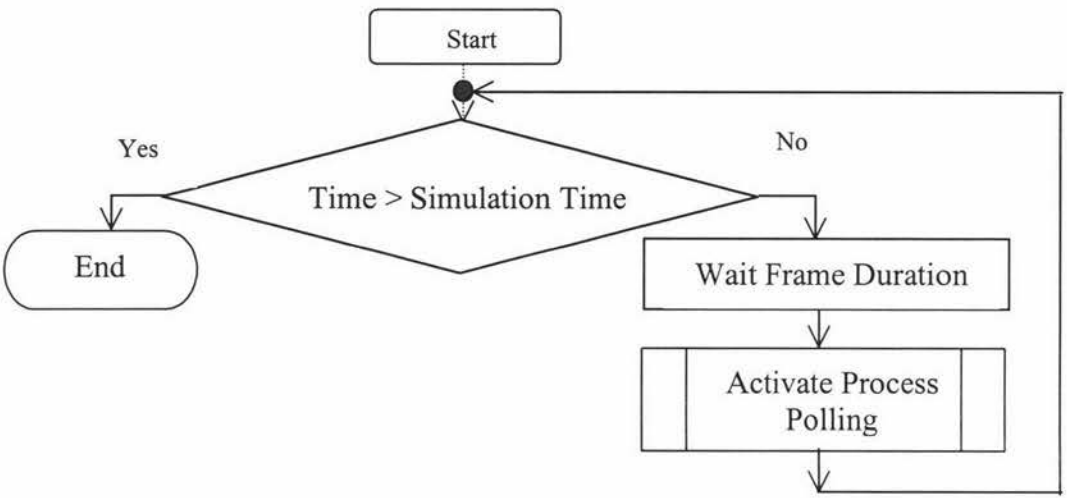


Figure 4.5b Flow diagram of the Process Timer

Process PLOT is used to update some cell information regularly at a pre-defined interval. Process TERMINAL models the traffic generators. Process POLLING implements part of the MLAN protocol. Process POLLING also activates Process VRESA and DRESA. Process VRESA calls routines Voice Transmission and Voice/Data Transmission while process DRESA calls Data Transmission and Data/Voice Transmission. All Transmission routines call routine SCHEDULER, which in turn specifies the slot number in which the terminal is transmitting and also synchronizes the terminal's transmission time with that of the system's clock. In order to simulate the operation of MLAN with multimedia traffic, the important modules are TERMINAL, POLLING, VRESA, DRESA, and the Transmission routine. These modules are described below.

4.5.1 Process TERMINAL

The process TERMINAL is initiated by the routine initialize and it would continue until the end of simulation. The process is shown in figure 4.5.1. At the beginning of the process it selects the type of terminal based on the ID number of the terminal. This is done to generate different types of data and speech terminals. If it is a speech terminal then the routine SILENCE is called which returns a silence period. The terminal would then remain in a wait state for the period of time returned by the routine SILENCE. After the waiting period the routine TALKSPURT is called which returns the talkspurt duration (talkspurts and silences were generated using Brady's distribution [67] described in section 4.2.1 and two random number generators). Then the talkspurt duration is used to calculate the number of cells generated from a talkspurt as per equation [4.2.2]. If the terminal is a data terminal, initially it waits for an exponential waiting time with a mean of T seconds. Then the appropriate number of data messages is generated using negative exponential distribution. This value is then multiplied by a factor of 1000000 (randomly selected) and then divided by 48 bytes to get the number of data cells needed to be transmitted. Irrespective of whether it is a data/speech terminal, after the number of cells that are to be transmitted are calculated, the corresponding terminal is filed in the Waiting.Q and the TERMINAL process is then suspended. The TERMINAL process is again reactivated after the corresponding terminal completes the transmission of its final cell.

4.5.2 Process POLLING

This process is shown in the figure 4.5.2. This process is initiated by Process TIMER. This process models the major part of MLAN protocol. At the beginning of the process it waits for the control cell to be broadcasted and then removes each terminal from the Waiting.Q. It then checks for the type of terminal whether it's a speech terminal or a data terminal and files the corresponding terminal in the Voice.Q or Data.Q depending upon the type of terminal. It then activates the corresponding resource allocation process VRESA or DRESA depending upon the type of the terminal. Although VRESA and DRESA are part of POLLING, they are considered as separate processes for programming simplicity.

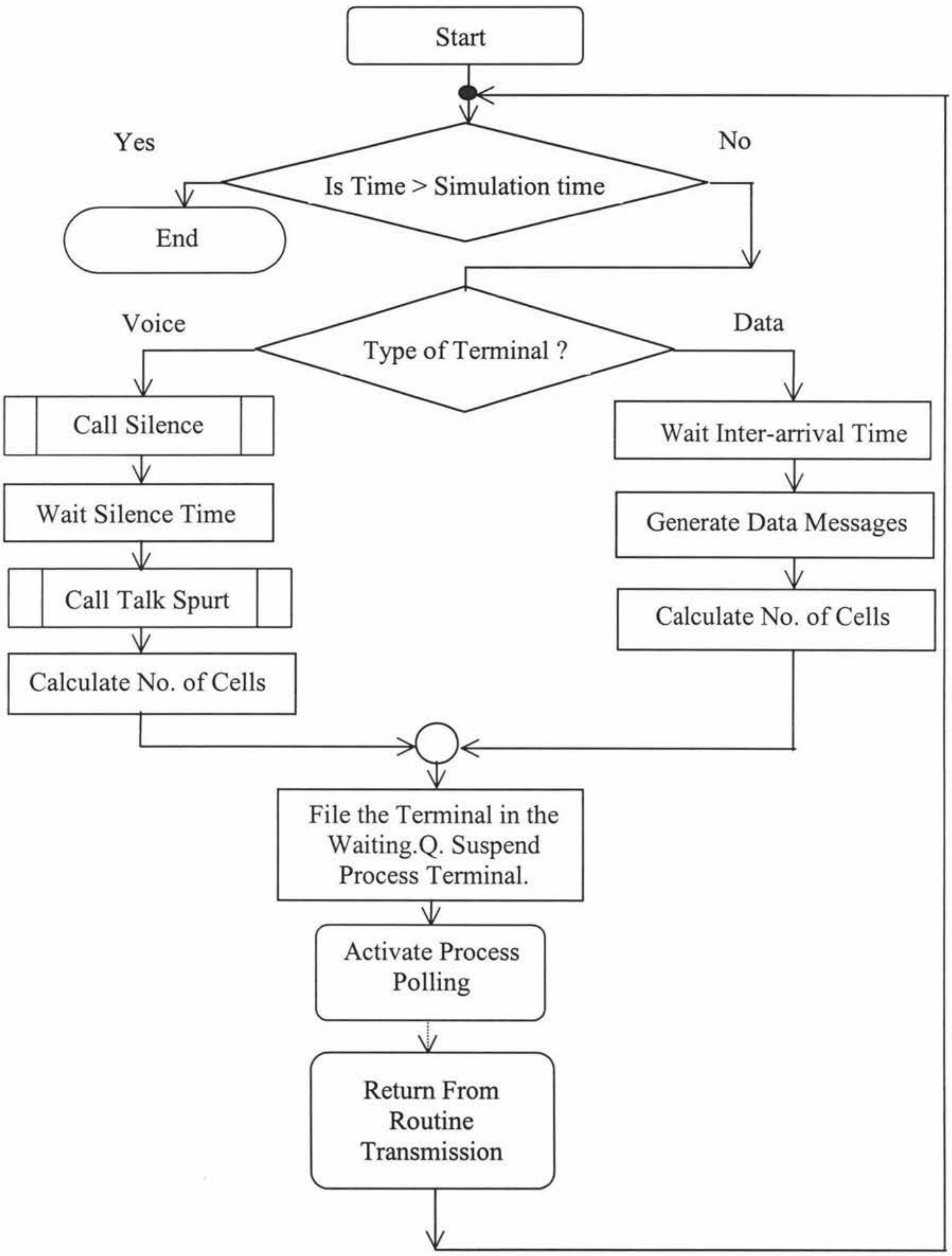


Figure 4.5.1 Flow diagram of the Process Terminal

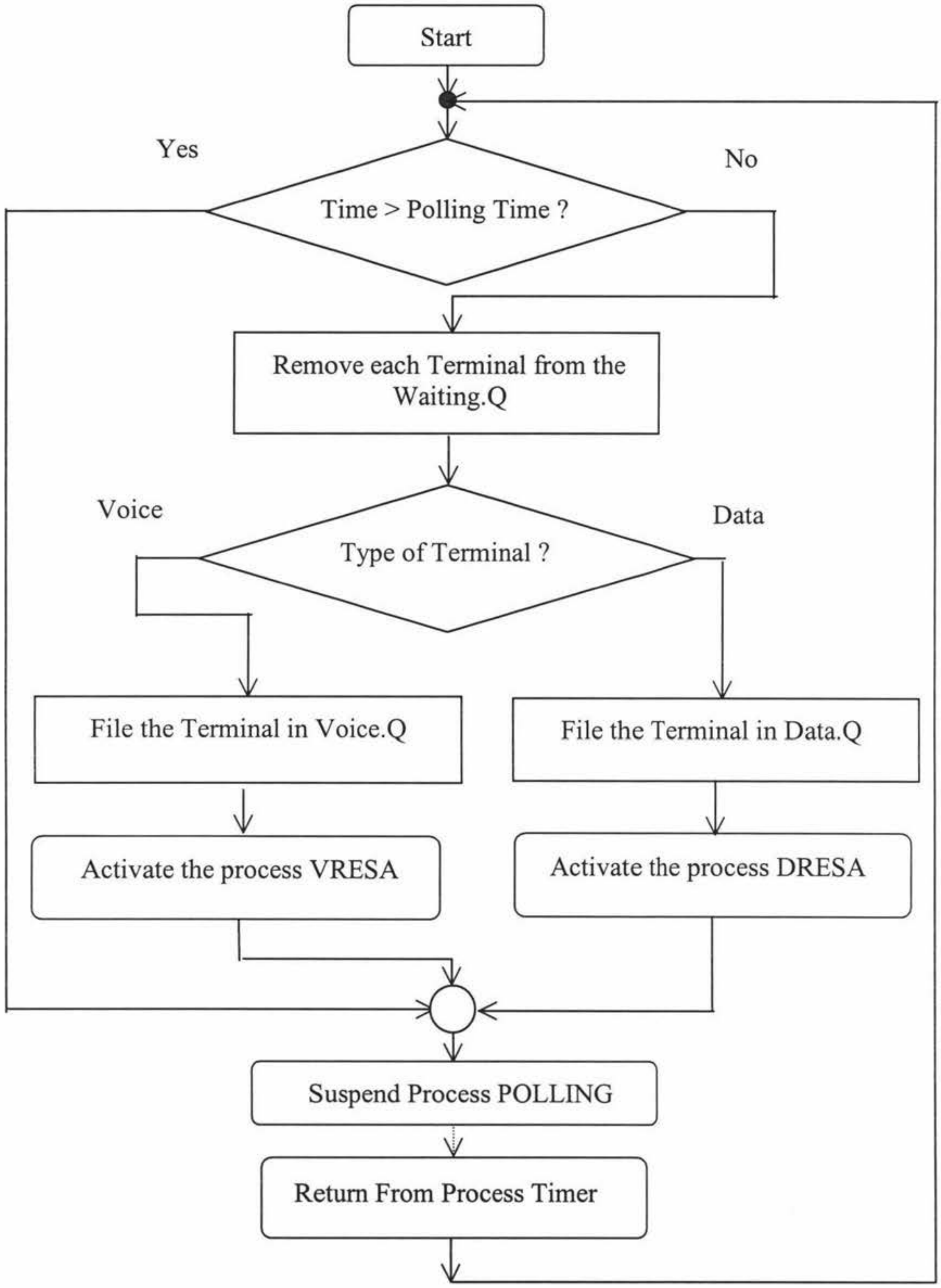


Figure 4.5.2. Flow diagram of the process POLLING

4.5.2.1 Process VRESA

This process is shown in figure 4.5.2.1. At the beginning of the process it checks for the availability of a free Voice slot. If it is available, it then allocates the first available free voice slot to the speech terminal and then sets the corresponding voice slot and decrements the voice slot counter. If a free voice slot is not available, it then checks for a free voice/data slot which if available allocates the first available such slot to the speech terminal and the corresponding slot is locked and decrements the voice/data slot counter. It then calls routine Voice Transmission and the process VRESA is suspended. If a free voice slot and a free voice/data slot is not available, then the process checks for the cell dropping threshold which if exceeded adds 1 to lost cells counter and subtracts 1 from the number of cells of the terminal. The number of the cells of the terminal is then checked. If the number of cells terminal is zero the corresponding terminal is removed from the voice.Q and the terminal is reactivated. The process is then suspended. If the number of cells of the terminal is not zero or the cell dropping threshold is not exceeded the corresponding terminal is removed from the voice.Q and filed in the waiting.Q. The process is then suspended.

4.5.2.2 Process DRESA

This process is shown in figure 4.5.2.2. At the beginning of the process it checks for the availability of a free data slot. If it is available, it then allocates the first available free data slot to the data terminal and then sets the corresponding data slot and decrements the data slot counter. If a free data slot is not available, and if there is no speech terminal either waiting in the voice.Q, or waiting in the waiting.Q, it then checks for a free voice/data slot which if available allocates the first such slot to the data terminal and sets the corresponding slot and decrements the voice/data slot counter. It then calls routine Data Transmission and the process DRESA is suspended. If a free data slot and a free voice/data slot is not available, the data terminal waits in the Data.Q until a data slot or a voice/data slot is available.

