

Performance Analysis of LAN, WAN & WLAN in Eritrea

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DECLARATION

I, Osman Mohammed Osman Kakay, Student Number 204000016, hereby declare that the thesis entitled Performance analysis of LAN, WAN & WLAN in Eritrea is a result of my own investigation and research, and presents my own work unless specifically referenced in the text. This work has not been submitted in part or in full for any other degree or to any other university.

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ABSTRACT

The dissertation addresses the communication issues of interconnecting the different government sectors LANs, and access to the global Internet. Network capacities are being purposely over-engineered in today's commercial Internet. Any network provider, be it a commercial Internet Service Provider (ISP) or Information Technology Service department at government sector, company or university site, will design network bandwidth resources in such a way that there will be virtually no data loss, even during the worst possible network utilization scenario. Thus, the service delivered by today's end-to-end wide area Internet would be perfect if it wasn't for the inter-domain connections, such as Internet access link to the ISP, or peering points between ISPs. The thesis studies the performance of the network in Eritrea, displaying the problems of Local Area Networks (LANs) and Wide Area Networks (WAN) and suggesting initial solutions and investigating the performance of (WAN) through the measured traffic analysis between Asmara LAN and Massawa LAN, using queuing theory system (M/M/1 and M/M/2) solution. The dissertation also uses OPNET IT Guru simulation software program to study the performance of LAN and WLAN in Eritrea. The items studied include traffic, collision, packet loss, and queue delay. Finally in order to follow the current trends, we study the performance of VOIP links in Eritrean WANs environment, with a focus on five different link capacities: 28 kbps, 33 kbps, 64 kbps, and 128 kbps for voice and 256/512 kbps for voice and data. Using the R value as a measure of mean opinion score (MOS), we determine that the 33 kbps link would be adequate for Eritrean WANs.

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LIST OF ACRONYMS

bit	smallest unit of digital information, a one or a zero.
byte	a set of bits that represent a single character. Usually there are 8 bits in a byte
bps	bits per second
Kbps	kilobits per second =1000 bits per second
Mbps	Megabit bits per second =1,000,000 bits per second
Gbps	Gigabits per second = 1,000,000,000 (one billion) bits per second
Tbps	Terabits per second = 1,000,000,000,000 (one trillion) bits per second
LAN	Local Area Network
WAN	Wide Area Network
WLAN	Wireless Local Area Network
ISDN	Integrated Service Digital Network
ADSL	Asymmetrical Digital Subscriber Line
ATM	Asynchronous Transfer Mode
FDDI	Fibre Distributed Data Interface
WAP	Wireless application protocol
WTP	Wireless Transaction protocol
WTLS	Wireless Transport Layer Security
VOIP	Voice over Internet Protocol
GDP	Gross Domestic Product
NIC	Network Interface Card
CSMA/CD	Carrier Sense Multiple Access with Collision Detection
SD	Starting Delimiter
ED	Ending Delimiter
AC	Access Control
BRI	Basic Rate Interface
PRI	Primary Rate Interface
MPOA	Multiple Protocol over Asynchronous Transfer Mode
QoS	Quality of Service
VoD	Video on Demand
PVC	Permanent Virtual Connection
PDN	Public Data Network

IR	Infrared
PC	Personal Computer
MAC	Media Access Control
BSS	Basic Service Set
AP	Access Point
ISM	Industrial Scientific Medical
IEEE	Institute of Electrical and Electronics Engineers
VHF	Very High Frequency
UHF	Ultra High Frequency
DSSS	Direct Sequence Spread Spectrum
FHSS	Frequency Hopping Spread Spectrum
TCP	Transmission Control Protocol
IP	Internet Protocol
IETF	Internet Engineering Task Force
HTTP	Hyper Text Transfer Protocol
FTP	File Transfer Protocol
MOS	Mean Opinion Score

Chapter One

Introduction

1.1 Introduction

The need for resource sharing and communication capabilities was behind the evolution of computer networks. One important and frequently used type of computer network is the Local Area network (LAN). LANs are characterized by limited range, and high speeds [1]. Wide Area Network (WAN), on the other hand, are characterized by a large operational range, at much lower speeds.

Communication networks are being highly integrated, and the complexity of their hardware and software architecture is ever increasing. One outcome of this trend is that the performance studies of such integrated networks, and interaction of protocols running on them are becoming extremely difficult [2]. High-speed network-based distributed systems often observe performance problems such as unexpectedly low network throughput or high latency. The reasons for the poor performance can be diverse and are frequently not understood [3]. It is often hard to track down performance problems because of the complex interaction between many distributed system components, and the fact that the performance problems in one place may be most visible somewhere else. Bottlenecks can occur in any of the components along the paths through which the data flows: the applications, the operating system, the device drivers, the network adapters on any of the sending or receiving hosts and/or in network components such as hubs, switches, bridges, and routers. One of the most demanding applications that current LANs/WANs support is server access [3]. With increased processor speeds and the number of workstations attached to the networks, network utilization has been increasing with detrimental effects on the overall network performance. It is difficult for users to verify in advance whether a given network technology will deliver the expected performance benefits, or to verify how to tune their existing system to take full advantage of high-speed network [4].

Over the last decade we have been witnessing a wonderful increase in the capabilities of our computing and communications systems [5]. Bandwidth rose three times faster than the computer power. This means that if the computer power doubles every eighteen months (per Moor's Law), and then communication power doubles every six month. This rate of increase is also known as Gilder's Law [6].

Based on Gilder's Law, one would conclude that the performance of distributed/networked application would improve at similar rates. Unfortunately, this is not the case, partly because interprocess communication does not improve at the rates predicted by Gilder's Law. These attractions led us to study the congestion problem, which is described in section 1.2.

1.2 Problem Statement

Eritrea is a small country on the northeast coast of Africa (Figure 1.1). Its area is about 124,300 sq. km with a population of 4.5 million. It is bounded in the northeast by the Red Sea, Saudi Arabia and Yemen, southeast by Djibouti, south by Ethiopia and west by Sudan [7]. The density of population exceeds around 27 people per sq km. The country has a coastline of about 1,200 km and 354 islands that are mostly uninhabited, and holds a strategic geopolitical position along the business shipping ports of Massawa and Assab. Surrounded by neighbouring countries with similar cultures and languages. Eritrea has nine ethnic groups: Tigre, Tigrinia, Arabic, Afar, Saho, Nara, Kunama, Belien and Hadrab. The official languages are Arabic, Tigrinia and English. The main economic activities in Eritrea are agriculture, fishing, salt, mining, and small industries such as food processing beverages, clothing and textile. The Eritrean population is divided equally between Muslims and Christians [7], [8].

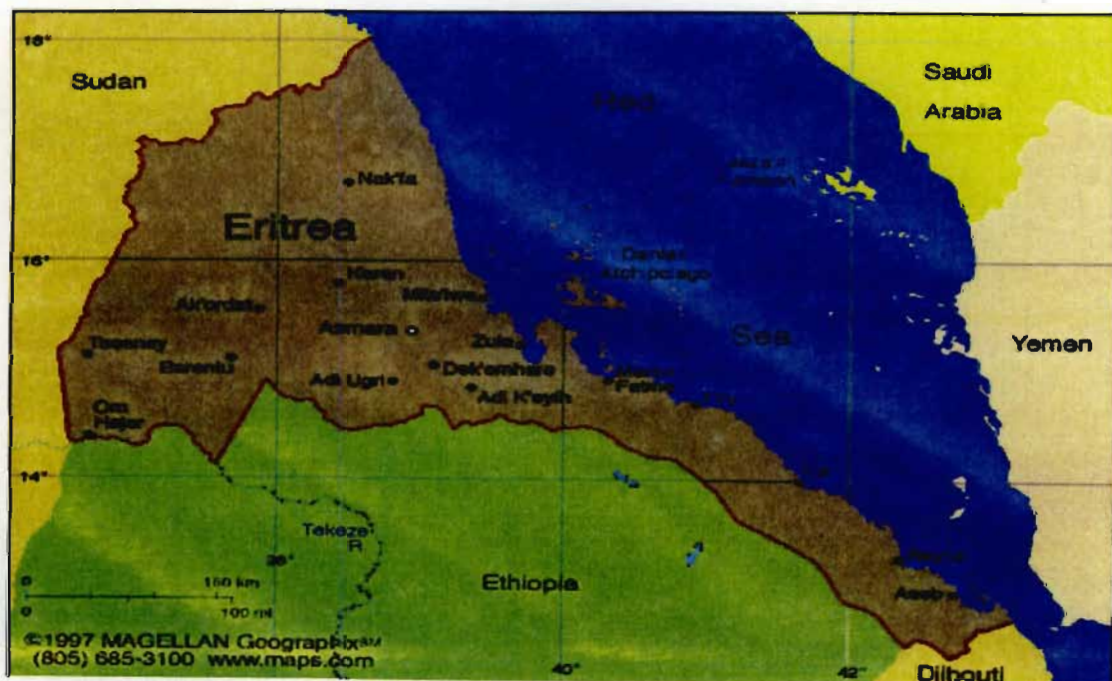


Figure 1.1 Map of Eritrea

Since independence from Ethiopia on 24 May 1993, Eritrea has faced the economic problems of a small, desperately poor country. Like that of many African nations, the economy is largely based on subsistence agriculture, with 80% of the population involved in farming and herding.

The Telecommunication Service of Eritrea is the national telecommunication operator in Eritrea responsible to the Eritrean Post and Telecommunications Authority of the Ministry of Transport and Communications. The communication department was established in 1996 as the regulatory body of the telecommunication sector [9]. In March 1998, the Eritrean Government issued a “Communication Proclamation” that regulates communications (telecommunications, broadcasting and postal services) in the State of Eritrea. Microwave transmission systems are used to interconnect urban centers, while fiber optic links are deployed in Asmara and Massawa primarily to interconnect exchanges (switching equipments) within the city [10].

The main international Internet gateway was started with 64Kbps. During 2002, the bandwidth was upgraded to 2 Mbps – inbound to Eritrea and 1 Mbps – outbound from Eritrea [10]. Internet service is provided by four agencies: Telecommunication, Eritrean Wide Area Network, Computer Technology Solution, and Tfanus Agency.

The major data communication network deployed in the country is LAN; WANs are also used on a small scale using dial up network to connect different LANs. Wireless Local Area Network (WLAN) is also deployed in some organisations to solve the barrier of data cabling problems. The typical speed of LAN is 10 Mbps, while the highest capacity link of a WAN is 56 kbps. The network connection devices used are hubs, switches, bridges, and routers.

In the Eritrea WAN, a serious congestion problem is experienced by different organizations such as the banks, government Department, and other organizations. Many users experience Internet failure when browsing web pages, down loading files, opening/sending mail, or opening application software such as Oracle, SQL, and Sybase. Multi-layered problems on Wide Area Network such as insufficient capacity, intermittent connectivity, packet loss, and greedy applications can all contribute to poor performance at remote branches. There is also the collision problem experienced in LANs. Finally there is need for different organizations to expand there networks from a single LAN to multiple LANs.

The congestion problem leads to performance problems. Possible reasons for the low performance problem could be: the operating system, the device drivers, the network adapters on any of the sending or receiving hosts and/or in network components such as hubs, switches, bridges, and routers. This study proposes to investigate and examine the network system to understand the problem and propose possible solutions that can assist in improving the performance of the networks.

1.3 Aim and objectives

The aim of this study is to investigate the performance of LAN, WAN, and WLAN in Eritrea. The specific objectives of the study are to:

- Investigate the disadvantages of network design and implementation of LANs WANs in Eritrea, and propose appropriate solutions.
- Examine and compare the performance of typical WAN traffic, (e.g. through Asmara LAN and Massawa LAN) via single path capacity link (19200 bps) and dual path capacity link (9600 bps).
- Examine the efficiency and effectiveness of LANs, WANs, and WLANs traffic, by studying the following parameters:
 - ❑ Network traffic throughput (via File Transfer Protocol (FTP), Email System, and Database System)
 - ❑ Packet loss ratio
 - ❑ Collision
 - ❑ Queue Delay
- Examine the performance of VoIP on the current Eritrean networks, through the following parameters:
 - ❑ Packet Loss
 - ❑ Delay

1.4 Structure of the thesis

Chapter 2 provides an overview of technology involved in LAN, WAN, and WLAN, network performance and previous work. Chapter 3 describes the tools and methods used in the study. Chapter 4 gives a detailed description of the traffic analysis between Asmara and Massawa using the Markov queuing model, LAN, and WLAN simulation result and their discussion. Chapter 5 studies the performance of VoIP in Eritrean WAN environment using speech coders (G723.1, G729, G711) with different capacity links such as 28kbps, 33kbps, 64kbps, and 128kbps. The conclusion and relevant recommendations are in chapter 6.

Chapter Two

Overview of Network Technology and Network Performance

2.1 Introduction

Network technology uses a variety of communication to achieve data, voice, and video connection between network users [11]. The goal of this chapter is to introduce a review for the various technologies used in LAN, WAN, and WLAN by discussing: the components of LAN, LAN topologies, connection devices, LAN cables, speed of LAN and WAN, standard technology of WLAN, and protocols. The chapter also gives an idea about network performance by discussing the concepts of: availability, response time, network utilization, and network throughput.

2.2 Local Area Network

A Local Area Network is a combination of hardware and software technologies that allow computers to share a variety of resources such as data, application programs, printers, and storage devices. A LAN allows messages to be sent between attached computers [12]. LAN services a limited geographic area at high speeds; usually 10 Mbps or higher [13]. A LAN consists of wiring, switches, hubs, and network interface cards that provide connectivity among clients, servers, printers, and other network-related elements in a local area. A LAN allows data and applications to be shared on multiple computers. A LAN also allows applications and or files to be accessed on a central server via wired connection [14]. Without a LAN it is difficult or impossible for companies to exploit information technology efficiently or competitively, given recent trends in computing.

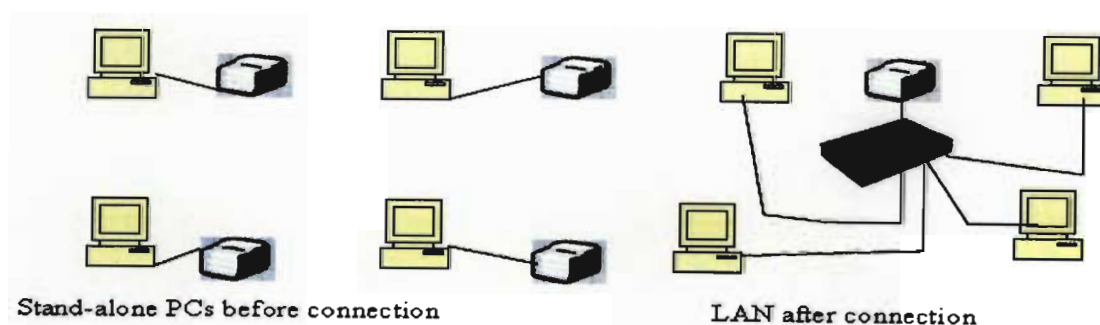


Figure 2.1 describing LAN connection

2.2.1 Why Local Area Network

LANs are recognition of the need for people to use data and, as a by-product, to pass data from one person to another [18]. There are many reasons for building networks. Networks can be built for academic reasons, research reasons, or for improving the performance and the efficiency of private and public organizations.

2.2.2 Local Area Network Topologies

Local area network topologies can be described using either a physical or logical perspective. Physical topologies describe the geometric arrangement of components that make up the LAN. The topology is not a map of the network. It's a theoretical construct that graphically conveys the shape and structure of the LAN [15]. A logical topology describes the possible connections between pairs of networked endpoints that can communicate. There are four basic types of LAN topology:

2.2.2.1 Bus Topology

A bus topology has a common central cable and all the devices are connected directly to it as shown in Figure 2.2. Each device has access to other devices on the network and can communicate directly.

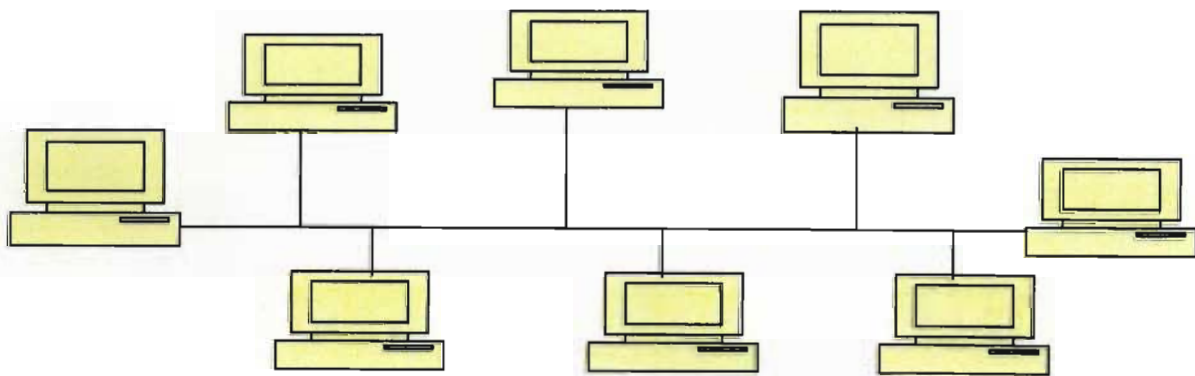


Figure 2.2. Bus topology.

A bus topology used as a network backbone is also appropriate for a temporary network or a small workgroup that has fewer than 10 computers. The advantage of bus topology is easy to reconfigure the network (add and remove users), and the transmission medium is highly reliable

(a failure of any workstation does not affect operation of other workstations [19]. The major disadvantage of the bus topology is the collision problem and cabling failure at the workstation will cripple the network [20].

2.2.2.2 Star Topology

One of the oldest types of network topologies is the star, which uses the same approach to sending and receiving messages as a telephone system [21]. In a star topology, each device connects to a central hub with a dedicated cable. One end of the cable is connected to the network interface card (NIC) on the device and the other end connects to a port on the hub. Figure 2.3 shows a star topology.

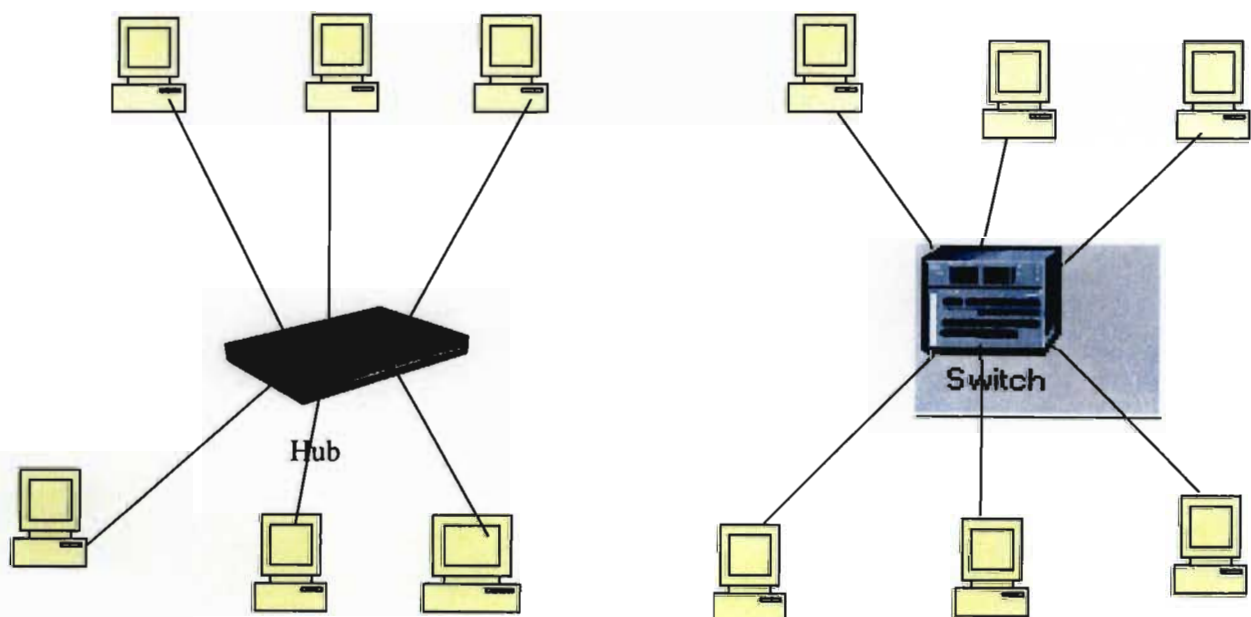


Figure 2.3 Star topology with Hub and Switch

A star topology is easy to install and expand or modify. The star topology is easy for troubleshooting, if one device fails, the other devices on the network are not affected. The hub can be used to isolate the problems. If the hub fails, the network also fails [18]. The advantage of star topology is that the network administrator can assign certain nodes higher status than others. The central computers will then look for signals from these higher-priority workstations before

recognizing other nodes. For networks with a few key users requiring immediate response from on-line inquiries, this feature of star topology can be extremely useful [21].

2.2.2.3 Ring Topology

Token Ring topologies continued the trend of sharing the transmission medium but differed from Ethernet in the critical area of the access method by regulating who can use the cable and when [22]. In ring topology, devices are connected in a ring or continuous loop on a cable, and there is no centralized device or hub. Each device connected to two devices in the circle. The ring topology is shown in the figure 2.4.

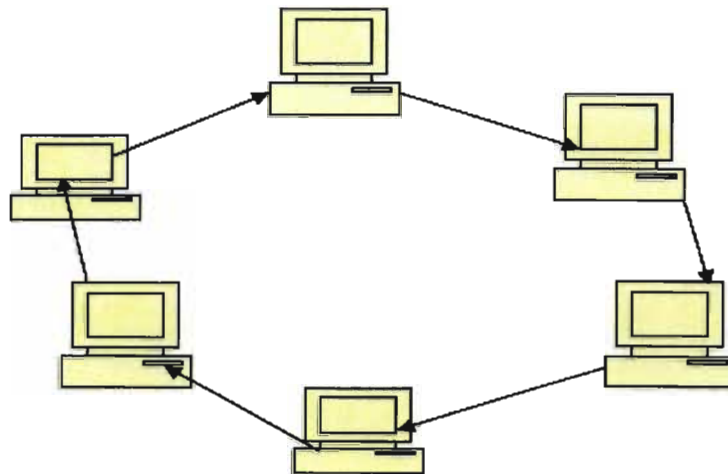


Figure 2.4. Each device is connected to the devices on either side of it in a ring topology

The ring topology can offer a high bandwidth, by providing each device with equal access to the network. The installation in ring topology is not difficult, adding or modifying devices on ring topology will interrupt the network operation, therefore the network might be required to shut down temporarily [20].