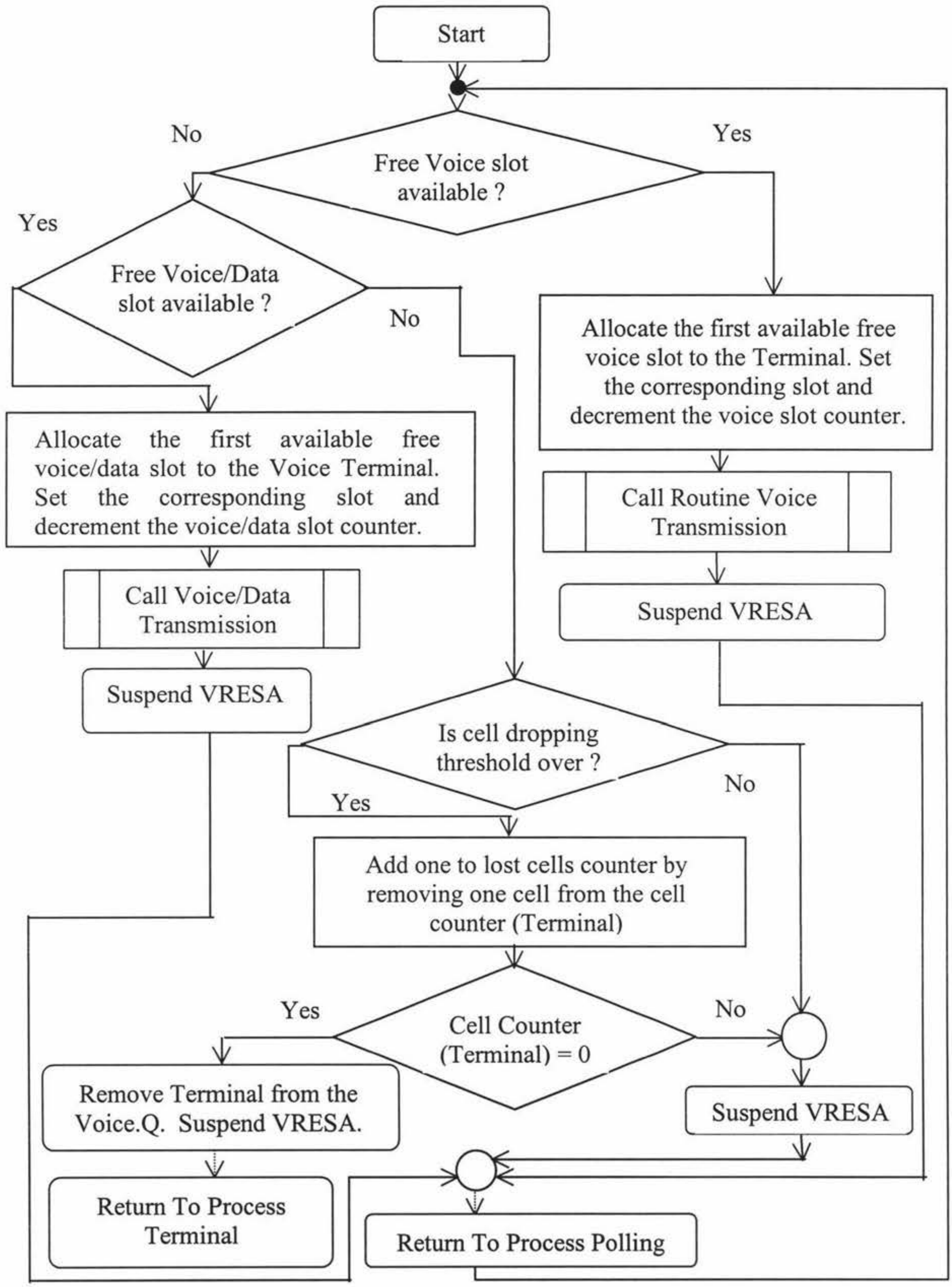


Figure 4.5.2.1 Flow diagram of the Process VRESA

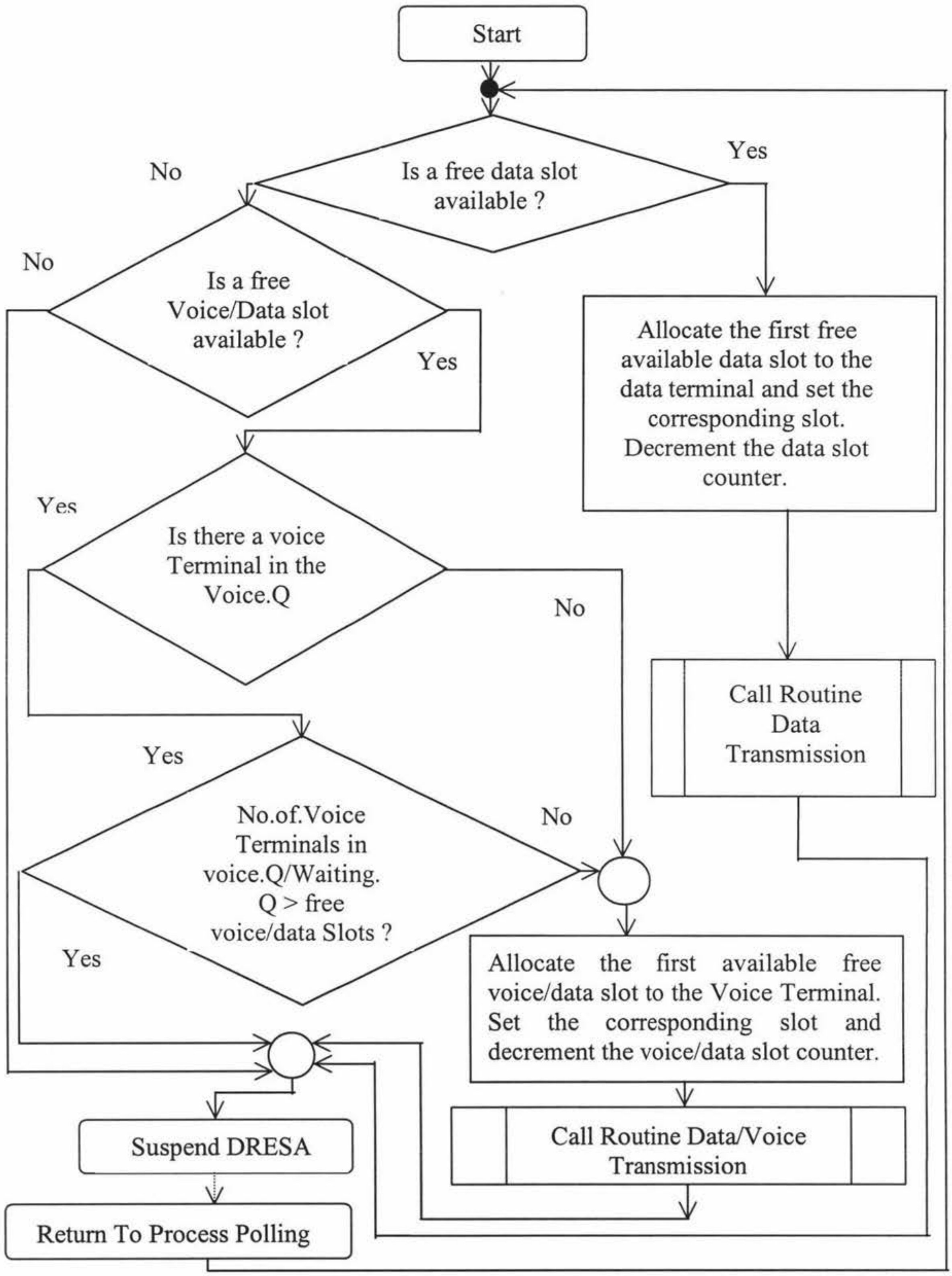


Figure 4.5.2.2 Flow diagram for the Process DRESA

4.5.3 Routine Transmission

This routine shown in figure 4.5.3 is called from the process VRESA or DRESA. Once a terminal is allocated a slot this routine is called. In this routine cell transmission is initiated with one cell per frame for each terminal that has gained transmission access and it continues until the end of the final cell of the terminal completes its transmission. Before the cell transmission is initiated, it calls routine SCHEDULER, which in turn specifies the slot number in which the terminal is transmitting and also synchronizes the terminal's transmission time with that of the system's clock. Once a terminal completes the transmission of its last cell, the corresponding slot is released, the corresponding slot counter is incremented, and the terminal is reactivated.

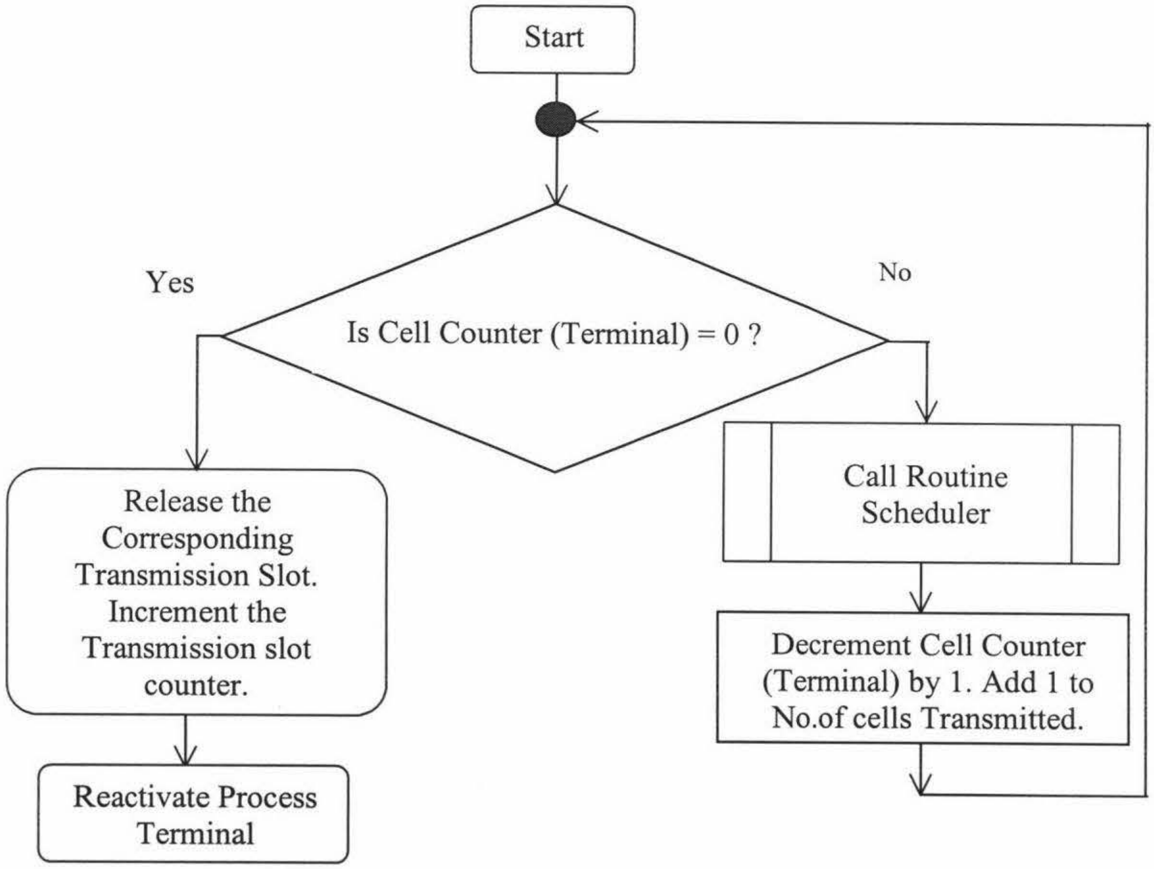


Figure 4.5.3 Flow diagram of the routine Transmission

4.6 Validation of the Simulation Model

In order to validate the simulation model one needs to validate the traffic generators. The data traffic generation is validated by gathering different traffic patterns at different interarrival times and verifying those traffic patterns one with the other. The voice traffic generation is validated by measuring the duration of talkspurt and silence and the voice activity and comparing these with the Brady's distribution values [65]. Then the MLAN protocol would be validated by simulating the protocol using the data and speech traffic models and then comparing with similar results published in [9].

4.6.1 Validation of the Data Traffic Generator model

In order to validate the data traffic generator, the data traffic generator model was simulated for 1000 simultaneous data terminals for a period of 300 seconds with a warm-up time of 100 seconds. The remaining time of 200 seconds was used to gather statistics. The simulation was carried with different interarrival times and the traffic plot is shown in figure 4.6.1. It may be observed from the figure 4.6.1, for smaller interarrival times the generated data volume is higher than the one with longer interarrival time which showsthe data traffic generator is operating according to design.

4.6.2 Validation of the Voice Traffic Generator model

In order to validate the voice traffic generator, the model was simulated for 30 simultaneous voice conversations for a period of 300 seconds. The simulation was carried for different sets of random number generated values. A random number known as "Talkseed" was used to generate talkspurts while "Silseed" was used to generate silences. Table 4.6.2 shows the duration of talkspurt and duration of silence along with the ratio of total talkspurts to the sum of total talkspurts and silences for different sets of "Talkseed" and "Silseed" combinations. These results indicate that the voice activity changes between 44% to 46%, which correspond to Brady's distribution [65]. The traffic plot of voice traffic generator is shown in figure 4.6.2.

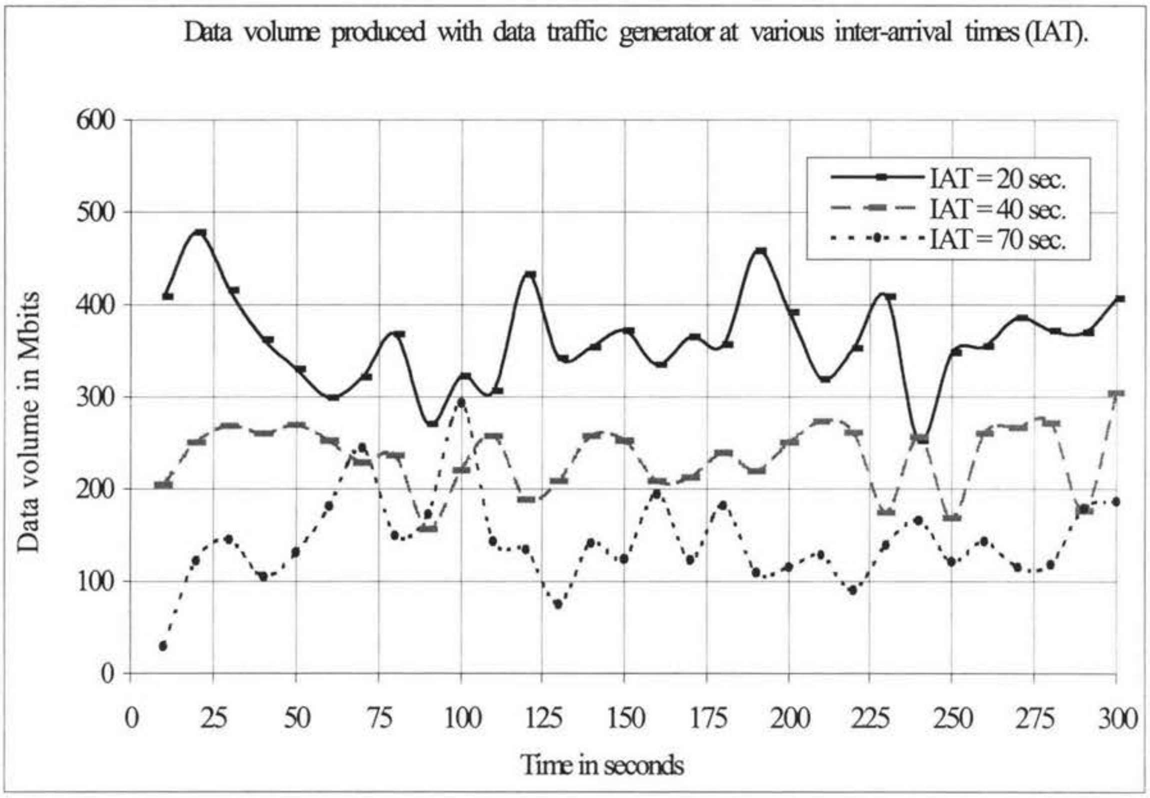


Figure 4.6.1 Traffic plot of the data traffic generator at different inter-arrival times.

Table 4.6.2 Durations of Talkspurt and Silence, and Voice activity with different random number generator values for a simulation length of 300 seconds with 30 simultaneous voice calls.

Talkseed, Silseed	**Talkspurt duration (Sec)**	**Silence duration (Sec)**	**% of Voice activity**
1,2	1.728	1.38	44.4
2,3	1.726	1.39	44.6
4,5	1.70	1.387	44.89
6,7	1.67	1.392	45.46
8,9	1.632	1.39	46.04

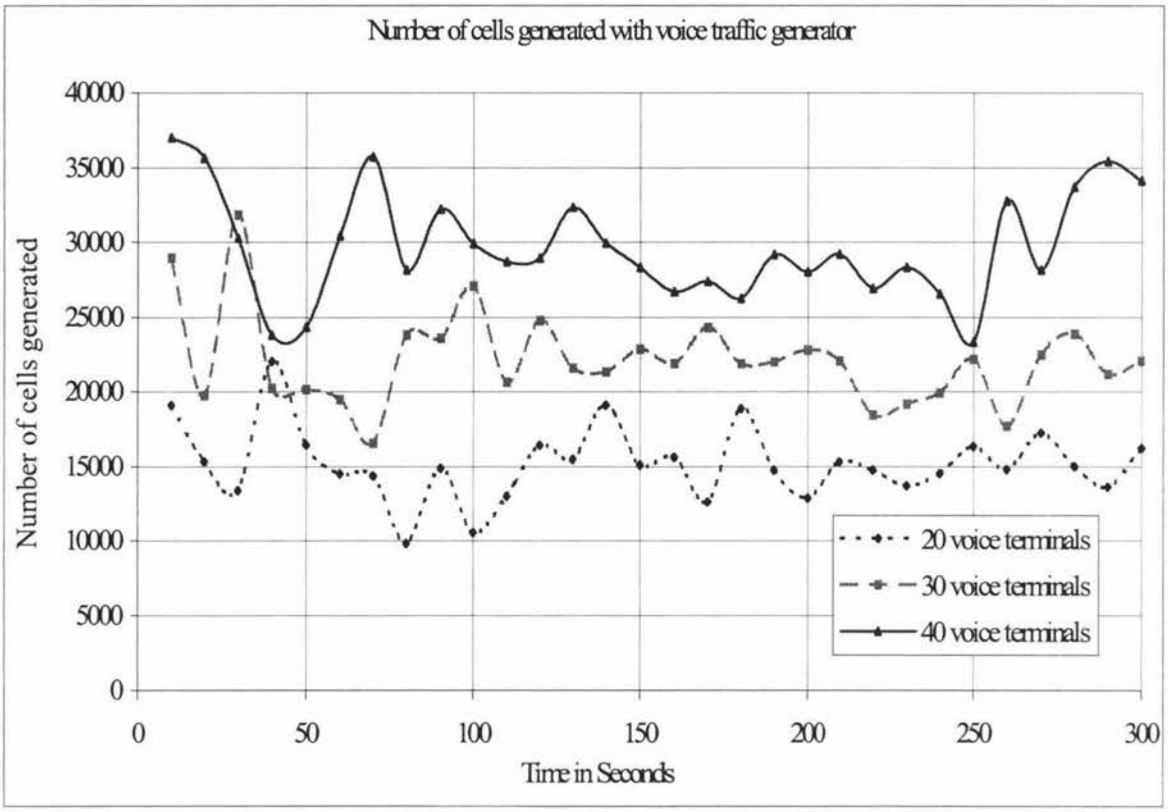


Figure 4.6.2 illustrates the traffic plot of the voice traffic generator.