2.2.2.4 Mesh Topology

The mesh topology is used for direct link between every device on the network. Figure 2.5 shows a mesh topology.

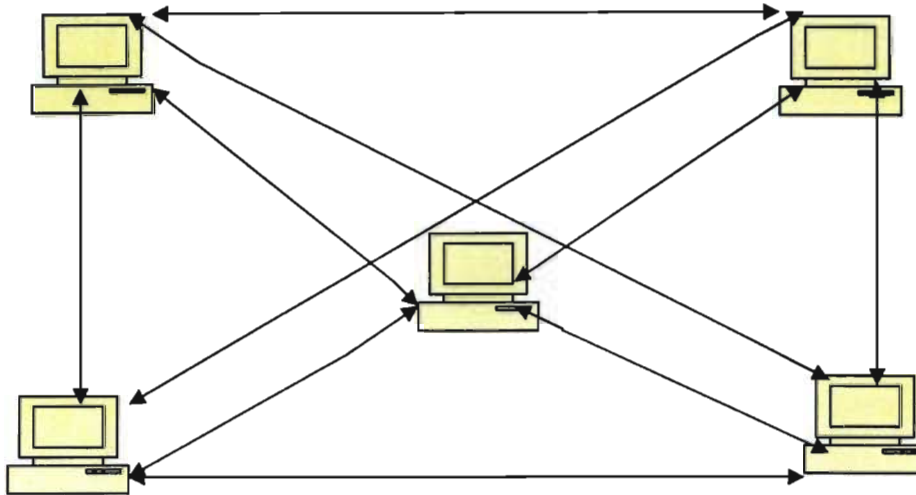


Figure 2.5. Each device mesh connected devices

Installation and management are more complicated in mesh topology. For example if the network contained 5 devices as shown above, 10 connections would be required. If the network had 6 connected devices, the number of the connection will increase to 15 connections. So, practically the mesh topology is not fair to use for a big LAN. It may be used for small peer-to-peer workgroup [21].

2.2.3 Types of LANs

Local Area Network consisting from three major types:

2.2.3.1 Ethernet

Ethernet was originally developed by Xerox Corporation in the mid-1970s and later modified by Digital, Intel, and Xerox (working together) [23]. This first incarnation was a 2.94 Mbps CSMA/CD (Carrier Sense Multiple Access with Collision Detection) system, which was used to

connect more than 100 workstations on a 1 km cable. Ethernet is a physical layer technology for local area networks. Ethernet networks can be built using twisted-pair, coaxial, and/or fibre optic cables. Ethernet network can use a bus or a star bus topology. IEEE 802.3 defines the transmission speed for Ethernet at 10 or 100 Mbps. Ethernet network uses CSMA/CD to control media access. This means each device on Ethernet network checks the network cable for traffic before sending a data transmission [15]. In 1995 IEEE defines a new standard, which is called Fast Ethernet (100 Mbps) in the form of IEEE 802.3u, this standardisation achieved ten times greater bandwidth and other new features such as full-duplex operation. In 1996 IEEE also announced 802.3z Gigabit Ethernet standardisation project, eleven companies founded the Gigabit Ethernet Alliance. However, the association increased to include more than 120 companies working in the fields of the networking, computers, and semiconductor industries. Ethernet technology is currently the most deployed technology for high-performance LAN environments. There is a need for faster network technologies to solve increased bandwidth demands. 10 Gigabit Ethernet has ten times the performances over Gigabit Ethernet today. With the addition of 10 Gigabit Ethernet to the Ethernet technology family, a LAN now can reach further distances and support even more bandwidth. The cost of Ethernet was very high, in early 1980. For example the cost of Ethernet Network Interface Card was around (\$600 or \$700). A better, more flexible NIC can be acquired today for under \$50 [18]. With switches, Ethernet networks are divided into segments small enough for collisions between nodes not to be a problem. The segments communicate with one another over dedicated (not shared) paths through the switch. The result is that each client station gets a real 3-4-Mb/s connection, which is all that most of them really need. The Fast Ethernet and Gigabit Ethernet lines seldom serve individual client stations, but are used, rather, to connect switches to each other and to servers [24].

Over the last few years we have seen Ethernet come out as the dominant technology for LANs and venture networking. Ethernet has also begun to make inroads as a network solution for storage facilities in corporate/hosting-collected data centres, and as an interconnect solution between Internet Service Provider points of presence (POPs) in Metropolitan network [25]. Fast Ethernet refers to a set of specifications developed by the IEEE 803 committee to provide a low cost, Ethernet compatible LAN operating at 100 Mbps. The blanket designation for these standards is 100Base-T. The committee defined a number of alternatives to be used with different transmission media [14]. Figure 2.6 shows the alternative media of IEEE 802.3 (100Mbps)

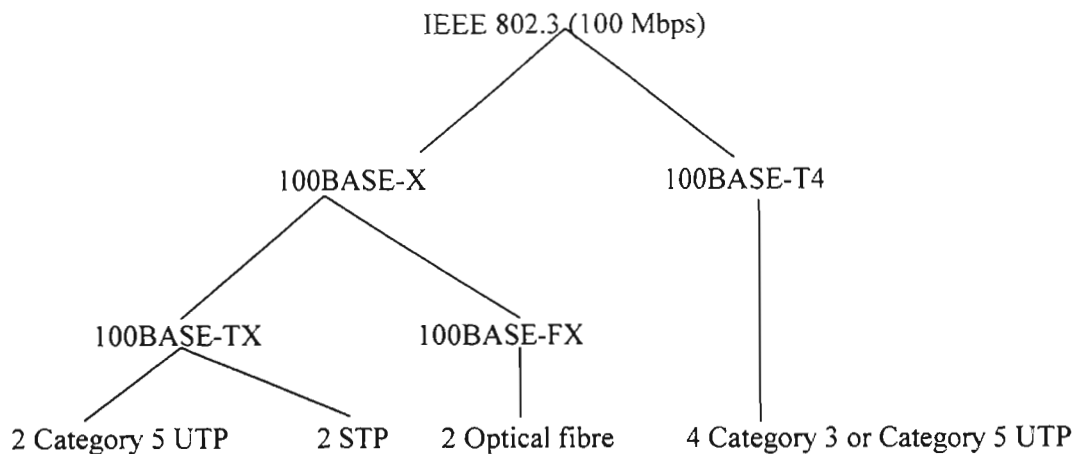


Figure 2.6 alternative media with IEEE 802.3 (100 Mbps) [13]

Fast Ethernet is a variation of Ethernet that can transmit data at speeds up to 100 Mbps over standard Ethernet cabling [23]. For traffic characteristics (bursty traffic, connectionless traffic without real time requirements) as in cases of sporadic file transfer, office communication or distributed data processing requiring the cooperation of different computers. Ethernet technology is particularly suitable [26].

2.2.3.2 Token ring network

The token is defined by a unique singling sequence that circulates on the medium following each information transfer [18]. There are two basic formats in token rings: tokens and frames. The token consists of three eight-bit sequence, or octets. The first octet is the starting delimiter (SD), the second is access control (AC), and the third is the ending delimiter (ED). When the token is captured, the station modifies it to a start-of-frame sequence and appends additional fields to make it a complete frame. When the information transfer is complete, the sending station generates a new token [18]. The first Token ring network architecture was attributed to Olaf Soderblum, who proposed such a network in 1969 [12]. The 802.5 standard was an outgrowth of research conducted by IBM. IBM's Token Ring Network, as developed by Texas Instruments, is an implementation of the standard [18]. Token ring network was available in the mid-1980 and was the basis for the IEEE 802.5 standards [18]. The speed of a token ring adapter card was 4 Mbps in 1985. An update released in 1989 addressed both 4 and 16 Mbps. The media that is used in a token ring was a shielded twisted pair, but currently it uses both media STP and UTP [18].

A token ring network that operates at both 4 and 16 Mbps supports up to 260 devices when using a shielded twisted-pair, or 72 devices using a telephone twisted-pair [21]. At 16 Mbps, the token ring network supports a frame size up to 18K (compared to the limit of 2K in the 4-Mbps) [21]. In a token ring, each device has an upstream and downstream neighbour. If one device wants to send a packet to another device on the same ring, it sends that packet to its downstream neighbour, who forwards it to its downstream neighbour, and so on until it reaches the destination [27]. A token ring network uses a token-passing method for media access. A token is a signal that moves around the network in sequence from device to device. The network in token ring can only operate at one speed, either 4 Mbps or 16 Mbps [20]. The token ring technique is based on the use of small frame, called a token that circulates when all stations are idle. A station wishing to transmit must wait until it detects a token passing [14]. The principle disadvantage of token ring is the requirement for token ring maintenance. Loss of token prevents further utilisation of the ring. Duplication of token can also disrupt ring operation [14]. Token ring network is a non-connection network. Because of the nature of token ring architecture, only one network node can send information at any given time [21]. The performance of token ring network is determined by the time it takes the token to go round the ring [22]. In a token ring LAN, the N nodes of LAN (hosts and routers) are connected in a ring by direct links. The topology of token ring defines the token-passing order. When node obtains the token and sends a frame, the frame propagates around the entire ring, thereby creating a virtual broadcast channel. The node that sends the frame has the responsibility of removing the frame from the ring [28].

2.2.3.3 Fibre Distributed Data Interface (FDDI)

FDDI is a 100 Mbps network architecture that was first specified in 1984 by the American National Standard Institute (ANSI) subcommittee entitled X3T9.5, responsible for data interfaces, to sponsor the development of FDDI [11]. FDDI is based on a dual ring-of-trees topology and its legitimate subsets, including a dual ring without trees and single tree. It enables flexible networks to be configured with path lengths ranging from a few meters up 100 km, accommodating up to 500 stations [29]. FDDI has been accepted as a standard since 1990, yet high cost has meant that companies have not implemented it widely, preferring to use it only on their LAN backbones. The cost of FDDI adapters has caused many companies to look at a twisted-pair wire alternative [21].

FDDI is a high-speed token passing ring network that uses fibre-optic transmission media, usually configured in a star ring topology. FDDI is commonly used in short runs near mainframes and minicomputers, for network backbones, and for networks that require high data rates and abundant bandwidth, such as high-end corporate networks [20]. FDDI LANs can be used to interconnect a company's departmental LANs in a metropolitan area. For example, a university might use an FDDI backbone to connect the LANs in different departments, or a government agency might use an FDDI backbone to connect departmental LANs in different buildings, as illustrated in figure 2.7.

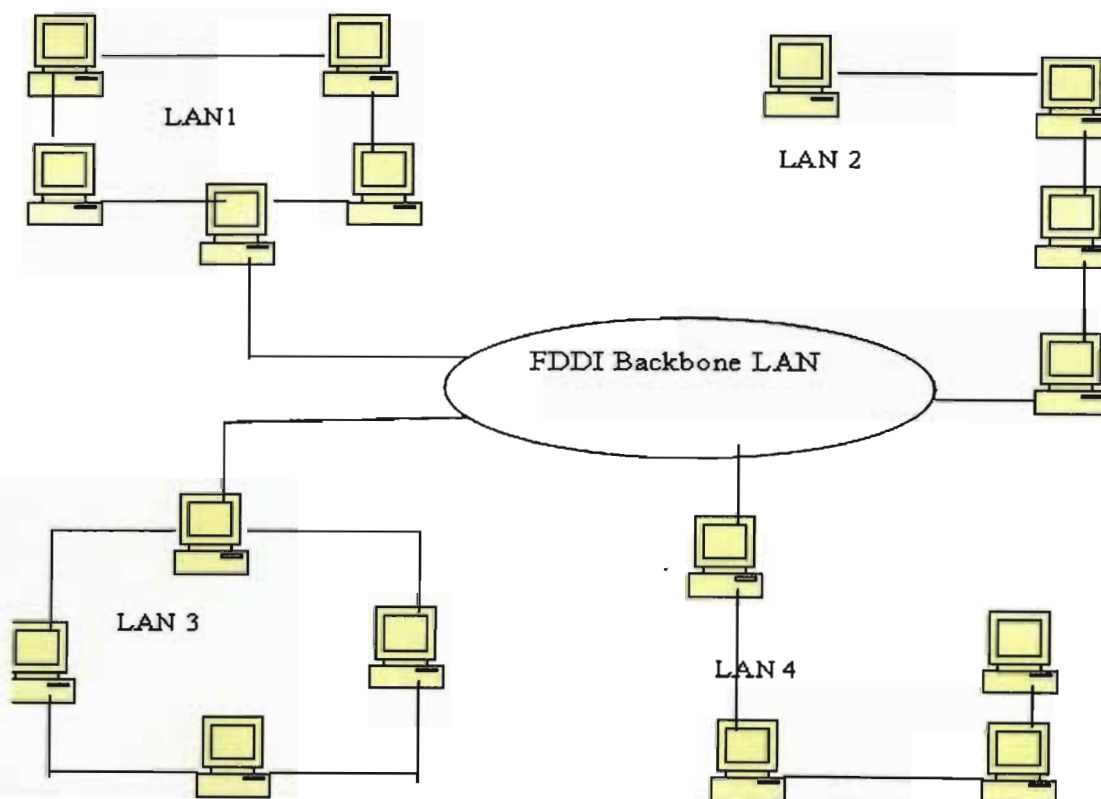


Figure 2.7 Connection of FDDI Backbone

2.2.4 LAN Network Operating System

LAN could use the following Network Operating System:

- ☐ Linux operating system for campus network
- ☐ Windows 3.11 for small business
- ☐ Windows NT 4 or Windows 2000 for server
- ☐ Novell Network operating system
- ☐ Macintosh networking operating system
- ☐ OS/2 networking

2.2.5 Internetworking or LAN linking

The term internetworking refers to linking individual LANs together. Workgroup LANs on different floors of building or in separate buildings on a business campus can be linked together so that all of the computing systems at that site are interconnected. There are two main reasons for implementing multiple LANs and internetworking them. One is to extend the geographic coverage of the network beyond what a single LAN can support to multiple floors in a building, to nearby buildings and to remote sites. The second reason for creating internetworking is to share traffic loads between more than one LAN [30]. There are two major types of devices used for internetworking: bridges and routers.

2.2.5.1 Bridges.

Bridges connect two or more LANs, and they make informed decisions about forwarding incoming signals. A bridge looks at the destination and source addresses of each frame. Whenever a bridge receives a frame with a MAC address that does not reside on the LAN segment on which it was generated, the bridge looks up that address in its bridging table and forwards the frame to the correct LAN segment for delivering. A bridge can isolate exclusively local traffic, keeping local frames from entering neighboring LANs [20]. Figure 2.8 shows bridge with more than one LAN.

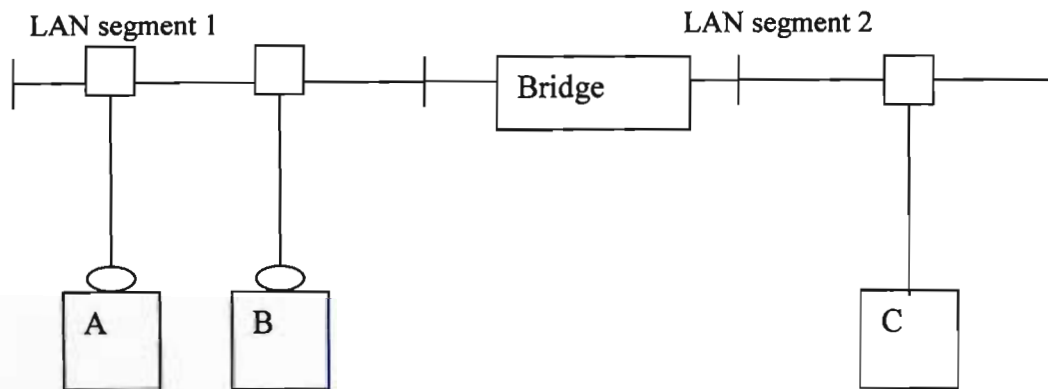


Figure 2.8 link between two LANs through a bridge

2.2.5.2 Router.

A router is a device that determines the way to transmit the data between two or more networks. A router can support different types of network technologies, such as Ethernet and Token Ring [15]. Figure 2.9 shows different LANs with a router.

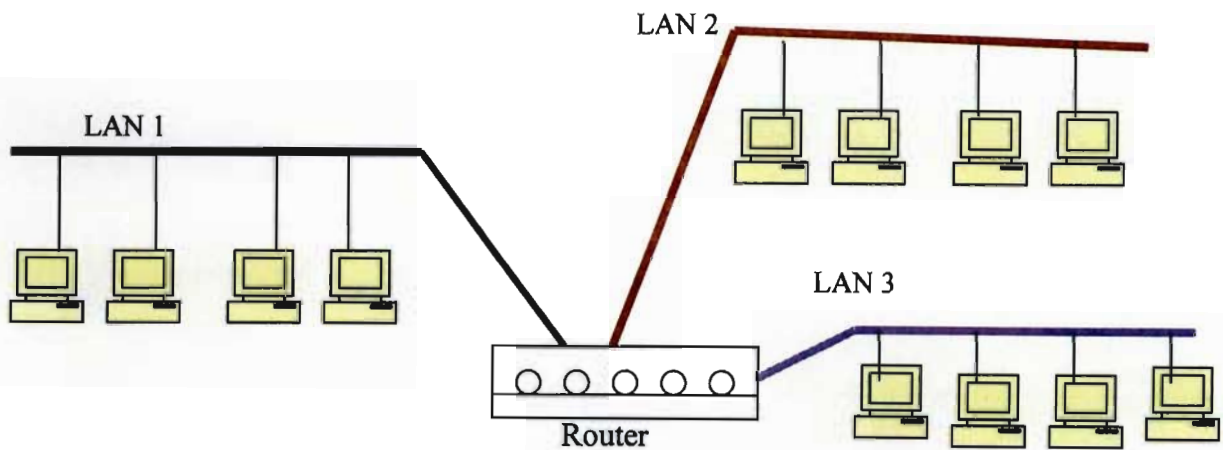


Figure 2.9 three LANs connected through a Router

2.2.5.3 Difference between Bridge and Router

This section will attempt to analyze the difference between the bridge and router. The bridge operates at the Data Link Layer (level 2) and does not understand anything about any communications protocol other than the physical medium, which is typically an Ethernet. A router operates at the Network Layer (level 3) and typically understands routing protocols inherent in TCP/IP or Decnet or XNS. Table 2.1 describes the difference between the bridge and the router through performance, network isolation, and cost [31].

Table 2.1 Comparisons between the Bridge and Router.

Issues	Bridge	Router
Performance	A bridge forward 2500 packet per second. It attempts to look deeper into each packet to perform such functions as security and access controls as the throughput drops	A router forward 1200 packet per second. It uses faster technology and special purpose hardware so that the throughput should rise.
Network Isolation	The traffic will propagate to the rest of the network.	The traffic is localized to the Ethernet segment where the “badly behaved” host is situated.
Cost	Costs less than a router since most of the box is customized hardware with very little software.	Costs are higher than bridge and it has simpler hardware but extensive software.

There are still major differences between router and bridge. If you have a small network (three to four Ethernet segments, with no more than 25 MAC address) then a routing bridge is the best solution. However, if your network comprises many segments and subnets and you have hundreds of MAC addresses defined, then a multiprotocol router is the best solution [31].

2.2.6 LAN Cables

The LAN must have cabling to link the individual workstation with the file server and other peripherals. There are many different types of cabling each with its own vocal supports.

2.2.6.1 Twisted-Pair Cable

Twisted-pair cable is bundled in groups of pairs. The number of twisted pairs per group can range from 2 to 3,000; many LANs use 25 pairs. The limitation of the cable length is 180 meters [21]. When a standard was first proposed for twisted-pair networks, the medium was able to handle transmission speeds of approximately 1 Mbps. Today, the industry standard, known as 10baseT, reflects the technological advances that make it possible to transmit information at 10 Mbps over a twisted-pair wire, and 100 Mbps transmission over unshielded twisted-pair wiring is emerging as a new standard [21].

2.2.6.2 Coaxial Cable

Coaxial cable is almost as easy to install and maintain. "Coax" is based on a central copper core encased in a plastic sheath that is then surrounded by an outer jacket composed of copper or aluminum acting as a conductor. This also provides protection. The signal is transmitted along the central core with the outer jacket forming a screen from outside electrical interference [21]. Thin coaxial cable is also referred to as thinnet. 10Base2 refers to the specifications for thin coaxial cable carrying Ethernet signals. The 2 refers to the approximate maximum segment length being 200 meters. In fact, the maximum segment length is 185 meters, and data rate is 10 Mbps. Thin coaxial cable is popular in school networks, especially linear bus networks. Thick coaxial cable is also referred to as thicknet. 10Base5 refers to the specifications for thick coaxial cable carrying Ethernet signals. The 5 refers to the maximum segment length being 500 meters and data rate is around 10 Mbps

2.2.6.3 Baseband Cable

Baseband coaxial cable has one channel that carries a single message at a very high speed. The advantage of baseband cabling is how easy it is to tap into this cable and connect or disconnect a

workstation without disturbing the network operation. The maximum recommended distance for a baseband LAN is approximately 3 kilometres [21].

2.2.6.4 Broadband Cable

Broadband coaxial cables have the capacity to carry several different signals broadcast at different frequencies simultaneously. All broadband systems can use a single cable with bi-directional amplifiers, or a dual cable system. Dual broadband cable uses one cable for inbound data moving toward the headend, and a second cable looped at the headend for the outbound carriers. The full-frequency spectrum is available for both inbound and outbound signals. Standard television channels transmit at 6 MHz. Because the band range approximately a round 300 MHz, it is possible to have as many as 50 channels broadcasting at data of 5 Mbps [21].