4.7 Validation of the MLAN Protocol model

The MLAN protocol has been simulated using data and voice traffic source models. Validation of the developed simulation model will be done by comparing the obtained results with published in [9]. The results used for comparison are throughput, and mean cell delay. For speech traffic, 1% of speech packet loss has been used as the performance criteria and for the data traffic end-to-end delay has been used as the performance criteria. Some of the simulation parameters used in the simulation for this are listed in table 4.7.

Table 4.7 illustrates the simulation parameters for validation using data and voice traffic generators

Simulation Parameters	Value(s)
Bus length.	1 km, 2 km and 3 km.
Bus speed.	100Mbps and 150Mbps.
Propagation delay.	5μs/km.
Number of Polling slots.	60.
Frame duration.	6 milliseconds.
Simulation time.	300 seconds.
Warm-up time.	60 seconds.
Terminal Activity	0 – 1 (i.e. 0 – 100%)
Speech cell dropping threshold	6ms

Before evaluating the performance of MLAN protocol for various parameters such as terminal activity, average queue length, average access delay, mean cell delay, and throughput which are defined as follows:

4.7.1 Terminal activity

Terminal activity is defined as the percentage of time a terminal is busy serving a call to the total time.

4.7.2 Average queue length

It is defined as the total number of terminals of a particular type (data or voice) waiting in the corresponding queue for a transmission slot at any given time.

4.7.3 Queuing delay

It is defined as follows:

$(T_{TS} - T_{CG})$ ... [4.7.3]

Where, T_{TS} specifies the time a terminal is allocated a transmission slot and T_{CG} specifies the time the corresponding terminal's call has been generated. In this case it may also be referred to as access delay.

4.7.4 Mean cell delay (ms)

The mean cell delay is defined as follows:

$(T_{Dept.} - T_{Arriv.})/N_{CT}$ ... [4.7.4]

Where $T_{Dept.}$ is the termination time of the current burst (for data) or talkspurt (for voice) of the terminal;

T_{Arriv} is the burst or talkspurt arrival time of the terminal;

N_{CT} is the number of transmitted cells of the current burst or talkspurt of the terminal ;

It may be noted that departure time of the terminal is referred to as the time the last cell of the terminal's burst or talkspurt is transmitted.

4.7.5 Normalized Throughput

Throughput is defined as follows [72]:

Throughput = Number of packets transmitted over a time.

However, the normalized throughput is defined as follows:

$(C_{NC} * T_S)$ / (Simulation time – warm-up time) ... [4.7.5.1]

Where C_{NC} represents the total number of cells transmitted after the warm-up time, T_S represents the transmission slot duration which is defined as follows:

T_S = (ATM cell size/BIT.RATE)*1000 + T_{PD} (ms) ... [4.7.5.2]

Where T_{PD} is the propagation delay associated with the bus length.

The propagation delay T_{PD} is given by

$T_{PD} = T_{PK}$ * L/1000 ... [4.7.5.3]

Where L is the length of the bus in meters and T_{PK} is the propagation delay/km, which is taken as 5 microseconds per kilometer since the propagation speed in vacuum is the speed of light which is about 3.33 μs/km. It is lower in other media depending upon their dielectric constant.

Number of transmission slots per frame = Frame duration/T_S ... [4.7.5.4]

Since propagation delay is directly proportional to bus length L, therefore, propagation delay increases with increase in bus length. Hence, transmission slot duration T_S increases (with the frame duration and the bus speed remaining constant) with increase in bus length. Increase in bus length gives rise to lower number of transmission slots as the number of transmission slots per frame are equal to the ratio of frame duration to the transmission slot duration as shown in equation [4.7.5.4]. From here onwards the normalized throughput is always referred to as throughput.

4.7.6 Efficiency

Efficiency is defined as follows:

Efficiency = (BWU_{cif})/ (TABW) ... [4.7.6]

Where BWU_{cif} is the bandwidth used to carry information;
TABW is the total available bandwidth.

The throughput of MLAN has been evaluated against
*Number of terminals supported
*Bus length
*Terminal activity

4.8 Discussion of Simulation results of MLAN with Data Traffic Generator

Some of the major factors that determine the efficiency of a protocol are as follows:

i. Throughput
ii. Average packet/cell delay
iii. Effect of data transmission rates used on throughput and average packet delay
iv. Effect of propagation delay
v. Efficiency

Performance factors of MLAN protocol which are the average queue length, access delay, mean cell delay, and throughput were obtained from the simulation model by varying number of terminals, bus length, and terminal activity levels. Simulation results are discussed in the following sections.

4.8.1 Terminal Activity

Terminal activity is varied by varying the inter-arrival time of the terminal call. The terminal activity with 48.4% and 24.2% are plotted and shown in figure 4.8.1.1a and 4.8.1.1b respectively.

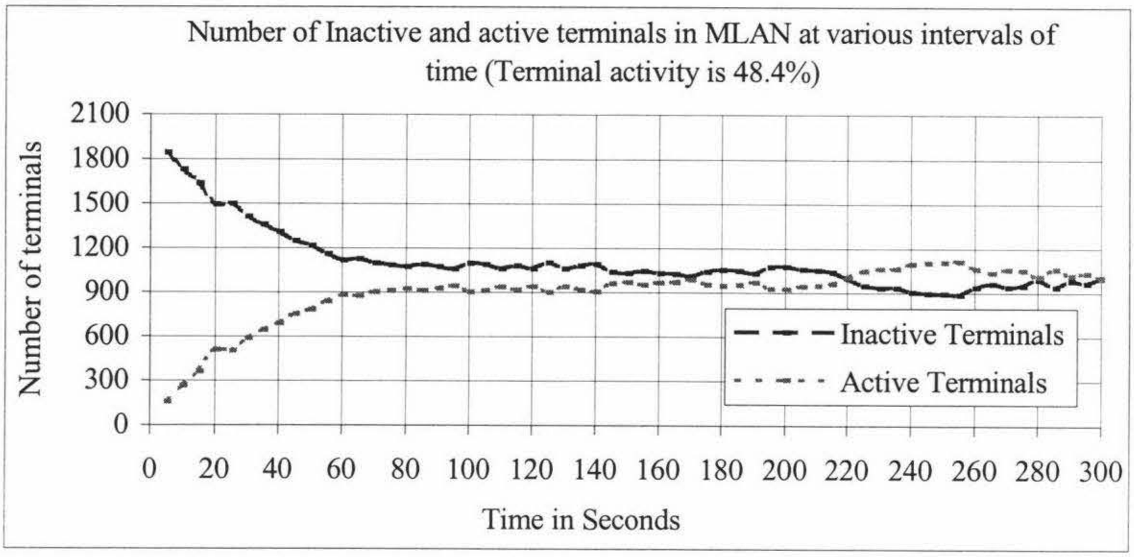


Figure 4.8.1.1a illustrates a terminal activity of 48.4% with data terminals.

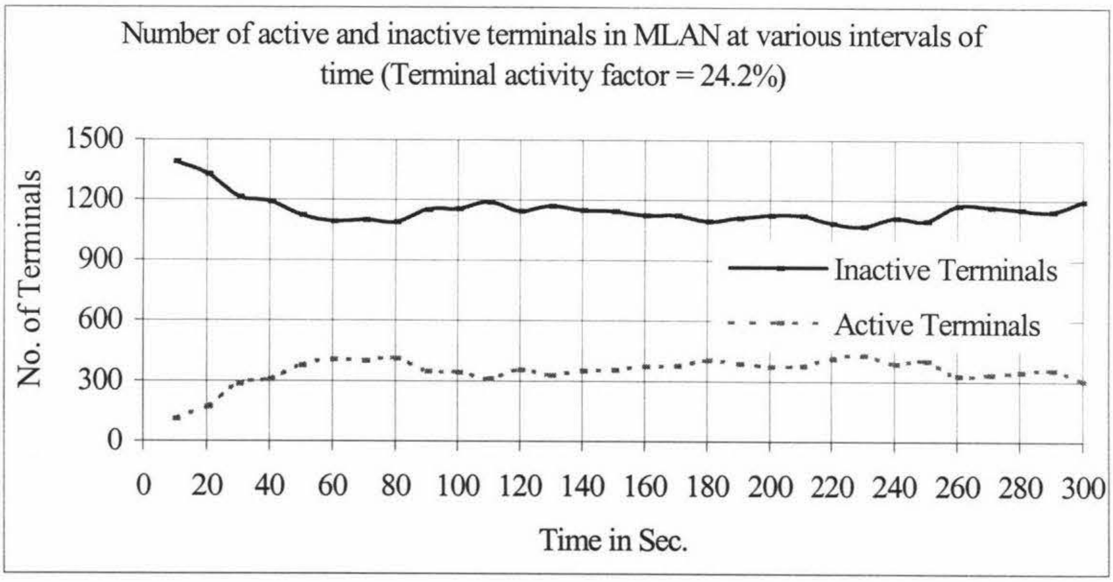


Figure 4.8.1.1b illustrates a terminal activity of 24.2% with data terminals.

It may be observed from figure 4.8.1.1a that the average terminal activity is about 48.4%, which indicates that on an average (after the warm-up time), about 968 terminals are always active. Active means that these terminals have cells to transmit. Similarly figure 4.8.1.1b illustrates that the average terminal activity is 24.2%, which implies that on an average (after the warm-up time) about 387 terminals are always active.

The throughput of MLAN, queuing delay and the corresponding mean cell delay are plotted against terminal activity and is shown in figure 4.8.1.2.

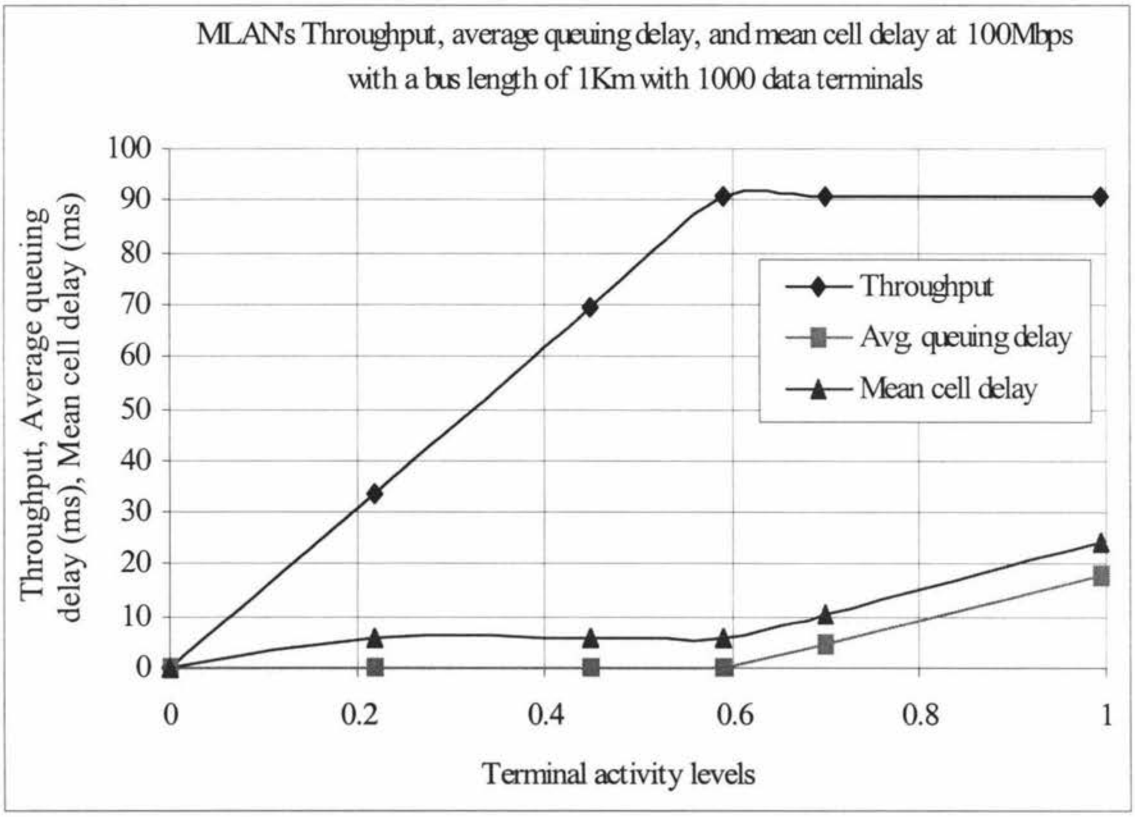


Figure 4.8.1.2 illustrates MLAN's throughput, average queuing delay and mean cell delay with data traffic generator at various terminal activity levels.

At a bus length of 1 km and bus speed of 100Mbps, it gives rise to 590 transmission slots apart from the polling overhead and other overheads for transmission headers etc. It may be observed from figure 4.8.1.2, the throughput increases linearly with increase in terminal activity up to a terminal activity of 0.59and above this terminal activity the throughput remains constant. This is because of the fact that at a terminal activity of 0.59 and above, with 1000 terminals at least 590 terminals remain active during simulation time. Above a terminal activity of 0.59, more than 590 terminals are always active. However, since more than 590 terminals can not be accommodated at any time on the bus, the throughput remains constant above this terminal activity. It may be observed that above a terminal activity of 0.59, the remaining terminals, which have cells to transmit wait in the queue for a transmission slot. It means that these cells wait in the buffers. Consecutively, average queuing delay and mean cell delay also increase exponentially as the traffic model is based on negative exponential distribution.

The overhead is calculated as follows:

The overhead in MLAN consists of the following parts:

i. C bits for control cell.
ii. ($P*N_I$) bits for polling where P is the polling field width and N_i the length of the inactive queue
iii. $T*N_T$ bits for transmission headers where T is the header width and N_T is the number of terminals transmitting.
iv. Polling cycle duration of ($50*T_S$).

At a bus length of 1 km, the total transmission slots are 649. With 590 terminals transmitting, at 100% load, the overhead apart from the polling cycle duration as follows:

C = 424 bits, i.e. 1 ATM cell.

$P*N_I = 5*(0.05)*590$, i.e. 150 bits, taken as 1 ATM cell.

$T*N_T = (4*590)$, i.e. 2360 bits, which is about 6 ATM cells. The total overhead is taken as 59 transmission slots (59 ATM cells).

Since, the polling cycle duration is 50*T_S, increase in bus length leads to increase in T_S, thereby reducing the total number of transmission slots, which reduces the throughput.

The number of terminals that can be supported in MLAN at various terminal activity levels at a bus length of 1 km with a bus speed of 100Mbps is shown in figure 4.8.1.3. From figure 4.8.1.3, it may be observed that at about 25% terminal activity, the MLAN supports about four times more number of terminals with zero queuing delay.

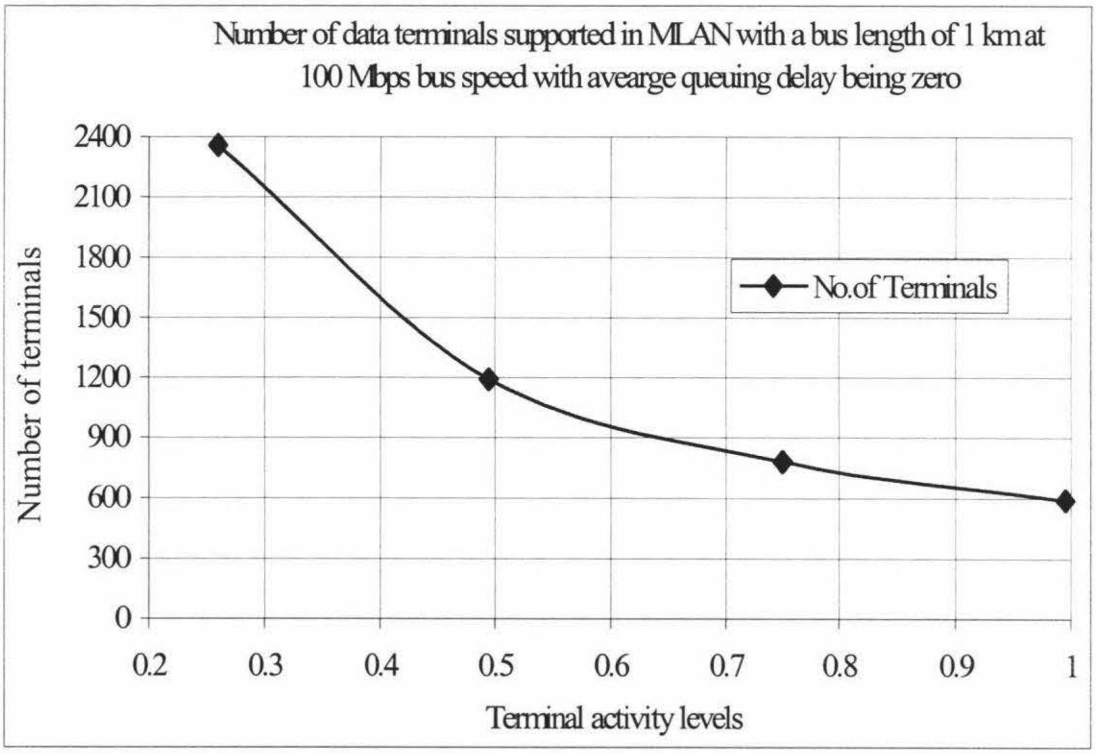


Figure 4.8.1.3. Illustrates the number of terminals supported at various terminal activity levels at a bus speed of 100Mbps with 1km bus length.

4.8.2 Average queue length

Figure 4.8.2 illustrates the average queue length of data terminals at various bus lengths of 1and 3 km with a bus speed of 100Mbps. At a bus speed of 100Mbps and bus length of 1 km the number of available transmission slots are 590.

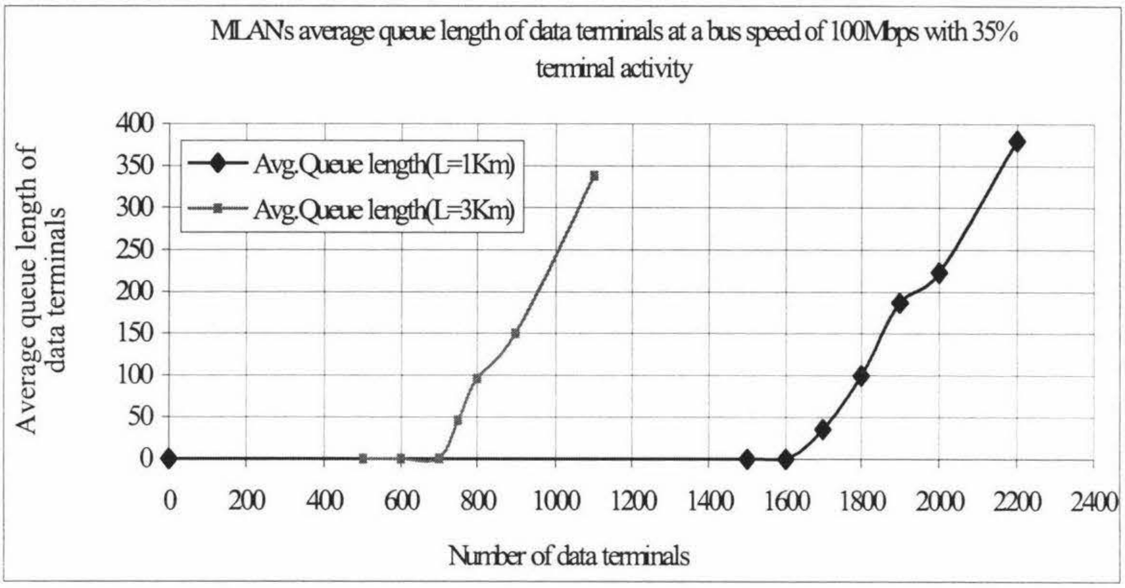


Figure 4.8.2 illustrates average queue length at various bus lengths.

It may be observed from figure 4.8.2, that the average queue length remains zero to about 700 at a bus length of 3 km and 1600 terminals at a bus length of 1 km respectively. Hence, with a terminal activity of 35%, total number terminals that can be supported are about 700 at a bus length of 3 km and 1650 terminals at a bus length of 1 km. It implies that the MLAN supports about 2.75 times the number of terminals at 35% terminal activity. Average queue length increases once, the number of terminals requiring a transmission slot are more than the number of available transmission slots. Average queue length also increases with increase in bus length. This is because of the fact that the number of available transmission slots decrease with increase in bus.

4.8.3 Queuing delay

Figure 4.8.3 illustrates average queuing delay of data terminals at various bus lengths of 1 and 3 Km with a bus speed of 100Mbps. It may be observed from figure 4.8.3, average queuing delay is zero up to 1600 terminals at a bus length of 1km which is about 2.75 times the number of transmission slots available at 100Mbps.

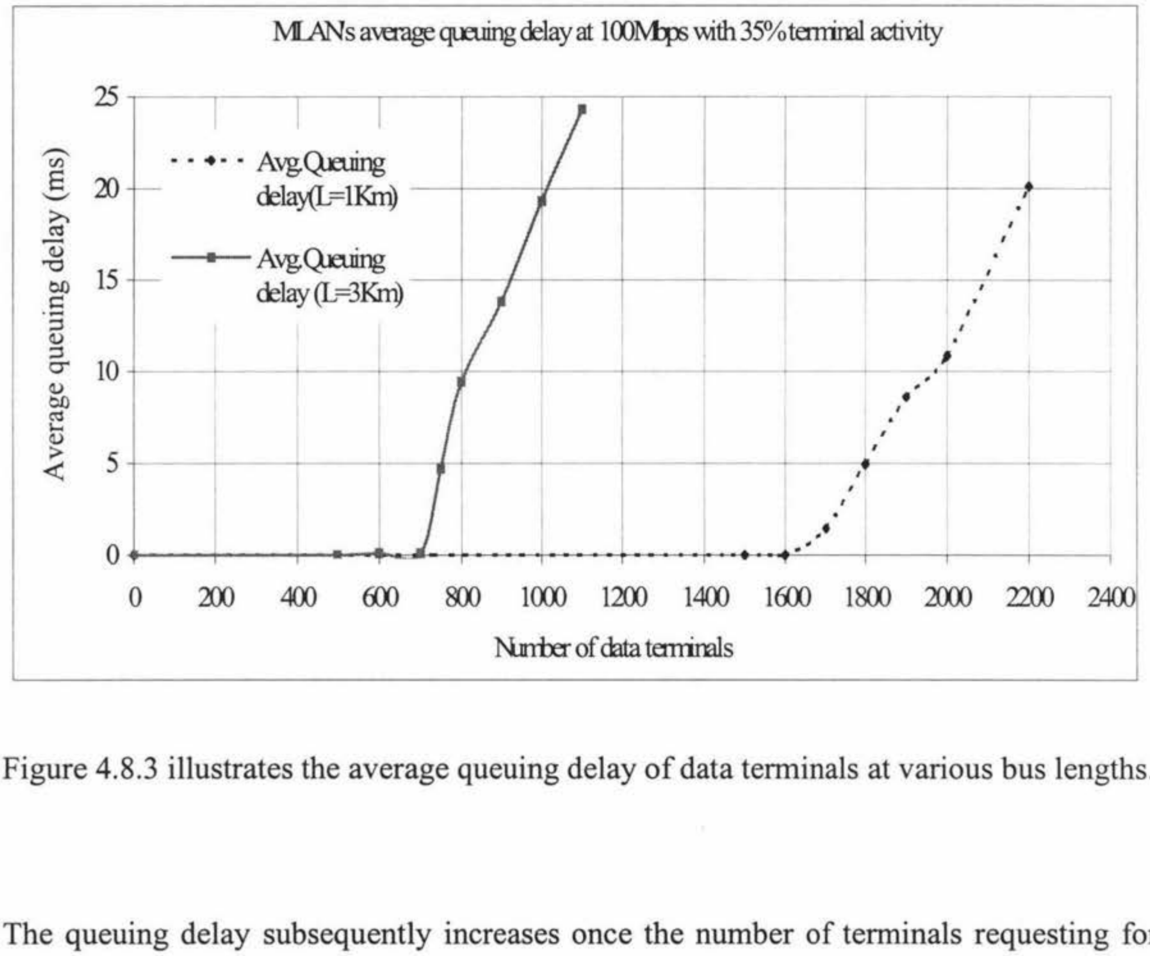


Figure 4.8.3 illustrates the average queuing delay of data terminals at various bus lengths.

The queuing delay subsequently increases once the number of terminals requesting for a transmission slot are more than the number of available transmission slots. This is once again because of the fact that the number of transmission slots available is less than the number of terminals waiting in the queue requiring a transmission slot.

4.8.4 Mean cell delay (ms)

Figure 4.8.4 illustrates the mean cell delay of data terminals at various bus lengths of 1and 3 Km with a bus speed of 100Mbps. The mean cell delay is equal to the sum of frame duration, queuing delay and the propagation delay.

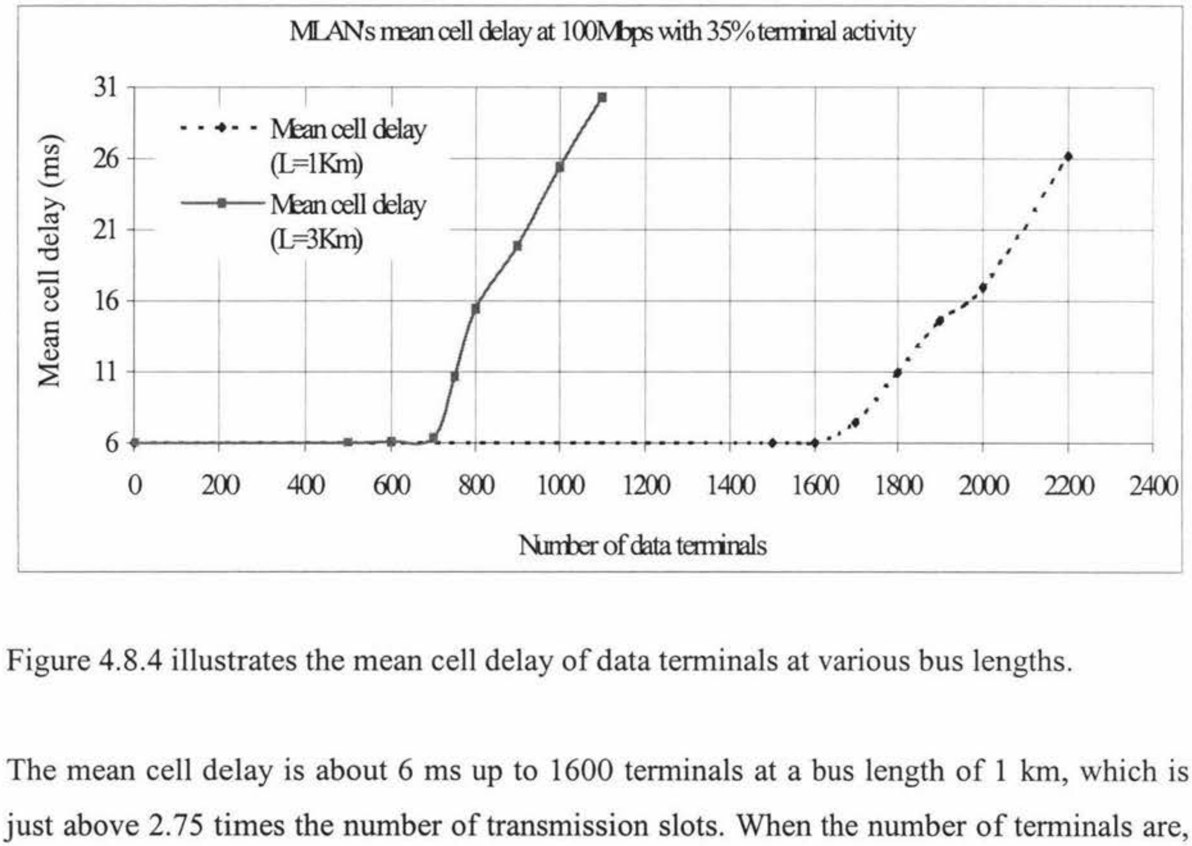


Figure 4.8.4 illustrates the mean cell delay of data terminals at various bus lengths.

The mean cell delay is about 6 ms up to 1600 terminals at a bus length of 1 km, which is just above 2.75 times the number of transmission slots. When the number of terminals are, subsequently increased, the mean cell delay also increased. This is once again because of the fact that the number of transmission slots available is less than the number of terminals waiting in the queue requiring a transmission slot which means that the corresponding cells of the terminal are waiting in the buffers. Mean cell delay also increases with increase in bus length.

4.8.5 Throughput

The throughput of MLAN protocol is calculated for data traffic with 35% terminal activity and is shown in figure 4.8.5.1.

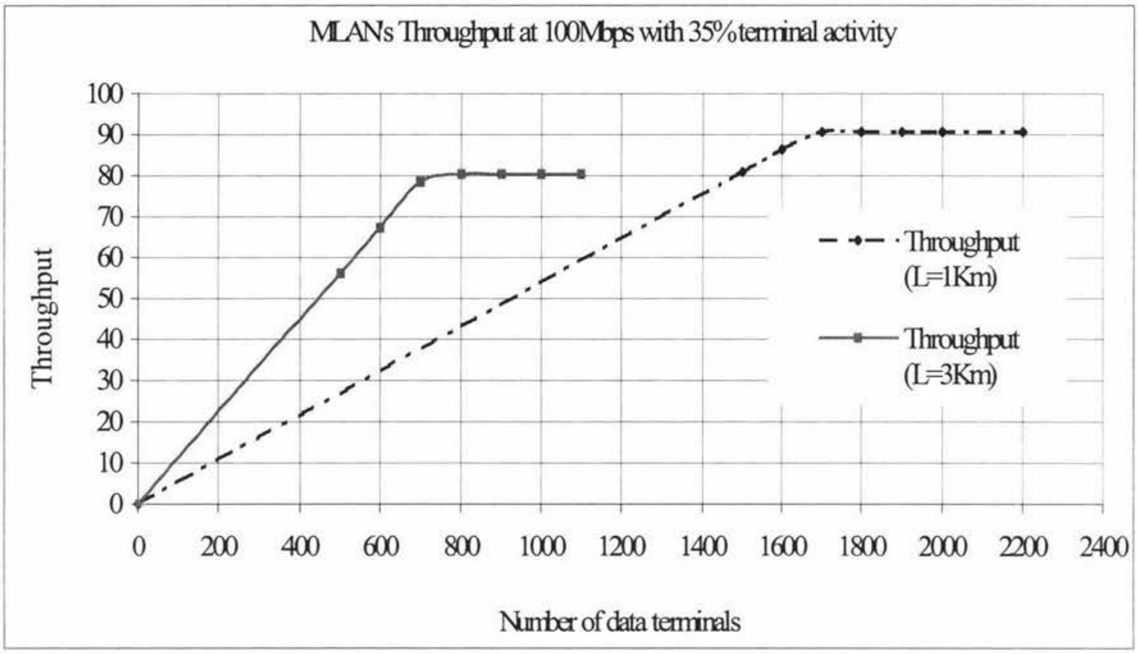


Figure 4.8.5.1 illustrates the throughput at various bus lengths.