2.2.6.5 Fibre-Optic Cable

One of the most interesting advances in media is the use of fibre optics in LANs. This type of data transmission has a number of advantages over twisted pair and coaxial cable. Besides data transmission rates far higher than either of those older media, fibre-optic cabling is immune to electromagnetic or radio frequency interference and capable of sending signals several miles without loss [21]. Fibre optic cable can carry large amounts of data for long distances at high bandwidths with low attenuation. The disadvantage of fibre optic cable is the cost, it is still the most expensive type of cable to install [15]. Fibre-optic cable is capable of high bandwidth (200 Mbps) transmission but usually over a distance of less than 1 km.

2.2.7 LAN Benefits

In many organisations, a LAN is the foundation for a system of office automation. LAN also provides a base for constructing a solid system for the technology that can improve the capability, efficiency and effectiveness of the workforce. LAN is not only for office automation, it is beneficial for factory automation. LAN has opened a good way for communications such as an Email system, File Sharing and File Transfer.

2.3 Wide Area Network

WAN is a computer network that spans a large geographical area and is usually made up of two or more LANs. Typically, a WAN consists of a number of interconnected switching nodes. A transmission from any one device is routed to internal nodes to a specified destination [12]. WANs are typically too big, geographically, to have single entity owners. Whether they are private networks of large corporations, or of government, WANs use many kinds of available transmission media, and in most cases parts of WAN use public networks common carrier transmission service [15]. WAN design considerations include:

- ❑ Number of users in different parts of organization
- ❑ Number of workstations and servers in different parts of the organization
- ❑ Bandwidth capacity and speed of each link of the WAN
- ❑ Administration of network resource
- ❑ Security requirements throughout the network

WAN connections are mostly characterized by the cost of renting media (wire) from a service provider to connect two or more campuses together. The larger the enterprise the more cost effective it becomes to deploy WAN switching equipment that optimizes the usage of bandwidth available on trunk lines rented from telecommunication [32]. WAN traffic volumes are predicted to increase by 300 to 600 percent in the next few years [32]. WAN connection may run over media such as analogue modems, Leased lines, ISDN, Frame Relay, X.25 and WAN ATM [32].

2.3.1 Wide Area Network Topologies

WANs can be configured in three ways: hierarchical, interconnected, and a combination of both.

2.3.1.1 Hierarchical Network

The hierarchical network topology, shown in Figure 2.10, is also called a tree structure. There is one root node (Node A). Several nodes at second level are directly connected to Node A. Each of those nodes can have a number of cascaded nodes attached [13].

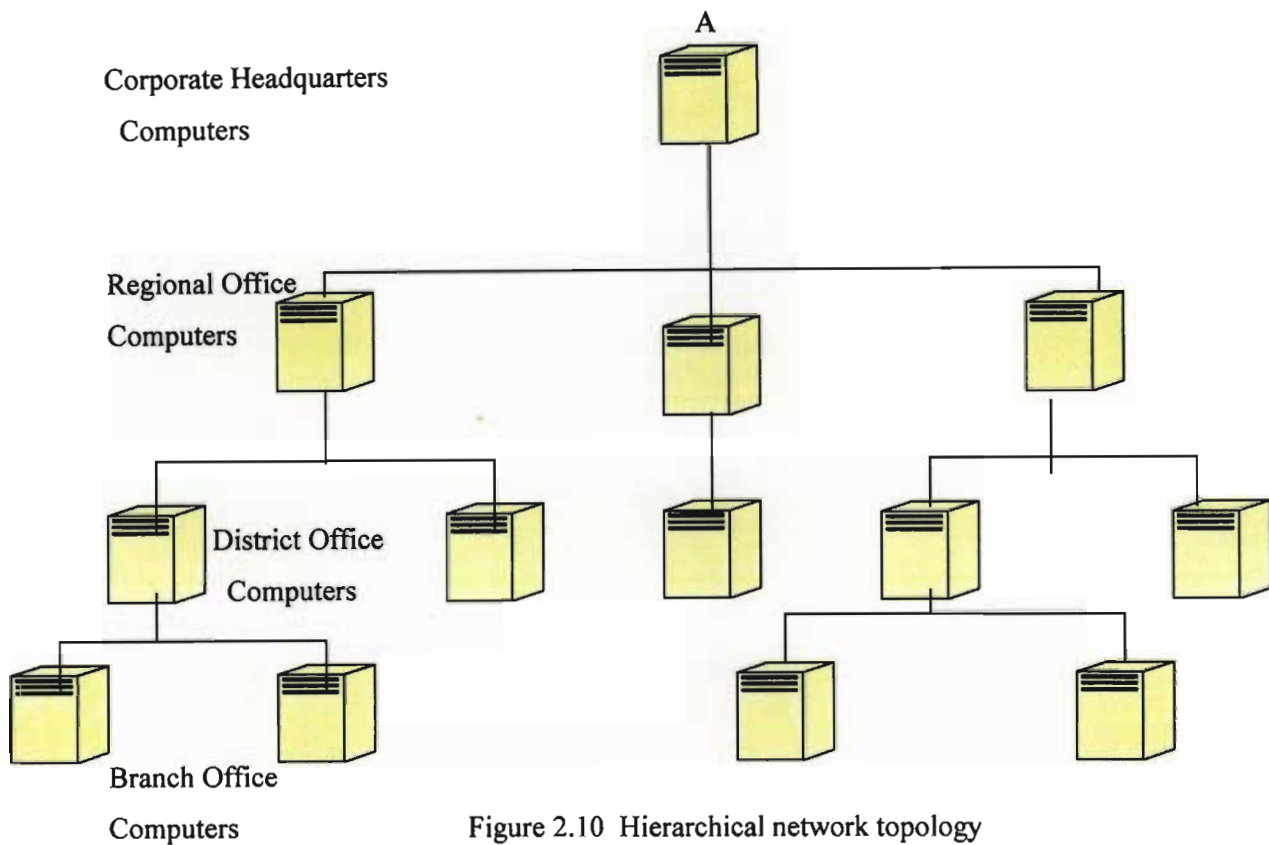


Figure 2.10 Hierarchical network topology

2.3.1.2 Interconnected Network

An interconnected network architecture is shown in Figure 2.11. This topology provides a high degree of flexibility because many paths are available between nodes. The performance of an interconnected system is generally good because direct links can be established between nodes with a high amount of data to exchange [13].

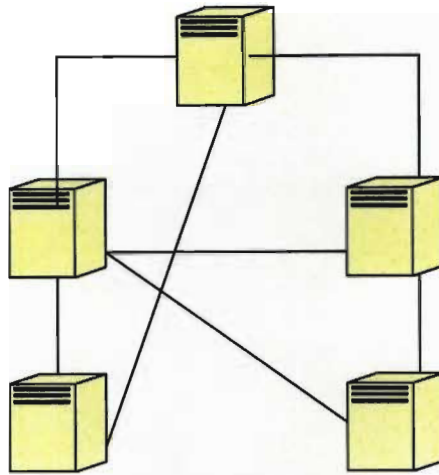


Figure 2.11 interconnected network topology

2.3.1.3 Combination Network

A combination of the two topologies just described is sometimes integrated into one network. One such combination is a backbone network. The backbone nodes are dedicated to message transfer and data communication, and the other nodes are used for both data processing and communication. In widely distributed systems with a large number of nodes, this configuration helps reduce the number of hops, the length of the path, and congestion problems. If the backbone is implemented as a ring or with multiple paths available, reliability is also high [13].

2.3.2 Types of WAN capacity link

2.3.2.1 Integrated Service Digital Network (ISDN) is a digital telephone technology based on regular copper telephone wire [15]. ISDN provides effective integration of technology and of service for interactive communication [32]. ISDN is designed to replace existing public telecommunication networks and deliver a wide variety of services. The ISDN is defined by the standardisation of user interfaces and implemented as a set of digital switches and paths supporting a broad range of traffic types and providing value-added processing services [14].

ISDN is intended to be a worldwide public telecommunications network to replace existing public telecommunication networks and deliver a wide variety of services [14].

ISDN also supports a variety of services related to voice communications (telephone calls) and non-voice communications (digital data exchange). In addition, ISDN supports circuit switching, packet switching, and non-switched services in the form of a dedicated line [14]. One ISDN line, with rates up to 128 kbps, can support half-dozen or more users. ISDN is commonly configured as a basic rate interface (BRI) or a primary rate interface (PRI). A BRI consists of 2 B (bearer) channels at 64 kbps and one D channel (for call set-up and take-down and other call management signalling) at 16 kbps. A PRI provides 23 individual 64-kbps B channels and one 64-kbps D channel [15]. Korea Telecommunication (KT) launched a commercial ISDN basic rate interface (BRI) service in December 1993. The number of ISDN users increased slowly until 1996 [34].

Because the demand for computer communication including Internet access grew rapidly, ISDN service became widespread in 1997. Since April 1997, ISDN primary rate interface (PRI) service has also become available. Through this interface, ISDN LAN service, which enables high-speed Internet access and LAN interconnection, is being provided for business subscribers. ISDN service is considered as one of the candidates to replace the dialup modem for Internet access. It is expected that the deployment of ISDN will be accelerated until new infrastructure based on ATM technology becomes available [34]. Broadband ISDN is a second-generation specification for ISDN that provides for high digital data rates [14]. Broadband ISDN is defined to support data rates over 600 Mbps [23].

Broadband ISDN was conceived as a network able to support the full range of imaginable services from low bit-rate telemetry to high rate and television transmission including voice, video, data, and multimedia. A key requirement for such applications was a high degree of flexibility on a number of timescales: long term to cope with uncertainties in service and their demands on the network, short term to cope with mobile customers and terminals, and instantaneous as customers switched connection and service [29].

2.3.2.1.1 International use of ISDN

In 1992 British Telecommunication (BT) found that internationally the main use of ISDN was for video Conferencing (52%), for File Data Transfer (25%) and with Kilostream Back-up in third place with 10% (see figure 2.12). The reason for the large amount of usage for video traffic is

attributed to many factors, one being an international standard for video which is easily transported to the ISDN and, a second, that many companies saw the advantage and cost saving of using videoconferencing [29].

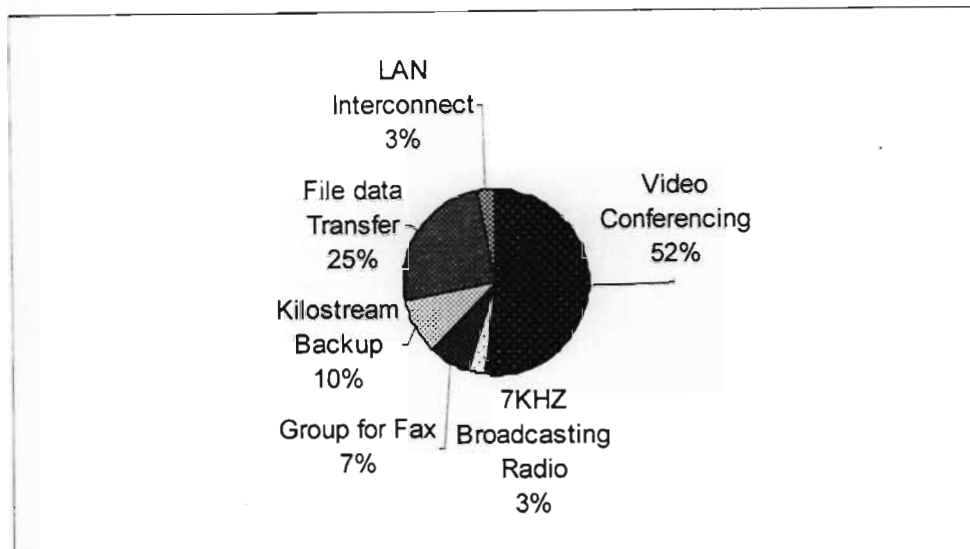


Figure 2.12 international use of ISDN [29]

2.3.2.2 Asynchronous Transfer Mode

ATM was chosen for the transmission technique in broadband ISDN because of its simplicity, its ability to transfer data rapidly between two points, and its ability to handle varying rates of data quickly. All data is packaged into fixed-sized units, called cells. Each cell consists of a 5-byte header followed by a 48-byte data field. ATM allows the multiplexing of numerous logical connections over a single physical connection [23]. Microsoft is satisfied that ATM is appropriate for transmission of combinations of data, video, voice, and teleconferencing applications [15]. ATM can transport WAN data at speeds much faster than Ethernet. Data transmission speeds of 155 Mbps and higher are available with ATM. The speed of ATM transmission is based on the line transmission technology being used, and ATM can be used with several. ATM implementations are available to transfer data at rates of from 25 to over 600 Mbps [15][34]. ATM network adapters and fibre-optic cable equipment are much more expensive than similar Ethernet. There are two real problems with using ATM in a large LAN. The first problem is the additional overhead of the various LAN Emulation servers required for either LANE or Multiple Protocol over ATM (MPOA) implementations. The second serious drawback is the high cost-to-bandwidth ratio [27]. Figure 2.13 describes the ATM cell structure. The 5-byte header carries sufficient information to route the cell across the network so that cells from different terminals

can be multiplexed together on the same bearer. On the bearer, the cell stream is continuous and cells are filled asynchronously as terminals have data to send. At the switch, the cell header is used to address a look-up table to determine the switch outlet required. The contents of the look-up table are pre-set by the management and control system. Since the switch is done in hardware, very high throughputs can be achieved; throughputs of the order of 10s or even 100s of gigabits/second are possible with today's technology [29].

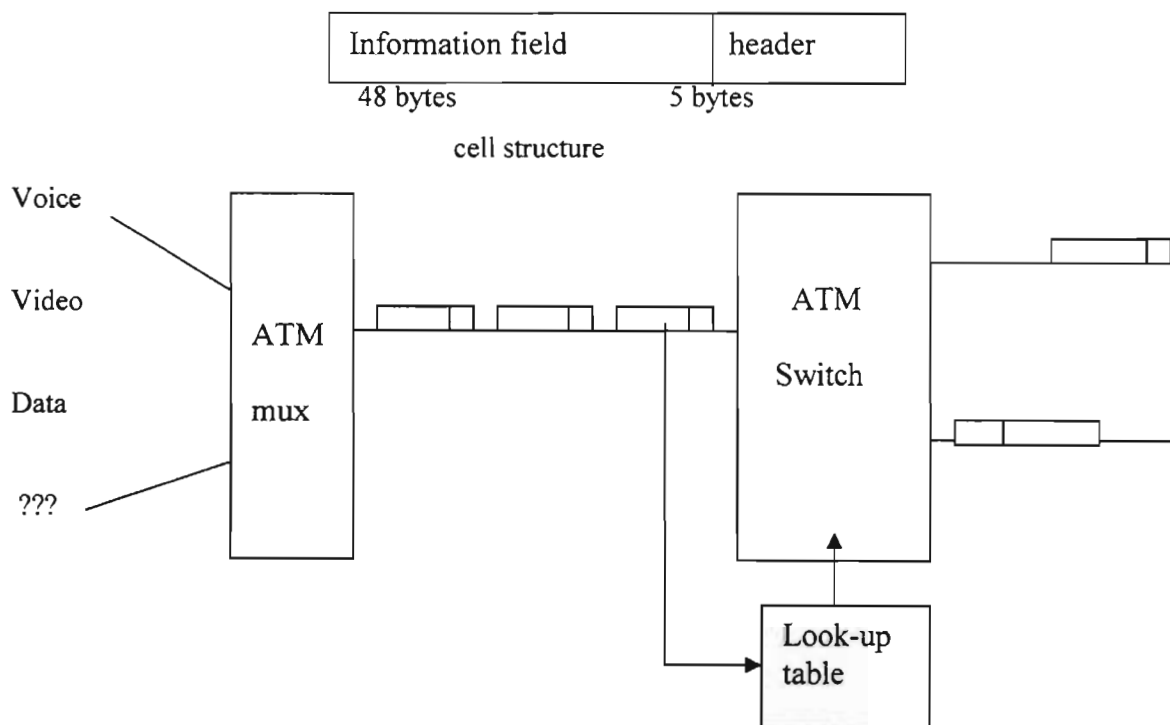


Figure 2.13 describing ATM structure

ATM is a natural candidate as the unifying transport solution for various types of data traffic over the public transport infrastructure. ATM is more than just a client adaptation mechanism, it provides a full complement of switching, multiplexing, and network function. ATM supports sophisticated traffic engineering, flexible quality of service and aware routing that provides granular partition of bandwidth, and multiservice integration [25].

2.3.2.2.1 The ATM switching hierarchy

There are four distinct types of ATM switches which make up the ATM switching hierarchy. Each type fulfils a distinct role. The complexity of each role is directly related to the scale and cost of the equipment [32]. The four main types of switch are:

Workgroup ATM switch

Workgroup ATM switches are primarily designed for non-delay sensitive Local Area Network traffic switching within a building or small campus, where bandwidth is inexpensive and plentiful [32].

Enterprise ATM backbone switch

Enterprise ATM backbone switches support engineering, non-delay-sensitive LAN interconnect traffic and legacy, delay-sensitive circuit switched data and voice traffic, within a campus, MAN or WAN environment [32].

Carrier/VAN service switch

Carrier and VAN services switches provide adaptation of multiple types of services, including circuit-switched and frame relay services, into ATM. They also optimise bandwidth usage with sophisticated traffic management and congestion control schemes, and have significant input and output buffers [32].

Core ATM switch

Core ATM switches have ATM interface only, and are used in very large enterprise VAN sites and the core of carrier networks. They provide traffic and congestion management and sophisticated trunk switching, and have capacities from 10 Gbps upwards [32].

2.3.2.3 Asymmetrical Digital Subscriber Line (ADSL)

ADSL is another telecommunication service possibly available from the local telephone company. ADSL requires a pair of special modems, one at your site and the other at the telephone company site. ADSL is generally cheaper than ISDN for the telephone to provide, and fast.

Speeds up to 5 Mbps are possible [15]. DSL was originally intended for video on demand (VoD). The video-over-DSL hype in the first half of the 1990s was not successful due to the need for large investment in centralised video equipment and major upgrades of the backbone network [36]. At the higher end, ADSL can transport more than 8 Mbps to the customer (downstream) and more than 1 Mbps upstream [36].

2.3.2.4 X.25

X.25 is a suite of protocols for a packet-switching network and is the oldest Wide Area Network switching method. X.25 networks use switched circuits to best route data based on current network conditions. X.25 was created to connect dumb terminals to mainframe computers, and requires error checking overhead and special packet assemblers and disassemblers to the network [15]. The use of X.25 involves a high-speed synchronous interface and requires a fair amount of software and computing power; therefore, when X.25 is used to connect to a Public Data Networks (PDN), it requires more than a simple dumb terminal. If a user does not have a workstation with sufficient power, or is using a low-speed asynchronous (DTE), X.25 cannot be used.

2.3.2.5 Frame Relay

Frame relay, designed in the late 1980s and widely deployed in 1990s as a service for the support of inter-LAN communications, is in many ways a second-generation X.25. Like X.25 it uses virtual circuits. It transmits data faster than X.25, because several portions of X.25 overhead were removed from its design [15]. Frame relay was naturally designed for much lower error rates. Frame relay is extensively used today to allow LANs on different corporate campuses to send data to each other at reasonable speeds [28]. Frame relay uses switched private virtual circuits (PVCs) to transmit frames of variable sizes. It can be configured for bandwidth on demand [15]. Frame relay networks are designed to operate efficiently at user data rates up to 2 Mbps [14]. Frame relay is an advanced packet-switching technology based on X.25. The frame relay standard defines a simplified packet switching protocol in which error control and flow control overheads are minimised. Further, routing of data packets at network nodes is performed on link level addresses, thus minimising the protocol-processing load. These features ensure a high data transfer rate with low transport delays on the transmission links together with high throughput at the network node [29]. Initially, a frame relay is offered as a Permanent Virtual Connection (PVC) service across the wide area between corporate network sites. Service data rates of 64kbit/s to 2Mbit/s are available now with evaluation to 34 Mbps (45 Mbps in North America) in the future.

Figure 2.14 shows in simple terms the provision of the frame relay service. Frame relay is an ideal vehicle for point-to-multipoint communications for LAN interconnection in offering an

HDLC-based variable length fast packet service [29]. Frame Relay is currently the most popular choice as X.25's resilience has become unnecessary in the light of the reliability of modern transmission equipment [37].