From the figure 4.8.5.1, it can be observed that the **throughput** of MLAN (measured for data traffic model) increases linearly with increase in the number terminals. At a bus length of 1 km, the total transmission slots are 590 (apart from 50 polling slots and 9 slots for other overhead such as control cell and header etc.). Hence, with 1600 terminals at 35% terminal activity, corresponds to about 560 terminals being active at any instant of time. This gives rise to a throughput of about 86%. Once the number of terminals become 1700, the throughput reaches 90.8% and then remains constant thereafter, as by then, all the 590 transmission slots are completely utilized by the terminals. Any further increase in the number of terminals would not increase the throughput as all the transmission slots are completely utilized by all the terminals. The linear increase is because of the fact that the number of transmitting terminals is less than the number of available transmission slots and each transmitting terminal is allocated only one transmission slot. However, the throughput decreases with increase in bus length. It may also be observed that the throughput appears to be higher at a longer bus length for a lower number of terminals. This is because of the fact that for a longer bus length the number of available transmission slots are less than those at a lower bus length. Hence, for smaller number of transmission slots to be fully utilized requires smaller number of terminals.

The throughput decreases from 90.8% at a bus length of 1 Km to 80.45% at a bus length of 3 Km and these are in accordance with the results published in literature [9]. This is because of the fact that the throughput depends upon the number of transmission slots and the number of transmission slots decrease with increase in bus length. It may also be observed that the throughput does not reach 100% due to the polling and other overhead given in section 4.8.1.

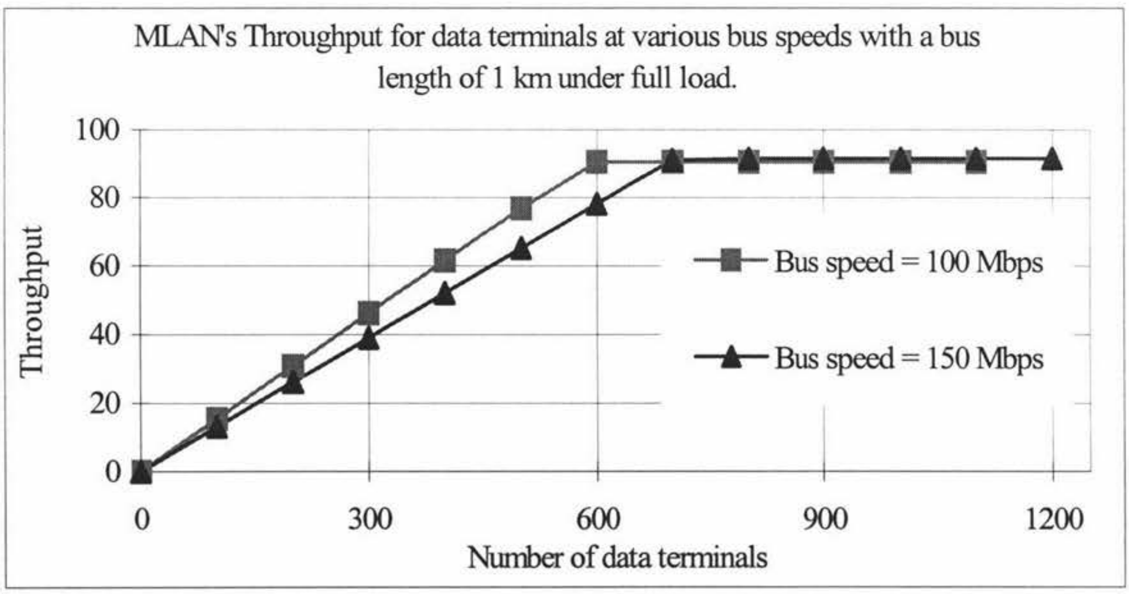


Figure 4.8.5.2 illustrating throughput at various bus speeds with a bus length of 1 Km.

However, from figure 4.8.5.2, it may also be observed that the throughput is independent of the bus speed, which is once again in agreement with the result published in [9].

4.8.6 Efficiency

At a bus length of 1 km, if the propagation delay is zero, the total available slots (including polling duration) are equal to 1415. However, due to propagation delay associated with bus length, the total available slots are equal to 649.

Hence, the bandwidth wasted due to propagation delay = (1415-649)/1415, which is equal to 54%.

At a bus length of 3 km, the bandwidth wasted due to propagation delay = (1415-312)/1415, which is equal to 79%.

Hence the efficiency of MLAN falls sharply due to propagation delay associated with bus length. The efficiency decreases with increase in bus length.

Tables 4.8.5a, and b show the characteristics of the data terminals at 35% terminal activity levels at various bus speeds.

Table 4.8.5a illustrates the performance of MLAN protocol for data traffic at a bus speed of 100Mbps with a terminal activity of 35%.

	L = 1km	L = 3km
Number of Terminals supported	1680	720
Average queuing delay (ms)	0	0
Mean cell delay (ms)	6	6
% of Throughput	90.8	80.45

Table 4.8.5b illustrates the performance of MLAN protocol for data traffic at a bus speed of 150Mbps with a terminal activity of 35%.

	L = 1 Km	L = 3 km
Number of Terminals supported	2015	785
Average queuing delay (ms)	0	0
Mean cell delay (ms)	6	6
% of Throughput	91.0	80.47

4.9 Simulation results of MLAN with Voice traffic generator

The performance of MLAN has been evaluated against various parameters such as access delay, and percentage of speech cell loss by varying number of terminals, and bus length at various bus speeds and the corresponding results would be discussed now.

4.9.1 Voice cell loss

The voice cell loss is defined as follows:

$$N_{CD} / (N_{CD} + N_{CST}) \qquad \ldots \quad 4.9.1.1$$

N_{CD} represent the number of cells dropped;
N_{CST} represent the number of cells successfully transmitted.

Figure 4.9.1 illustrates the percentage of voice cell loss of voice terminals at various bus lengths of 1, 2, and 3 km with a bus speed of 100Mbps. It may be observed from figure 4.9.1 that the percentage of voice cell loss is less than 1, which is the acceptable value for voice terminals [35] up to a number of terminals equal to 555 i.e., more than twice the number of transmission slots. To be more accurate it is nearly 2.2 times the number of transmission slots. This clearly indicates that each transmission slot supports 2.2 voice terminals with an acceptable cell loss of 1% [35]. With 0% cell loss, the each transmission slot supported about 2 voice terminals.

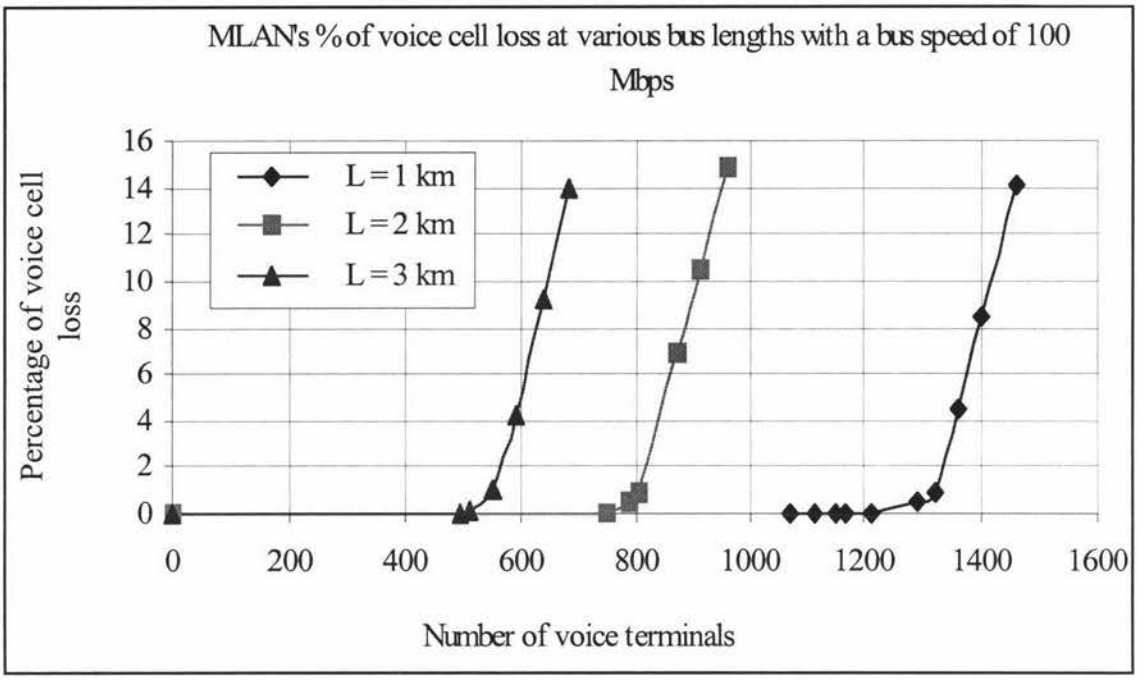


Figure 4.9.1 illustrates the % of voice cell loss at various bus lengths with a bus speed of 100Mbps.

Further, the reason for supporting 2.2 terminals in each transmission slot is the fact that the speech activity is about 45%, theoretically for this activity level, 2.22 terminals per transmission slot should be supported. The percentage of voice cell loss increases when the number of terminals requesting for a transmission slot are more than 555 as the terminals queue up for a transmission slot. While waiting in the queue the terminals which are unable to gain access to a transmission slot keep dropping cells once the cell dropping threshold is exceeded. The percentage of voice cell loss also increased with increase in bus length. Tables 4.9.1a, and b illustrate the number of voice/speech terminals that can be supported at bus speeds of 100 and 150Mbps respectively at various bus lengths with less than 1% speech cell loss.

Table 4.9.1a illustrates the performance of MLAN protocol for voice traffic at a bus speed of 100Mbps.

	L = 1 Km	L = 2 km	L = 3 Km
Number of Terminals supported	1320	805	555
Multiplexing ratio	2.24	2.23	2.21
% of speech cell loss	0.95	0.94	0.97

Table 4.9.1b illustrates the performance of MLAN protocol for voice traffic at a bus speed of 150Mbps.

	L = 1 Km	L = 2 km	L = 3 Km
Number of Terminals supported	1575	900	605
Multiplexing ratio	2.24	2.23	2.21
% of speech cell loss	0.92	0.95	0.96

From tables 4.9.1a and b, it may be noted that the MLAN supported more than 2.22 terminals per transmission slot at lower bus lengths, which is because of the fact that the speech activity varied between 44.4% to 46% with different random number values.

Table 4.9.1c illustrate the number of voice/speech terminals that can be supported at bus speed of 100Mbps at various bus lengths with 0% speech cell loss.

	L = 1km	L = 2km	L = 3km
Number of Terminals supported	1210	740	500
Multiplexing ratio	2.05	2.05	1.99
% of speech cell loss	0	0	0

Table 4.9.1d illustrates the number of voice/speech terminals that can be supported at bus speed of 150Mbps at various bus lengths with 0% speech cell loss.

	L = 1km	L = 2km	L = 3km
Number of Terminals supported	1440	825	545
Multiplexing ratio	2.05	2.05	1.99
% of speech cell loss	0	0	0

From tables 4.9.1c and d, it may be observed that with 0% voice cell loss, the MLAN supported nearly 2.05 terminals per transmission slot. However, at a bus length of 3km, it supported 1.99 terminals per transmission slot.

4.10 Performance of MLAN with Voice and Data Traffic

The MLAN protocol is then evaluated by combining the Voice and Data Traffic generators. Half of the total transmission slots are distributed equally between voice and data terminals. The remaining half, are allocated to both types of traffic with priority to voice traffic. In the case of data and voice terminals requesting for the resource, priority is given to the voice terminal as voice being real time source, while data transmission is deferred and the request is kept in a buffer. During the beginning of every frame, it is checked if both voice and data terminals are requesting for a resource that is commonly available. If a voice terminal is requesting for a resource, then voice terminal is given the resource and the voice terminal releases it only after its last cell of the current talkspurt is transmitted. In the absence of a voice terminal, a data terminal is given the resource, which would release it only after transmitting its last cell of the current burst. The disadvantage in this case is that if the average file size of the data terminal is large, the priority mechanism will not be of great use, which is reflected in the results shown in figure 4.10.

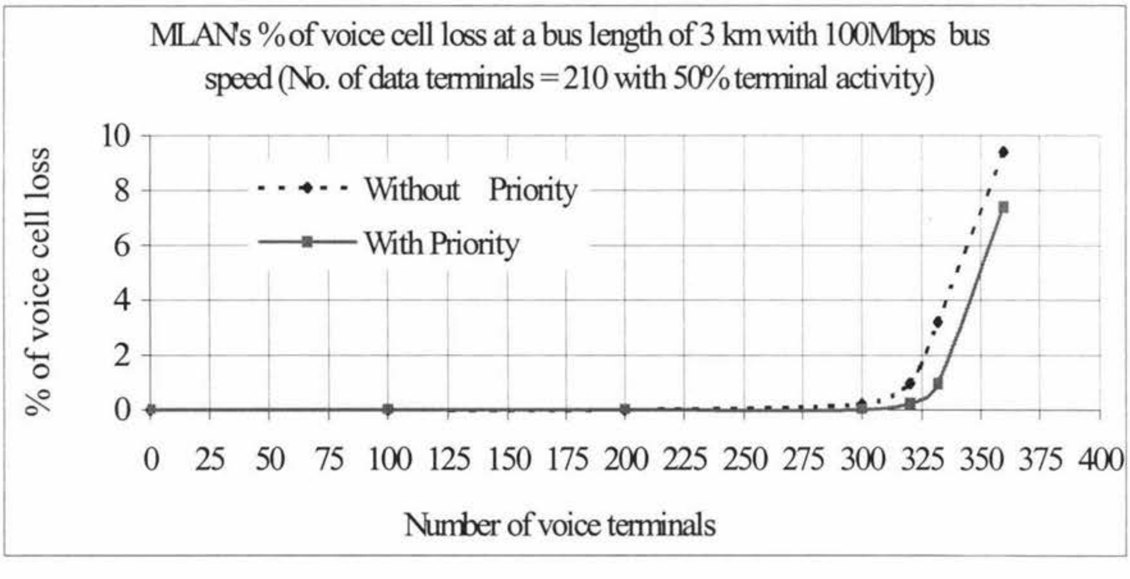


Figure 4.10 illustrates the % of voice cell loss of voice terminals with and without priority.

Figure 4.10 depicts the access delay for voice terminals (with and without priority) at a bus length of 3 km, with 210 data terminals. The terminal activity factor used in this case is 50%. It may be observed from the figure 4.10 that only a few more voice terminals (4% higher) can be supported using the priority mechanism.

Table 4.10 illustrates the characteristics of voice and data terminals at a bus length of 3 km with bus speed of 100Mbps with and without priority for voice terminals.

	With priority	Without priority
Total number of available transmission slots	251	251
Number of data terminals supported	210	210
Number of voice terminals supported	332	320
Multiplexing ratio	2.15	2.11
% of speech cell loss	0.98	0.97

4.11 Simulation of Fiber Distributed Data Interface (FDDI) and Fast Ethernet

As has been mentioned in the beginning of this chapter, to compare the results of MLAN with the other shared media high speed LANs, Fiber Distributed Data Interface (FDDI) and Fast Ethernet have been simulated using COMNET III, a network simulation package. FDDI and Fast Ethernet have been chosen for further study, as both of them are high-speed shared media LANs. FDDI supports both synchronous traffic, which consists of delay sensitive traffic such as voice, and asynchronous traffic that consists of delay insensitive traffic such as File Transfer Protocol and mail. It has been mentioned previously in chapter 2, that FDDI in its synchronous mode has low access latency and low jitter. FDDI also guarantees a bounded access delay and a predictable average bandwidth for synchronous traffic. Fast Ethernet has been chosen for further study, as Fast Ethernet is a natural upgrade path for existing Ethernet users. An overview of the FDDI protocol presented below.

4.12 Fiber Distributed Data Interface (FDDI)

The FDDI [10, 11, 12, 13, 73] is a 100Mbps local/metropolitan area network. It employs optical fiber as the transmission medium and is based on a token access method. Based on two counter rotating rings, an FDDI network provides an interconnection for information exchanges among up to 500 stations over distances of up to 100 km. Its transmission capacity of 100 Mb/s enables high performance interconnections in both the front-end and the back-end computer environment. If installed as a backbone it may interconnect lower speed sub-networks or concentrators.