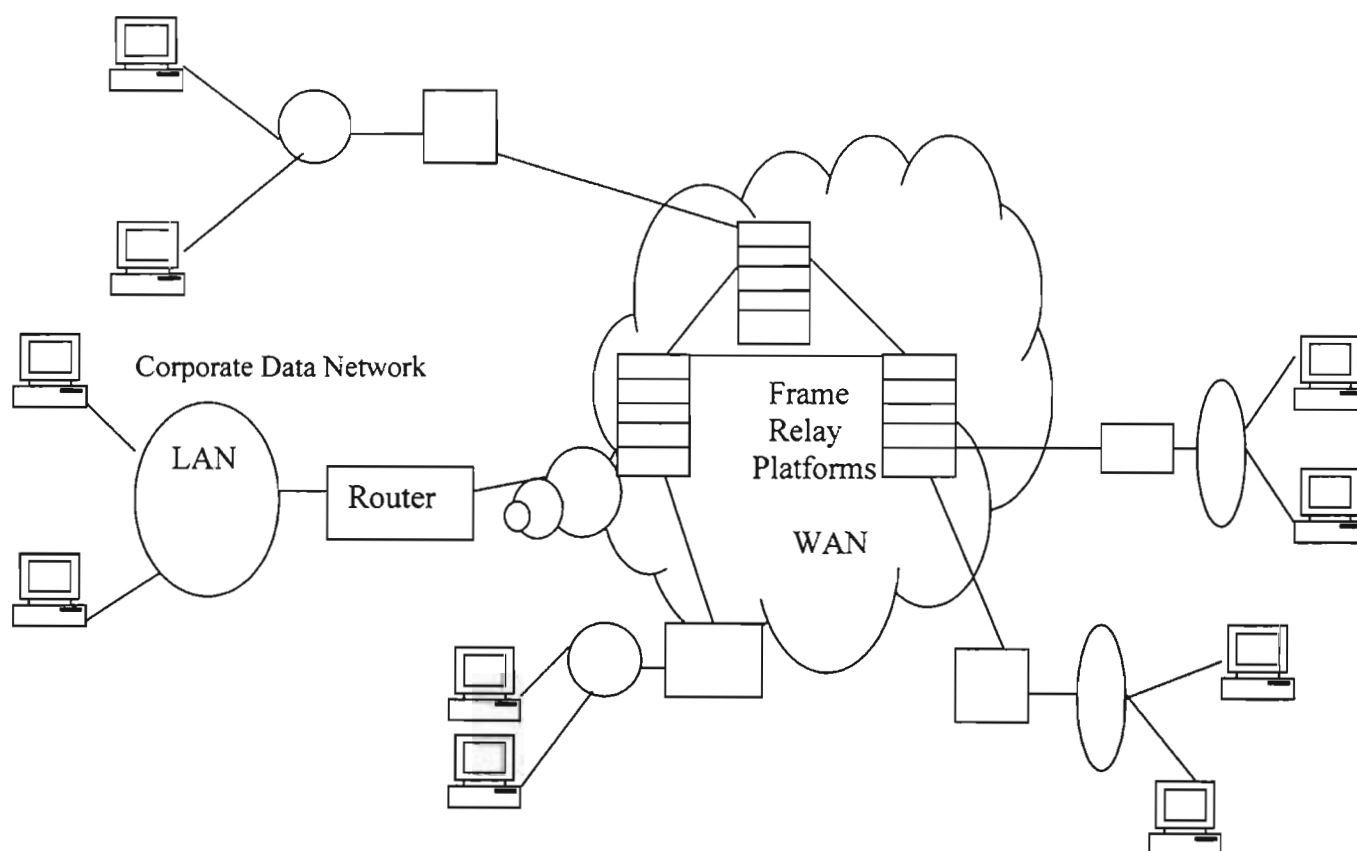


Figure 2.14 Frame Relay service concept

2.3. 3 Comparing WANs with LANs

Table 2.2 describing the comparison between LANs and WANs through distance, topology, protocols, routing and media:

Table 2.2 comparing between LAN and WAN

Issue	LAN	WAN
Distance	LAN serves a limited geographical area, typically within one building or building complex. The maximum allowable distance a LAN can cover is generally a few miles.	WAN can cover a large geographical area, or it can be limited to a small area.
Topology	LAN topologies are usually bus, ring, or star. Sometimes the topology is a combination of those, such as star bus and star ring.	WAN topologies are typically hierarchical or interconnected, although rings and stars are also used.
Protocols	LAN data link protocols are usually CSMA/CD or token passing.	WAN use asynchronous or synchronous protocols. The protocol of choice for most WANs today is a bit asynchronous Protocol such as HDLC.
Routing	In a LAN routing is either broadcast with CSMA/CD or always-transmit-to-the neighbour node in token passing.	WANs often use an adaptive routing technique, meaning that the routing may change based on conditions such as congestion, failed links, and failed processor or probability function for weighted routing.
Media	LAN media are usually twisted-pair wires, coaxial cable, fibre optic cable, or one of the new wireless media.	WAN media are often obtained through a common carrier and may consist of a variety of media, such as telephone wires, fibre optic cable, coaxial cable, microwave radio, and satellite.

2.4 Wireless Local Area Network

Wireless LAN transmits data between workstations and a file server using high-frequency radio waves (902-928 MHz). Using an omni-directional antenna, the workstation and file server may be separated by as much as 800 feet. It is also possible to incorporate a directional antenna, which then allows a distance of 8 km between workstation and file server. The network cards support the Ethernet protocol and simply consist of a single board inserted into an expansion card slot within the workstation. The simplicity of installation is clearly an advantage of a wireless local area network; a disadvantage is the relatively slow data transmission speed of 2 Mbps [23]. Figure 2.15 describes Wireless LAN installation.

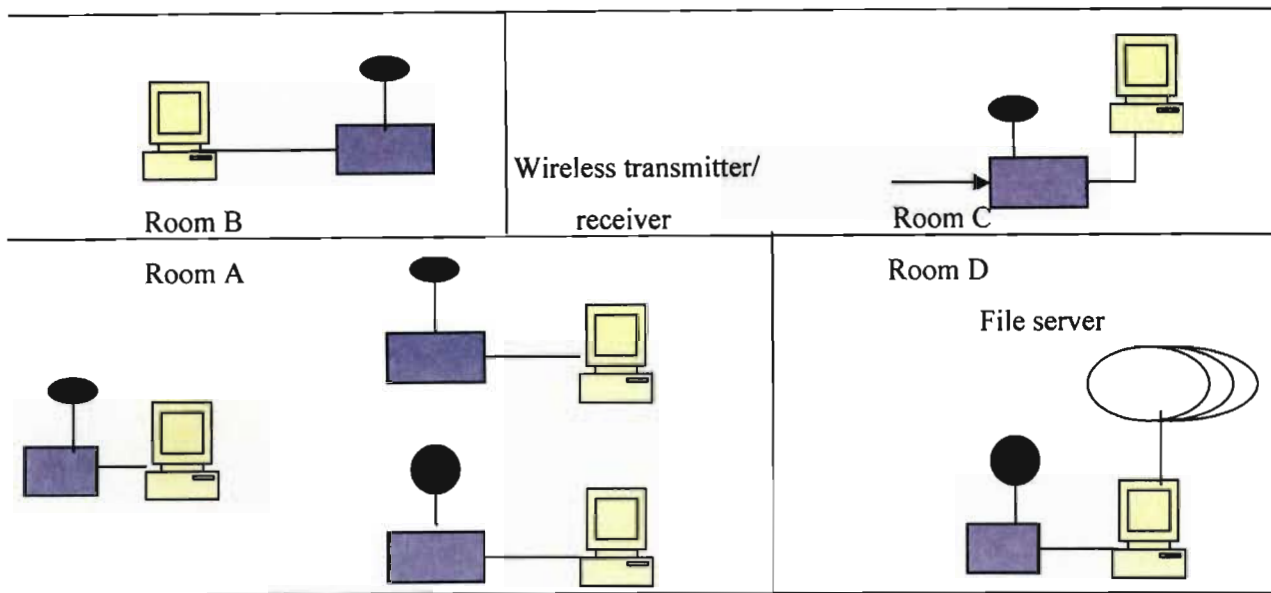


Figure 2.15 Wireless LAN workstations communicating with local file server

In recent years, wireless LANs have come to occupy a significant niche in the LAN market. Increasingly, organisations are finding that wireless LAN are indispensable to traditional wired LANs, to satisfy requirements for mobility, relocation, ad hoc networking, and coverage of location difficult to wire. Until relatively recently, wireless LANs were little used. The reason for this included high prices, low data rates, occupational safety concerns, and licensing requirements. As these problems have been addressed, the popularity of wireless LANs has grown rapidly [14]. WLAN transceivers can provide mobile connectivity in offices, hospitals, and factories, obviating the need cumbersome wired network [38].

Early wireless LAN products, introduced in the late 1980s were marketed as substitutes for traditional wired LANs. A wireless LAN saves the cost of installation of LAN cabling and eases the task of relocation and other modification to network structures [14]. However in a number of environments, there is a role for the wireless LAN as an alternative to a wired LAN. Examples include buildings with large open areas, such as manufacturing plants, stock exchange trading floors, and warehouse, older buildings with insufficient twisted pair cables are needed and where drilling holes for new LANs are not economical. In all of these cases, a wireless LAN provides an effective and more attractive alternative.

2.4.1 Wireless LAN Technology

Wireless LANs are generally categorised according to the transmission technique that is used. All current wireless LAN products fall into one of the following categories:

- (i) **Infrared (IR) LANs:** An individual cell of an IR LAN is limited to a single room, because infrared light does not penetrate opaque [14].
- (ii) **Spread spectrum LANs:** these LANs operate in the ISM (Industrial, Scientific, and Medical) bands, so that no FCC licensing is required for their use in the United States [14].
- (iii) **Narrowband microwave:** these LANs operate at microwave frequencies but do not use spread spectrum. Some of these products operate at frequencies that required FCC licensing, while others use one of the unlicensed ISM bands [14]

Portable or laptop PCs equipped with their own wireless LAN adapters can create an instant LAN connection merely by getting within the range of a server-based wireless LAN adapter or wireless hub. In this way, a student or employee can sit down anywhere and log into LAN as long as he/she is within range of a wireless hub and has the proper wireless adapter installed in their portable PC. These implementations are especially helpfully in large warehouse or inventory settings [12]. Meeting rooms could be equipped with wireless hubs to allow spontaneous workgroup to log into network resources without running cables all over the meeting room.

2.4.2 Wireless LAN Standard Technology

The IEEE standard for WLANs started in 1988 as IEEE 802.4L, a part of the IEEE 802.4 token bus wired LAN standard. In 1990 the IEEE 802.4L changed its name to IEEE 802.11 to form a standalone WLAN standard in IEEE 802 LAN standard organisation. The 802.11 standard specifies data rates up to 2 Mb/s using spread spectrum technologies in the 2.4 GHz ISM bands.

The HIPERLAN standard was developed by the RES-10 group of ETSI as a pan-European standard for high-speed wireless local networks. The so-called HIPERLAN I, the first technology defined by this standard group, began in 1992 and was completed in 1997. It supports data rates of 2-23 Mb/s using traditional radio modulation techniques in 5.2 GHz band [39].

2.4.2.1 IEEE 802.11 LANs

The IEEE 802.11 standard defines the physical layer and media access control (MAC) layer for the wireless local area network. The standard defines three different physical layers for the 802.11 wireless LAN, each physical layer in different frequency ranges and at rates of 1 Mbps and 2 Mbps [28]. The fundamental building block of the 802.11 is the cell, known as the basic service set (BSS) in 802.11 parlances. A BSS typically contains one or more wireless stations and central base station, known as an access point (AP) in 802.11 terminologies. The wireless stations, which may be either fixed or mobile, and the central base station communicate among themselves using the IEEE 802.11 wireless MAC protocol [28]. Figure 2.16 illustrates the principle components of the 802.11 wireless LAN architecture.

The IEEE 802.11 can operate in buildings and other premises such as shops, malls and hospitals as well as in outdoor areas such as campuses and factories. The IEEE project 802.11 has specified the different components of a wireless LAN. A single cell connects to the wired LAN through an access point (AP) [40]. Wireless Access points that are on the consumer market today were designed to connect a small group of trusted people to a wired network and lock out everyone else [41].