The basic operation of the FDDI MAC protocol is similar to that of a token ring. Access to the network is controlled by passing a permission token around the ring. In order to transmit a station needs to capture a token. The protocol adopted for the network access mechanism is the Timed Token Rotation (TTR) protocol. Under this protocol, each station measures the time that has elapsed since a token was last received. The initialization procedures establish the Target Token Rotation Time (TTRT) equal to the lowest value that is bid by any of the stations. Two classes of service are defined. Synchronous service (guaranteed bandwidth and response time for applications with predictable bandwidth and response time requirements) allows use of a token whenever MAC has synchronous frames queued for transmission. Asynchronous service (dynamic bandwidth sharing for applications with bursty or potentially unlimited bandwidth requirements) allows use of a token only when the time since a token last was received has not exceeded the established TTRT.

Each station has an allocated synchronous bandwidth, and asynchronous service instantaneously allocates bandwidth that is unallocated or unused. A station gains the right to transmit when it detects the passing token. First it transmits the frames of the highest priority synchronous access class. The remaining transmission time can be used sending asynchronous frames. The amount of time a station is allowed to transmit asynchronous frames depends on the time of the successive token arrivals at this station in order to satisfy the maximum token rotation time. Since the protocol allows multiple frame transmissions per token arrival and a station has to pass on the token immediately after the end of frame transmission, it provides efficient use of the high transmission capacity.

4.13 The architecture of FDDI and Operation of its Timers

The FDDI's architecture is shown in figure 4.13a. FDDI's physical layer is divided into two sub-layers such as Physical Media Dependent (PMD) sub-layer and Physical Media Independent (PHY) sub-layer.

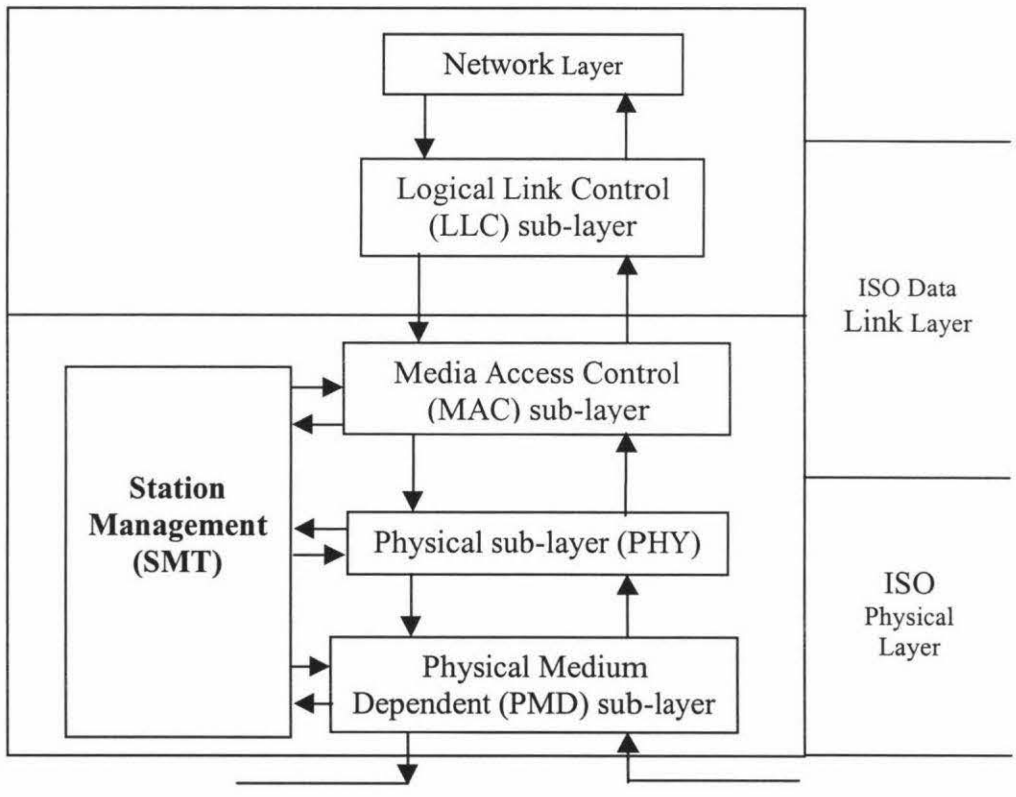


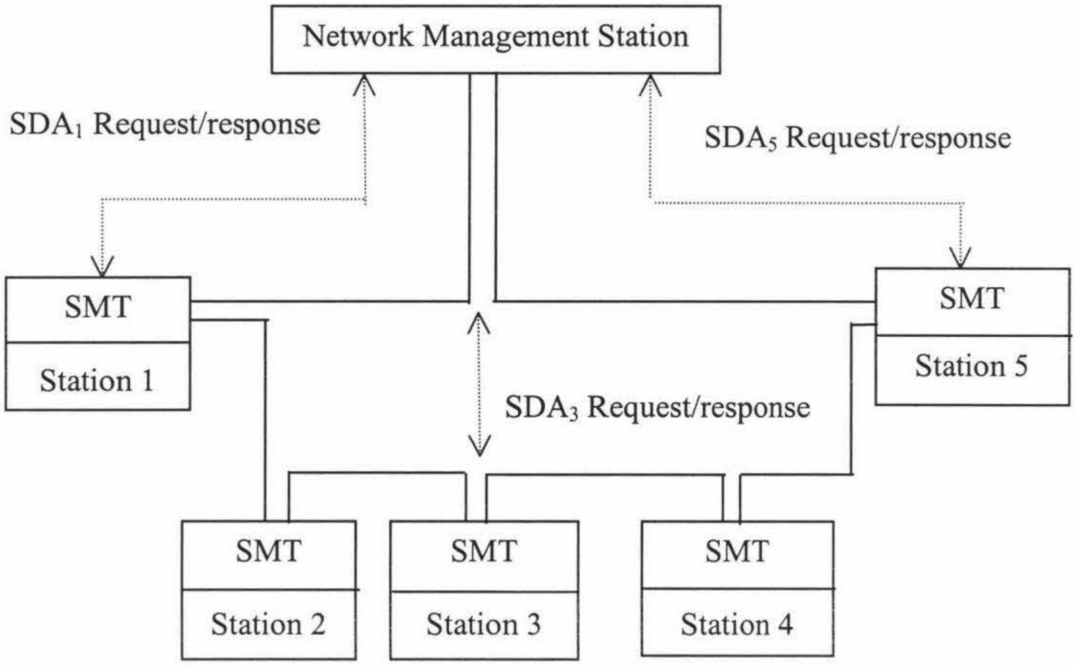
Figure 4.13a FDDI protocol architecture.

The basic function of the PMD is to convert the signal provided by the PHY layer to a form that is suitable for the underlying media. For example, when optical fibre is used as the medium of transmission, the PMD converts the electrical signal provided by the PHY into optical signal for transmission and the optical signal from the media into electrical signal for the PHY layer.

The sub-layer between PMD and MAC is given by the Physical Layer Protocol (PHY), which specifies the coding scheme for data transmission. Based on the PMD, the (PHY) specifies the encoding, decoding, clocking, and data framing. The Medium Access Control (MAC) protocol is based on the principle of token passing and defines the token handling and as well as frame transmission and reception. Finally, a Station Management (SMT) provides a link-level management for FDDI.

The MAC protocol is based on a control token and in addition to normal data traffic, optionally the ring can also support delay sensitive traffic that requires a guaranteed maximum access delay, for example, digitized speech. As part of the ring initialization process, all stations negotiate an "Operative Target Token Rotation Time (T-Opr). The initialization procedures establish the Target Token Rotation Time (TTRT) equal to the lowest value that is bid by any of the stations. Since the protocol guarantees an average token rotation time not larger than T-Opr and a maximum token rotation time not larger than twice T-Opr, each station must request a Target Token Rotation Time (TTRT) which is half as large as the response time required [12]. Following the operation of token passing, a station gains access to the medium by removing the token from the ring before it begins transmitting queued frames.

To cater for synchronous data, those stations that support such traffic are allocated a fixed portion of the ring bandwidth that may be used each time the token is received. This is known as the synchronous allocation time (SAT) and defines the maximum length of time for which a station can transmit synchronous data each time it receives the token. Synchronous data is not controlled by the timed token rotation protocol; instead allocation of ring bandwidth to individual stations for synchronous data is controlled by separate ring management station. The general scheme is shown in figure 4.13b.



SMT: Station Management

Stations 1, 3, 5 support asynchronous and synchronous data.

Stations 2 and 4 support asynchronous data only.

SDA_n: Synchronous data allocation time for station "n".

$\sum_{i=1}^{n} SDA_i$: Ring Target Token Rotation Time, TTRT

Figure 4.13b FDDI synchronous data negotiation

All requests for synchronous bandwidth are expressed as a proportion of the ring TTRT. These requests are sent to the network management station and provided there is free synchronous bandwidth available, the stations are each allocated their requested amount. It may be noted that neither all stations need to provide synchronous data service, nor all portions of synchronous bandwidth need to be the same. To implement the scheme, each station must be able to communicate with the network management station. This is, achieved by each station having its own ring management – known as station management (SMT) – agent that communicates with the ring management station, via the ring.

As mentioned earlier, the MAC protocol supports synchronous and asynchronous services. It is very important to note that the total bandwidth available to users depends upon the value of T-Opr. A lower T-Opr value causes more token rotations resulting in a lower throughput [12].

Asynchronous traffic is controlled by a Token Rotation Timer (TRT), which is maintained at each station. The timer is set to T-Opr each time a token arrives before TRT reaches T-Opr. In such a situation the token is referred to as early token. If the TRT expires before it sees the token, the token is considered late, which is indicated by setting a late counter. If the TRT expires a second time, the station recognizes that a fault has occurred.

The Token Holding Time (THT) is used to control the amount of time that a station can transmit asynchronous class frames. It is loaded with the value remaining in the TRT timer every time the token arrives at the station. If the token rotates faster around the ring (ring is lightly loaded; not many stations are transmitting) there will be more time remaining in the TRT when the token returns to the station. Therefore, a larger time value will be loaded in to the THT timer allowing this station to transmit more. A station transmits asynchronous frames until the THT timer expires. This timer is only enabled once a station sees a token, enough time remaining in the TRT (TRT not expired), and the token is captured. This timer is for asynchronous transmissions and is enabled at the end of synchronous transmissions, as synchronous transmissions are higher priority. If the timer expires in the middle of a transmission, the current transmission is completed before releasing the token.

The FDDI network model is simulated using the COMNET III, a network simulation package. A brief description of COMNET III is given below.

4.14 COMNET III

COMNET III provides building blocks, which can be used to define a communication network. To obtain appropriate simulation results a network model must be defined properly. The COMNET III simulation package, based on discrete event simulation, is a performance analysis tool for computer and communication networks in different technologies such as packet switching, message switching or circuit switching and LAN and WAN internetworks. Based on the network description, its protocols, algorithms and traffic load, it simulates the operation of the network and provides results. Network models are created using graphical icons and they are defined appropriately. All functions of the network design, model execution and presentation of results will be done through a single window in this simulation package. The package can simulate almost any type of fixed network [74].

4.15 FDDI Network Simulation Model

The FDDI network's COMNET-III simulation model is shown in figure 4.15.1. As has been mention previously, the model was simulated using the COMNET III, a network performance analysis package. Negative exponential distribution was used to generate data traffic. Traffic volume in the simulation model was varied. The simulation was carried for duration of 100 seconds and the performance was evaluated for throughput and packet delay, which includes the response time. The mean inter-arrival time is taken as 15ms. Traffic plot is shown in figure 4.15.2. It may be observed by comparing the figures 4.6.1 and 4.15.2 that this traffic pattern is almost similar to that of the one used for MLAN simulation.

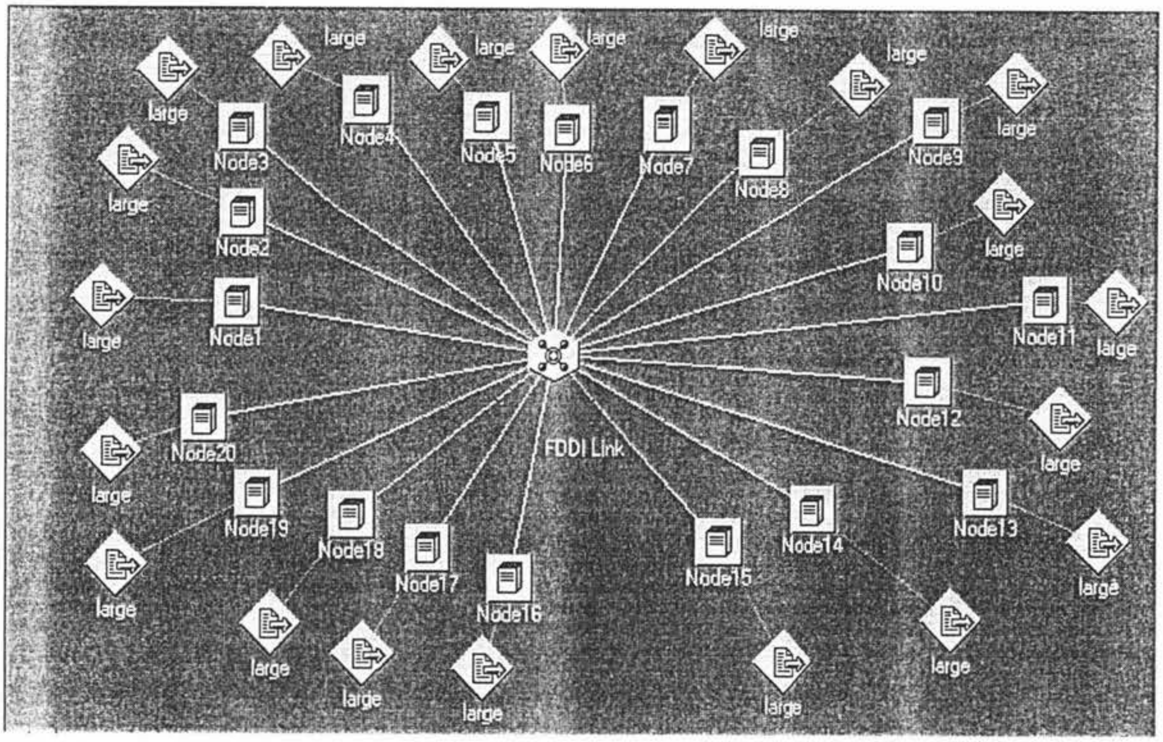


Figure 4.15.1 Network simulation model of FDDI

However, in MLAN with 100% load, the number of terminals considered are 251 at 100Mbps, while in this case the number of terminals considered are only 20. This is because the academic license agreement with COMNET III vendors does not allow to simulate more than 20 nodes at any time, which is incorporated in the COMNET III package. Hence, the overall load has been adjusted to be nearly the same as that generated in MLAN by generating the required file sizes at each node in COMNET III. The network model used for the simulation of FDDI with COMNET III is shown in figure 4.15.1. It may be observed from the figure 4.15.1, each traffic source is connected to each node and all the nodes are in turn connected to the FDDI ring. The FDDI network is simulated by varying the traffic load at the terminals. The FDDI network is simulated only for data traffic. Figure 4.15.3 illustrates the throughput and average packet delay of FDDI against traffic load. The simulation parameters are shown in table 4.15.

Table 4.15 shows the simulation parameters of FDDI.

Simulation time	100 seconds
Warm-up time	30 seconds
Transmission Bandwidth	100Mbps
Token Passing delay	0.00075 ms
Target Token Rotation Time	1ms and 1.5ms.
Propagation delay	0.015 ms
Session limit	1024
Frame minimum	32 bytes
Frame maximum	4500 bytes
Frame overhead	32 bytes

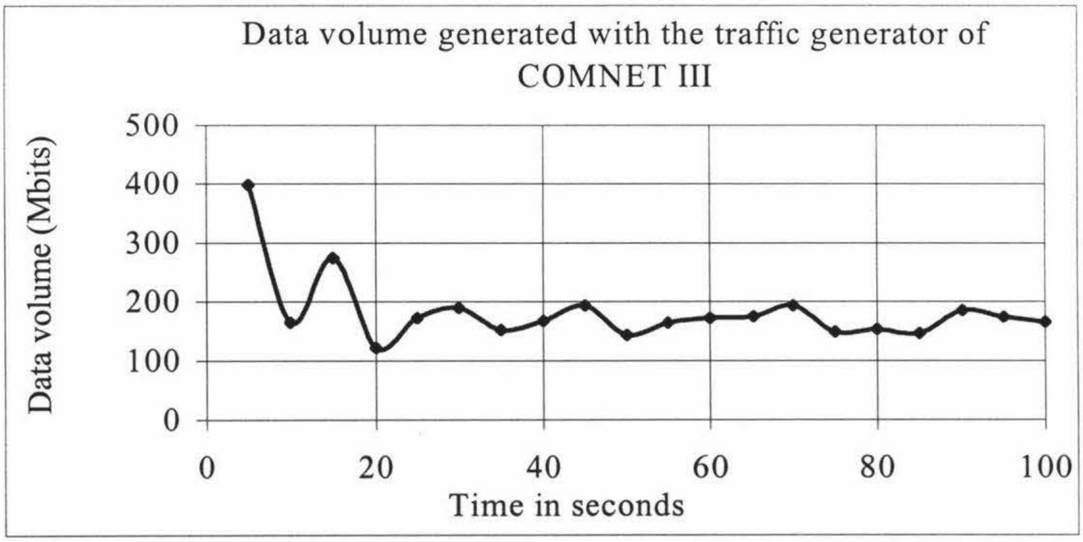


Figure 4.15.2. Traffic plot of message generator.

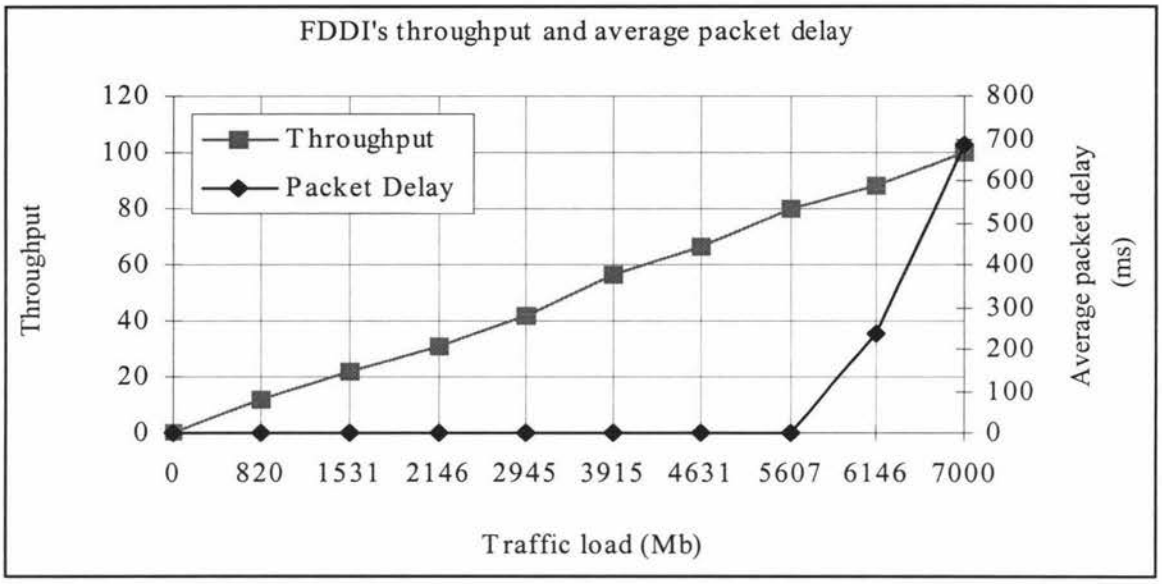


Figure 4.15.3a shows the variation of Throughput and average Packet delay of FDDI against traffic load with a TTRT of 1ms.

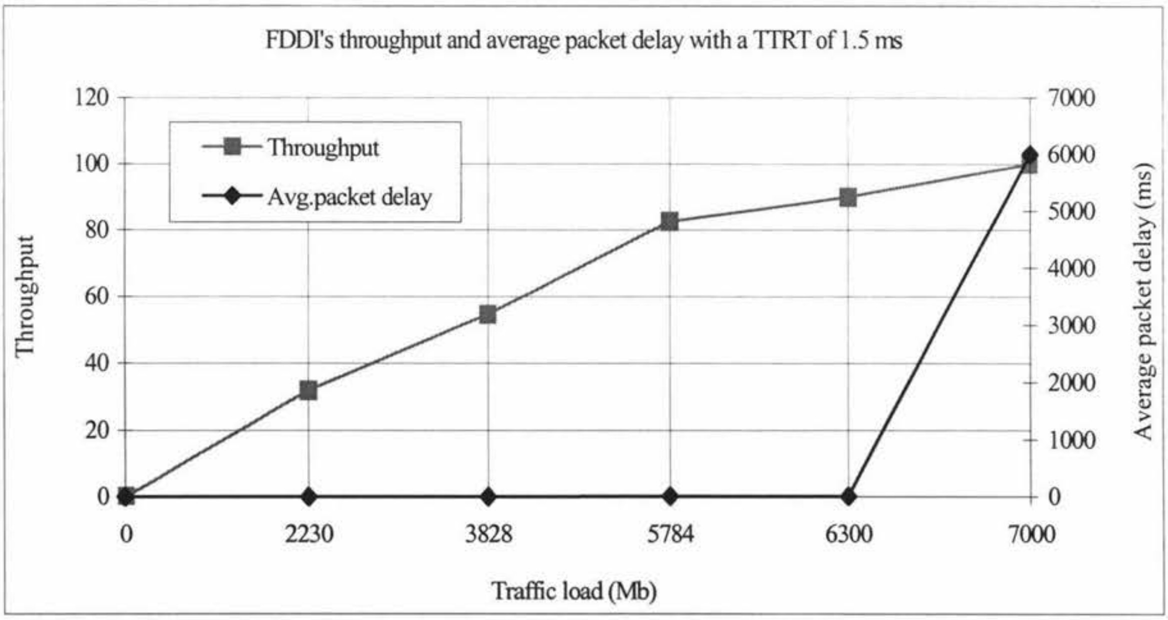


Figure 4.15.3b shows the variation of Throughput and average Packet delay of FDDI against traffic load with a TTRT of 1.5 ms

The throughput of FDDI is calculated as the ratio of the average bandwidth used to the total bandwidth available as a percentage. It may be observed from figure 4.15.3a, the throughput of FDDI increases linearly with increasing traffic load and at a throughput of just about 80%, the average packet delay is about 0.9 ms. When the traffic load is increased further, although the throughput increases the packet delay also increases. This is because of a lower value of TTRT (T-Opr) which is chosen as 1ms to meet the stringent delay requirements of multimedia traffic the throughput is less at lower delays as lower TTRT value causes more number of token rotations thereby resulting in a lower throughput. The result obtained from the simulation model coincides with that published in the literature [12] for a TTRT value of 1 ms, which also validates the simulation model. With a TTRT of 1.5 ms, the throughput of FDDI is bout 83% with 3.8 ms packet delay. However at a throughput of about 90%, the packet delay is about 9 ms. At a throughput of 100%, the packet delay is 6 sec.

The throughput of MLAN is about 80.45% at a bus length of 3 km. The throughput of MLAN is however, is independent of traffic load, traffic composition, and bus speed. However, the throughput of MLAN depends upon the bus length and hence would be much lower than 80.45% at longer bus lengths. Bandwidth is wasted due to the propagation delay. While, MLAN is targeted for small campus environment of a few km, FDDI can be used for LAN as well as MAN environment.

4.16 Fast Ethernet (100 Base-T) and its Simulation Model

The initial motivation for developing "Fast Ethernet" [7, 14, 75] was the realization that it would be easy to design and much less expensive to build than FDDI. Further, if the new low-cost LAN were compatible with 10 Mb/s 10BASE-T (Ethernet) technology, the new LAN would be easy to integrate into existing LANs, and therefore, should share the success of 10BASE-T.

Fast Ethernet also referred to as 100BASE-T uses the same Collision Sense Multiple Access with Collision Detection (CSMA/CD) Media Access Control Protocol (MAC) that is at the core of 10 Mb/s Ethernet. All MAC timing parameters are sped up a factor of 10 with the remainder of the MAC algorithm remains unchanged [14]. The protocol sub-layers of 100Base-T and Ethernet are shown in figure 4.16.1 and important sublayers are briefly are described below.

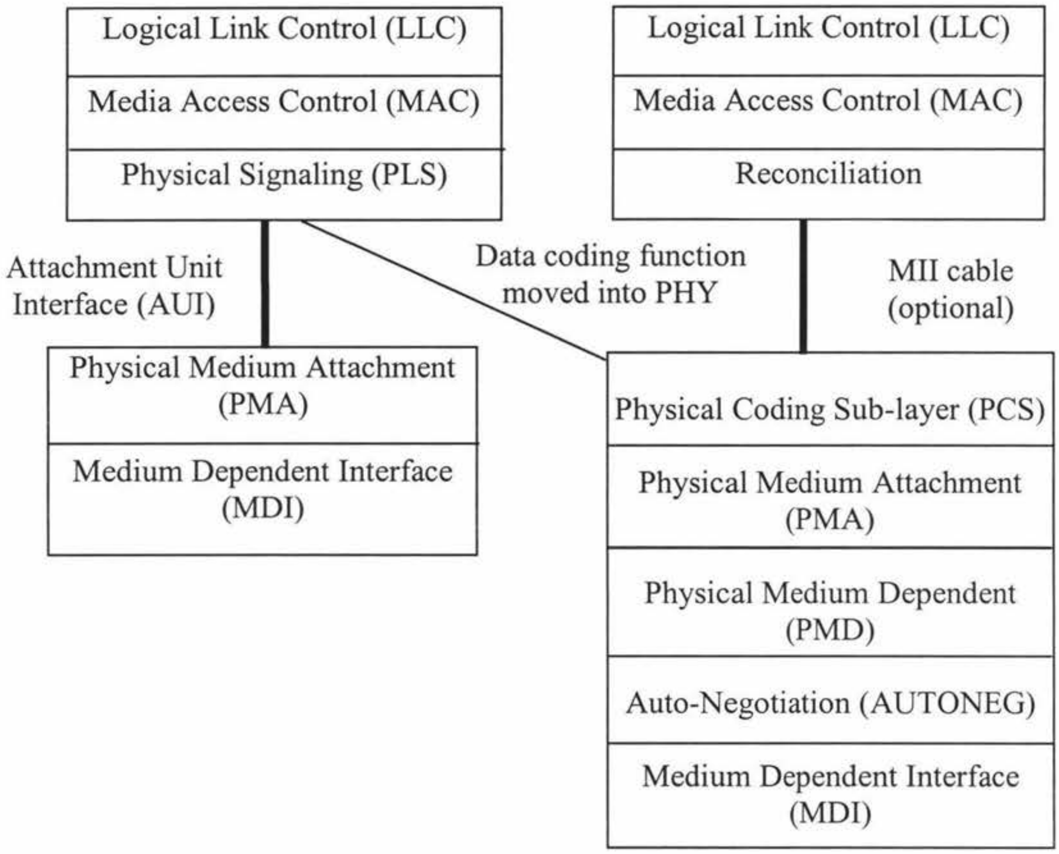


Figure 4.16.1 Fast Ethernet and 10 Mb/s Ethernet Protocol sub-layers.

The **Media Independent Interface (MII)** has been defined between the Reconciliation Sub-layer and the Physical Coding Sub-layer (PCS). The MII is essentially an interconnection between a network adapter and an external Fast Ethernet transceiver. The MII isolates the upper protocol layers (MAC, packet buffers, and adapter I/O interface) from the transceiver functions (transmit amplifier, wave shaping, and receive discriminator). The part below the MII, the part that connects to the physical cabling, is usually called a transceiver. The official Fast Ethernet term for transceiver is Physical Layer Device (PHY). The PHY consists of the analog circuitry necessary to communicate with the physical transmission medium. The PHY is a combination of the Physical coding sub-layer, Physical Media Attachment sub-layer (PMA), Physical Media Dependent sub-layer and Auto-negotiation (AUTONEG) sub-layers.

The **Physical Coding Sub-layer (PCS)** is responsible for coding transmitted data into a form suitable for the physical medium and decoding it at the receiver.

The **Reconciliation sub-layer** also referred to as "weenie" sub-layer translates the terminology used in the MAC into the terminology appropriate for the MII. To the MAC, it is a transparent, functionless sub-layer. Signal just passes through it. There are no options, configuration choices, or user-accessible features in this sub-layer.

The Auto-Negotiation sub-layer provides extensive support for determination of link options and optimal settings. With Auto-Negotiation enabled, an adapter card may determine for itself what the capabilities are at the far end of the link and select the best operational mode as needed.

The simulation model of Fast Ethernet is shown in figure 4.16.2 and the simulation parameters are shown in table 4.16.

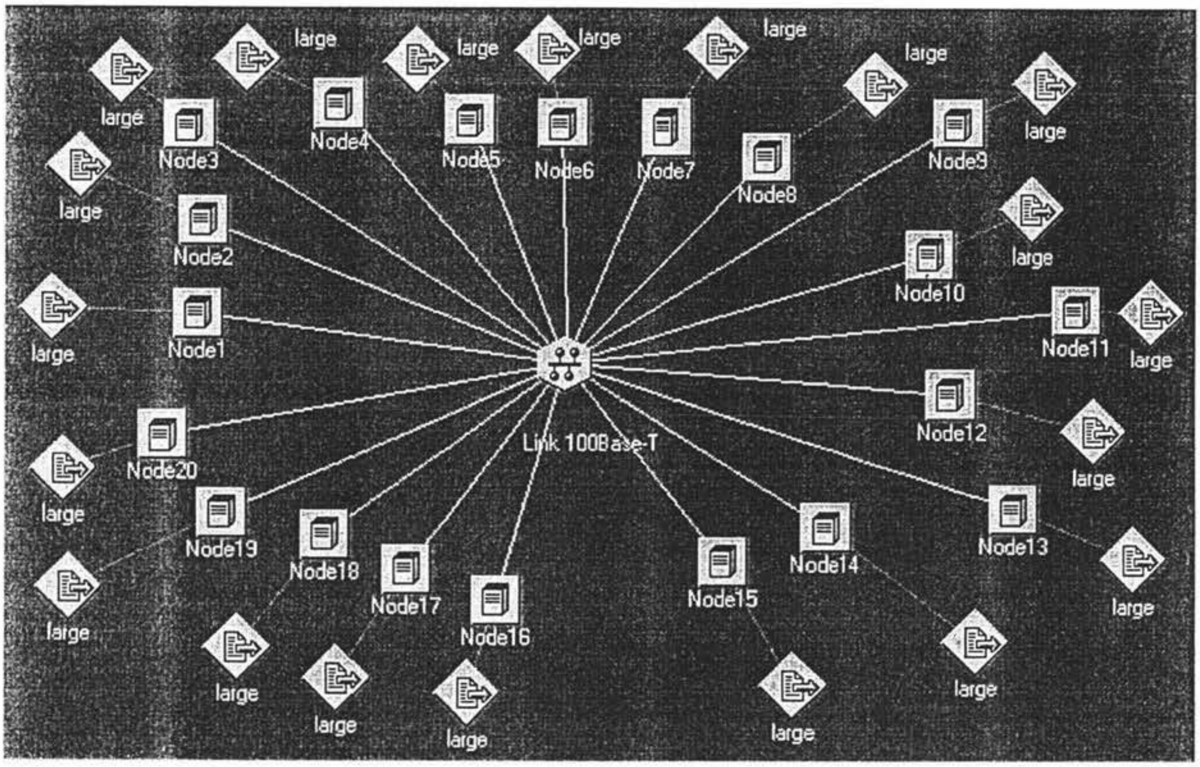


Figure 4.16.2 Network simulation model of Fast Ethernet

Table 4.16 illustrates the simulation parameters for the simulation model of Fast Ethernet shown in figure 4.16.1

Simulation time	100 seconds
Warm-up time	30 seconds
Transmission Bandwidth	100Mbps
Collision Window	0.0075 ms
Jam interval	0.00032 ms
Interframe gap	0.00096
Propagation delay	0.00085 ms
Session Limit	1024
Frame minimum	32 bytes
Frame maximum	1526 bytes
Frame overhead	30 bytes
Slot time	0.00512 ms

Fast Ethernet is also simulated using COMNET III and its throughput and average packet delay, which includes response time are evaluated and are shown in figure 4.16.3a. The traffic generator used is the same as the one used for simulating FDDI, except that the file size generated by each source is kept to a smaller value.

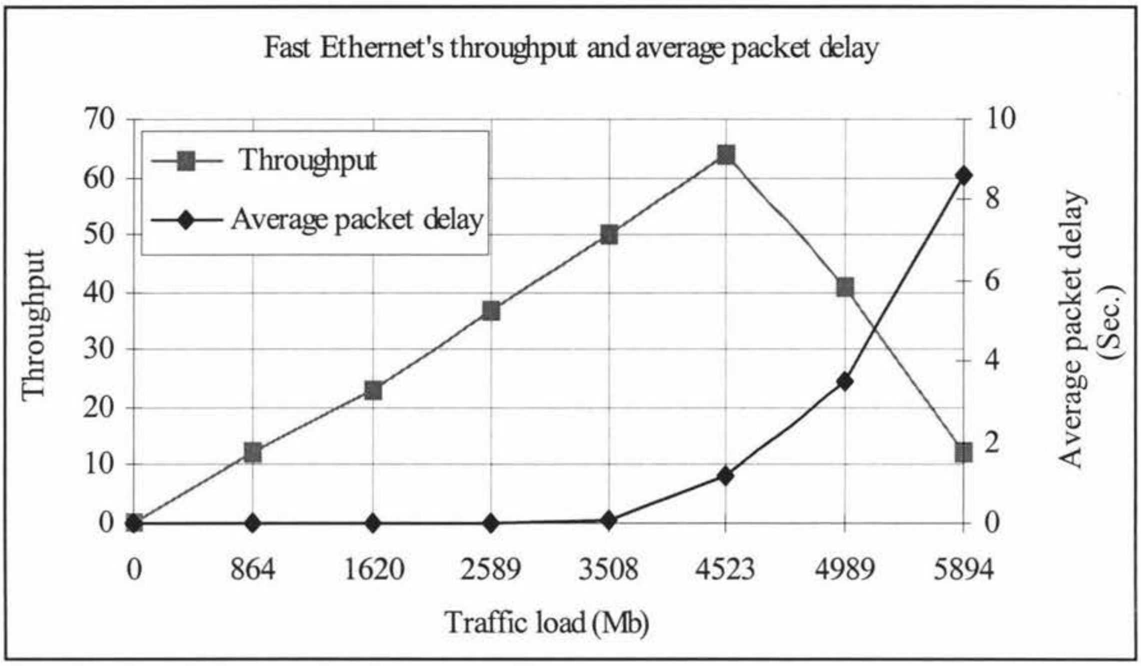


Figure 4.16.3a illustrates the variation of Throughput and Average packet delay of Fast Ethernet against traffic load.

It may be observed from the figure 4.16.3a the throughput of Fast Ethernet is about 64% at a traffic load of about 4550 Mb and it then has fallen sharply to 12% at a load of 4950 Mb. Subsequently, the average packet delay also increases. This is because of the fact that the protocol is essentially the same as CSMA/CD and therefore, the throughput is only slightly better than the CSMA/CD. The collision statistics are shown in figure 4.16.3b to confirm that the throughput indeed falls sharply to about 12%.

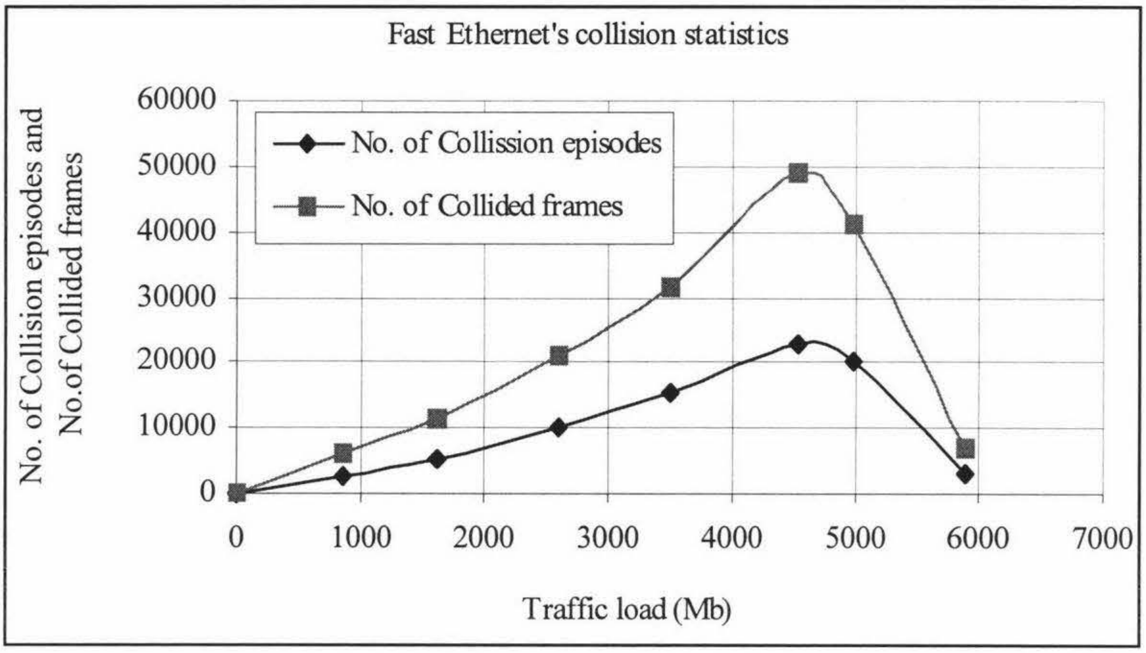


Figure 4.16.3b illustrates the collision statistics of Fast Ethernet against traffic load.

4.17 A comparison of MLAN, FDDI, and Fast Ethernet

A comparison of MLAN, FDDI and Fast Ethernet is illustrated in table 4.17.

Table 4.17 shows a comparison of FDDI, 100 Base-T, and MLAN

MLAN	FDDI	100 Base-T
Can be used as a LAN within a campus.	Mainly used as a backbone network connecting various LANs serving up to 100 km.	Can be used as a LAN within a campus.
Throughput is independent of traffic load, traffic composition and bus speed. However, throughput depends upon the bus length and is therefore, useful in a small campus of a few km. Throughput is 80.5% at a bus length of 3 km. However, throughput decreases with increase in bus length as propagation delay consumes a fair amount of bandwidth at longer bus lengths.	Throughput depends upon the traffic load and the value of TTRT. Configuring a lower value of TTRT, which is required for multimedia traffic, results in low utilization of ring. Throughput is about 80% for a TTRT of 1 ms [12]. However, the throughput of FDDI is 100% at higher TTRT values. Even at TTRT of 1 ms, throughput is higher, but then the packet delay is also higher.	Throughput depends upon the traffic load and is of not much use at heavy traffic loads. Throughput is just about 64%.
Packet delay depends upon the bus length and the number of terminals. At 100Mbps, 3 km bus length with 251 terminals, packet delay is equal to the propagation delay which is 0.015 ms.	Packet delay depends upon TTRT value. At a TTRT value of 1 ms, packet delay is 0.1 ms.	Packet delay depends upon the traffic load and traffic type (composition).
At a bus length of 3km, efficiency of MLAN decreases by about 80%, due to the propagation delay associated with the bus length. This is the inherent disadvantage of broadcast bus LANs.	Efficiency of FDDI is 100%, i.e. there is no effect of propagation delay on the throughput.	Efficiency of Fast Ethernet also decreases due to propagation delay by about 55% to 75 %. This is the inherent disadvantage of broadcast bus LANs.
Can connect up to 250 terminals at 100% terminal activity at a bus length of 3km.	Can connect up to 500 terminals up to 100km.	Support for number of terminals varies with segment length and load.

From the table it may be deduced that FDDI with a throughput of about 80% at a TTRT of 1ms is more useful as a backbone network connecting various LANs or can be used as a metropolitan area network. However, the efficiency of FDDI is 100% and the effect of propagation delay on the bus length is negligible. Although FDDI is used for connecting high performance workstations, due to higher cost, is used primarily for backbone networks. However, of late, with the falling of hardware costs, high performance workstations are connected as FDDI LAN. The throughput of 100Base-T is about 64%, which is much lower than that of MLAN and FDDI. However, Fast Ethernet is often seen as a migration path for existing Ethernet users.

On the other hand, MLAN is quite useful in a small campus environment, as its throughput is independent of traffic composition (the throughput was found to be the same with voice traffic or data traffic or any combination of voice and data traffic), and operates in native ATM mode, thereby eliminating the need for ATM adaptation layer. The throughput of MLAN is about 80.5% at a bus length of 3 km and this throughput is independent of traffic composition. It's throughput and packet delay are independent of bus speed. The disadvantage being that MLAN can not be used above a bus length of 3 km as its throughput falls sharply with increase in bus length. Efficiency of MLAN protocol is very low of about 21% for a 3 km bus length and 46% for a 1km bus length. Low efficiency is due to the wasted bandwidth to support propagation delay, which is the main disadvantage of broadcast bus LANs.

4.18 Conclusions

The simulation results of MLAN using source models of voice and data clearly indicate that the performance of MLAN is independent of traffic load, traffic composition, and bus speed. At a bus length of 3 km, the throughput is about 80.5%. At a terminal activity level of 50%, support for number of terminals can be increased by about 100% more than the one at a terminal activity of 50%, without any degradation in the performance such as response time (access delay/queuing delay) that is crucial for multimedia applications. Further, the mean cell delay is independent of bus speed. The main disadvantage in MLAN is the waste of bandwidth of due to propagation delay. Hence, MLAN is not meant for MAN environment, but is only meant for a small LAN environment. However, research needs to be carried to find out mechanisms to avoid wastage of bandwidth.

Simulation results obtained from the simulation model of FDDI show that the throughput of FDDI is about 80% at a TTRT of 1ms. This throughput is nearly the same as that of MLAN. Besides, as has been mentioned in chapter 2, FDDI can not support isochronous traffic, which is highly desirable in interactive multimedia applications. To support isochronous traffic the basic FDDI is extended to FDDI-II. In addition, FDDI uses two interfaces per terminal, and therefore, is expensive for a small campus.

The throughput of Fast Ethernet is just about 64% and therefore, may not be of great use when compared with MLAN and FDDI. However, as has been mentioned previously, Fast Ethernet is seen more as a migration to Ethernet (10Base-T) and therefore can use the same wiring structure of 10Base-T. This is the main advantage of Fast Ethernet.

The usefulness of MLAN lies in the fact that it operates in native ATM mode, thereby eliminating the need for ATM adaptation layer when it is connected to a public ATM network.

Chapter 5

Conclusions and Future Work

5.0 Conclusions

In this study, the performance of the Multimedia Local ATM Network (MLAN) [9] has been evaluated by using voice and data traffic source models. The performance was evaluated by computer simulation. In addition two other high speed LANs such as FDDI and Fast Ethernet were simulated and their performance was evaluated using COMNET-III, a communication network simulation package. The performance of MLAN is compared with that of FDDI and Fast Ethernet.

The Throughput of MLAN is evaluated at different terminal activity levels, with a bus speed of 100Mbps and 150Mbps at bus lengths of 1km and 3 km respectively. At 100% activity level, the throughput has dropped from 91% to 80.5% when the bus length is increased from 1 km to 3 km. Throughput is found to be independent of bus speed and the traffic composition, as the throughput is the same with data traffic or voice traffic or any combination of data and voice traffic. However, the efficiency of MLAN has fallen by about 79% due to increase in propagation delay associated with bus length, due to its slot structure.

Simulation results using data traffic generator have shown that at a terminal activity level of 50%, the number of terminals on the bus can be doubled compared to 100% terminal activity level without any increase in the average cell delay. MLAN supports about 2.0 terminals per transmission slot at 50% terminal activity. It was further found to be validated, when the MLAN was simulated using voice traffic generator (the speech activity is about 45%) which has shown that the MLAN supports about 2.2 voice terminals per transmission slot with 1% speech packet loss. The MLAN supports about 2.05 voice terminals per transmission slot with 0% speech packet loss.

While the mean cell delay of MLAN is found to be almost independent of the bus speed, it increases with increase in terminal activity level. The mean cell delay also increases with increase in bus length.

When voice traffic is given priority over data traffic, the multiplexing factor was better and the total number of terminals that were supported has increased by about 4% over that without priority. However, this priority mechanism is not of much use, particularly when the burst size produced by data traffic generator is very large. Although not implemented in this study, the priority mechanism may be implemented so that data traffic is given resource frame by frame, in spite of this being expensive as it requires more buffer space.

The efficiency of MLAN is found to be about 21% at a bus length of 3km and 46% at a bus length of 1km respectively. The decrease in efficiency is due to the wasted bandwidth to support propagation delay associated with bus length.

The throughput of FDDI is found to be about 80% with a TTRT value of 1 ms with a mean packet delay of about 0.1 ms. The throughput is about 100% at higher packet delay of about 700ms, which may not be desirable for multimedia traffic. The throughput of FDDI is, therefore, dependent on the value of TTRT. The value of TTRT should be greater than a certain threshold value because very low value of TTRT would result in the token rotating more number of times around the ring, thereby resulting in lower utilization of the ring. Although not implemented in this study, FDDI, with its synchronous operation mode and bandwidth guarantee, voice traffic always gets priority. Further, there is no effect of propagation delay on FDDI and its efficiency is very close to 100% as the overhead consumes about less than 1% of bandwidth.

The throughput of 100Base-T is found to be about 64% with a mean packet delay of 9 sec. At heavy traffic loads the throughput has fallen sharply to about 12% as the protocol is based on CSMA/CD [14]. The efficiency of Fast Ethernet decreases by about 55% to 75% due to propagation delay associated with bus length.

The throughput of MLAN is nearly the same as that of FDDI and is quite higher than that of 100Base-T whose throughput was found to be just about 64%. However, FDDI does not support truly isochronous traffic, which is highly desirable in interactive multimedia applications [76]. Isochronous traffic allows fixed number of packets of data to be delivered in fixed time interval for which purpose the basic FDDI has been extended to FDDI-II in which bandwidth allocation is done in terms of 64kbps, which again for bursty sources is a waste of bandwidth. While, FDDI requires two interfaces per terminal, the MLAN uses only one interface per terminal [9]. Further, the MLAN protocol operates in native ATM mode whereas FDDI needs to be interfaced to ATM broadband network using ATM adaptation layers. Although, FDDI has low access latency and low jitter in its synchronous mode, due to higher cost, FDDI networks are used primarily for backbone networks. However, in many cases, FDDI is used as a LAN to connect high performance workstations.

The performance of MLAN is dependent only on the number of terminals and the bus length. Hence, the MLAN is suitable only for campus environment and not for MAN environment [9]. This is because quite a fair amount of bandwidth is wasted to support propagation delay associated with bus length. If a larger area is desired, more number of MLANs may be interconnected through an ATM switch. Besides, once the ATM switches are employed to interconnect the various LANs to the public network, MLAN may be more preferable compared to other shared media high speed LANs. This is because MLAN has the advantage of operating in native ATM mode. Hence, MLAN does not require Internetworking Functions (IWF) at the LAN to the public network interface thereby eliminating the need for ATM adaptation layer and here lies the extreme usefulness of MLAN.

5.1 Scope for Future Work

In this study, only the Voice traffic generator and the Data traffic generator were considered to evaluate the performance of MLAN. The performance of MLAN needs to be evaluated by using the Constant Bit Rate (CBR) and the Variable Bit Rate (VBR) video source models and to see the ability of MLAN to handle the multi-bit rate traffic. Some priority mechanism needs to be implemented which can give highest priority for voice over CBR and VBR video in addition to data. The efficiency of MLAN protocol needs to be improved so that wastage of bandwidth to support propagation delay is kept to a minimum. This may probably be done by using a smaller bus length of 0.5km and interconnecting more number of MLANs through an ATM switch. In addition, interworking with public ATM network needs to be done as the global Telecommunication industry is converging on ATM.

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