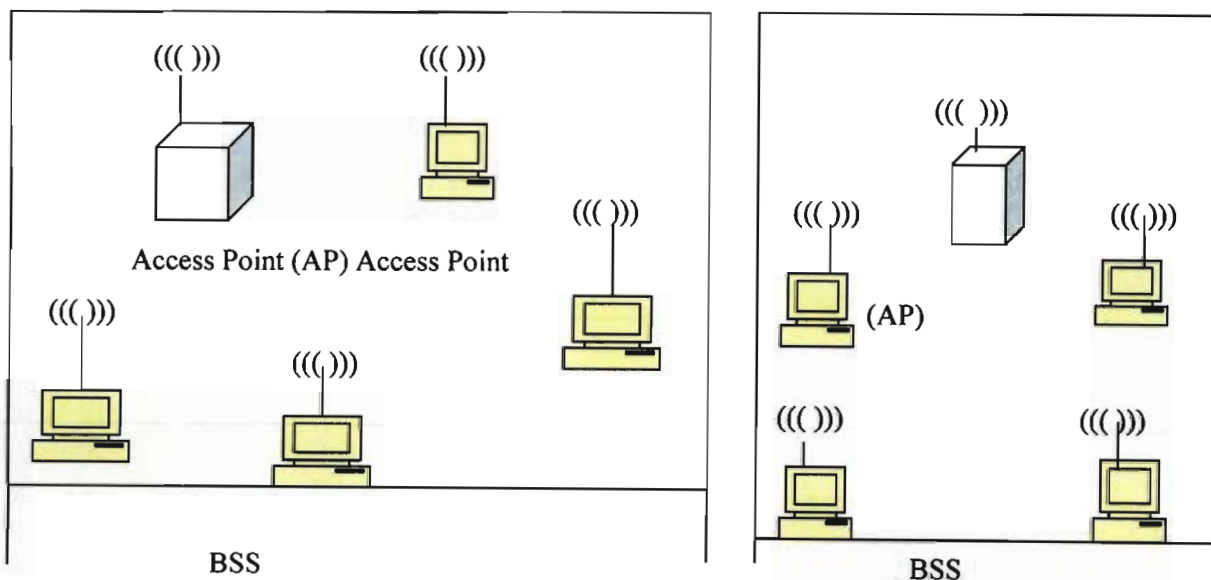


Figure 2.16 the IEEE 802.11 LAN architecture

Among all communications devices operating in license-free 2.4-GHz industrial, scientific, and medical (ISM) bands, wireless LAN adopting the IEEE 802.11 standard have achieved the strongest growth. They provide data rates of up to 54 Mbps that can be shared between the uplink and downlink transmission streams. These high data rates prompted many cellular operations in Europe to consider offering public access wireless LAN in busy areas like airports, hotels, convention centres, cafes, etc [42].

Wireless Local Area Network has been deployed very fast in Africa. Nigeria was one of the African countries interested in using radio in Wide Area Network as well as frequencies in the VHF/UHF bands. Microwave bands have been used in the provision of wireless solutions to data transmission [43]. Wireless data service has taken a full advantage of existing base of data terminals and landline applications, and at the same time providing flexibility for innovative approaches [44].

2.4.3 Wireless LAN in the market

One of the most challenging issues facing the WLAN industry is expanding the market. Those involved in the traditional WLAN industry promote privately owned WLAN applications, such as campus area networks, a market for PCMCIA cards and access points in large quantities. The more visionary service providers are eager to promote nomadic WWLA applications in public places such as airports to generate a new source of income through service charges. Today only WLAN products exist in the market; WATM services are expected to only appear in the market by the turn of the century. In the past, the WLAN industry had a difficult time predicting the development of the market. In 1990 the first generation WLAN products appeared on the market. Early market predictions for WLANs were around \$0.5-2 billion for mid-1990s. However, by the time WLAN products appeared on the market, the less troublesome twisted-pair wiring technology, similar to existing telephone wiring, had already replaced coax LAN technology, so the first-generation WLANs did not meet market predictions [39]. The second-generation WLAN industry evolved in two directions. One group developed PCMCIA card WLANs for laptops to address the need for local mobility and its related applications. The second group added directional antennas to the first generation shoebox-type WLAN products and marketed them as inter-LAN bridges for outdoor applications. The existing WLAN products available on PCMCIA cards are either direct-sequence spread spectrum (DSSS) or frequency hopping spread spectrum (FHSS) operating in ISM bands. Diffused IR (infrared) technology is used for nomadic access in shorter distance for applications such as access for laptop to printer or in specific areas within

hospitals, such as radiology departments, where radio signal use is not encouraged [39]. The market currently aims at four categories of applications [45]: healthcare, factory floors, banking, and education institutions. In the healthcare market, in addition to traditional equipment such as laptops, notebooks, and handheld terminals, special wireless service such as electronic thermometer and blood pressure monitoring devices are expected to be involved in wireless local communication. These devices are used to provide mobile access to clinical and pharmaceutical databases for the physician as well as to enter personal health data. In manufacturing floors and a factory environment, in addition to accessing databases and updating them, wireless networks enable rapid modification of assembly lines and provide instant network access. In the banking industry, WLANs facilitate extending network facilitates to other branches, upgrading systems without disrupting banking operation, reorganizing and rearranging branches, and access to the Internet. In the educational environment, WLANs facilitate distance learning using wireless classrooms, and provides access to the Internet, computational facilitates, and database servers to students using notebook computers. Wireless has the advantage that such a peripheral network can be set up in a few months time [46].

2.5 Transmission Control Protocol

The TCP/IP standard transport level protocol that provides the reliable, full duplex, stream service on which many application protocols depend [47]. TCP is used to provide all reliability needed to communicate across an Internet. TCP also computes a checksum over the data to guarantee that there is no corrupted data during transmission. TCP also uses sequence numbers to ensure that the data arrives in order, and eliminates any duplicate of packet automatically [48]. TCP is the most accepted reliable, connection-oriented, full duplex, byte stream transport protocol that is in use today. It is an end-to-end protocol in wired networks that supports flow and congestion control.

2.5.1 Internet Protocol (IP)

IP is a connectionless protocol, which defines the basic unit of data transfer used throughout a TCP/IP Internet work [49]. IP also is designed to allow multiple routes between any two points, and to route packets of data round damaged routers.

IP is a central switching point in the protocol software. It accepts arriving datagrams from the network interface software as well as outgoing datagrams higher-level protocols generated. After routing a datagram, IP sends it either to one of the network interfaces or to a higher-level protocol

on the local machine [48]. IP is most likely the world's single most popular network protocol. Data moves over an IP-based network in the form of packets; each IP packet contains both a header (that specifies source, destination, and other information about the data) and the message data itself. IP supports the notion of unique addressing for computers on a network. Current IP (IPv4) addresses contain four bytes (32 bits) that are sufficient to address most computers on the Internet. IP supports protocol layering as defined in the OSI reference model. Popular higher-level protocols like Hyper Text Transfer Protocol, TCP, and User Datagram Protocol are built directly on top of IP. Likewise, IP can travel over several different lower-level data link interfaces like Ethernet and ATM. IP originated with UNIX networking in the 1970s [49].

2.5.2 Internet Protocol (IPv4)

Version 4 of Internet Protocol provides the essential communication mechanism of the TCP/IP collection and the global Internet; the long life of version 4 shows that the design is flexible and great. Since the time IPv4 was designed, processor performance has increased over two orders of magnitude, typical memory sizes have increased by a factor of 32, network bandwidth of the Internet backbone has risen by a factor of 800, LAN technologies emerged, and the number of hosts on the Internet has increased from a handful to millions [47].

2.5.3 Internet Protocol (IPv6)

The Internet Engineering Task Force (IETF) has developed a collection of protocols and principles known as IP version 6 (IPv6). This new version, formerly called IP The Next Generation (IPng), embodies the concepts of many-planned techniques for updating the IPv4 protocol. The design of IPv6 is deliberately targeted for minimal impact on upper and lower layer protocols by avoiding the random addition of new features [20].

2.5.4 IP Multicast

IP multicast is the Internet concept of hardware multicasting. It allows transmission of an IP datagram to a set of hosts that form a single multicast group. IP multicast uses the best-effort delivery semantics as other IP datagram delivery, meaning that a multicast datagram can be lost, delayed, duplicated, or delivered out of order [47]. Membership in an IP multicast group is dynamic. A host may join or leave a group at any time. Membership in a group determines

whether the host will receive a datagram sent to the multicast group: a host may send a datagram to a multicast group. Each multicast group has unique multicast (class D) address.

2.5.5 Wireless application protocol

Wireless application protocol (WAP) is an application environment and set of communication protocols for wireless devices designed to enable manufacturer, vendor, and technology independent access to the Internet and advanced telephony services [50].

2.5.5.1 Brief history of Wireless Application Protocol

The history of WAP started on June 26, 1997, when three-industry heavyweight companies Ericsson, Motorola, and Nokia announced the creation of new technology for delivering Internet content to all types of mobile and wireless devices. In December 1997, the companies announced the formation of the WAP Forum Ltd. The purpose of the creation of WAP was to share information and to create an open standard. Each of the companies independently recognizes the coming up convergence of voice and data communications. In less than three years WAP evolved from an idea in the minds of three companies to a worldwide industry standard, currently being implemented [50].

2.5.5.2 Wireless Application Protocol Infrastructure

WAP requires two key pieces of software WAP browser for viewing content on smart phone and handheld devices, and WAP gateways. WAP gateways sit between the mobile device and Internet web server converting WAP requests to HTML requests and translating the resulting response into a compressed binary form for delivering to the mobile device [50].

2.5.6 Wireless Transaction protocol

WTP provide transaction service to WAP. It handles acknowledgment, it also provides retransmission of transactions in case they are not successfully received and removes duplicate transactions. WTP manages different classes of transaction for WAP devices: unreliable one-way request, reliable one-way request, and reliable two-request [50].

2.5.7 Wireless Transport Layer Security

Wireless Transport Layer Security provides service to protect the data, including data integrity, privacy, authentication, and denial-of-service protection. Data integrity guarantees that the data, which is sent, is the same data that is received. WAP privacy services confirm the guarantee during transaction between the WAP device and gateway. Authentication guarantees the authenticity of the client and application server. Denial-of-service detects and rejects data that comes in the form of unverified requests [50].

2.6 Network Performance

Network performance is a complex issue, with lots of independent variables that affect how clients access servers across a network. Most network performance tools use a combination of four separate elements to measure network performance [51]: availability, response time, network utilization, and network throughput.

2.6.1 Availability

The first step in measuring network performance is to determine if the network is even working. If traffic cannot traverse the network, you have bigger problems than just network performance issues. The simplest test for network availability is the ping program. By attempting to ping remote servers from a client device on the network, you can easily determine the state of your network. The ping program sends an Internet Control Message Protocol echo request packet to the destination host. When the echo asks it the packet has been received, the remote host immediately returns an echo reply packet to the sending device. Receiving an echo reply packet from the remote host means that there is an available network path between the client and server devices. If no echo reply packet is received, there is a problem with either a network device or a link along the path [52]. The format of the ping command is:

```
ping [-dfnrvR] [-c count] [-i wait] [-l preload] [-p pattern] [-s packetsize]

C:\Documents and Settings\osmank>ping 146.230.92.78
Pinging 146.230.92.78 with 32 bytes of data:
Reply from 146.230.92.78: bytes=32 time<1ms TTL=128
Reply from 146.230.92.78: bytes=32 time<1ms TTL=128
Reply from 146.230.92.78: bytes=32 time<1ms TTL=128
Reply from 146.230.92.78: bytes=32 time<1ms TTL=128

Ping statistics for 146.230.92.78:
    Packets: Sent = 4, Received = 4, Lost = 0 (0% loss),
    Approximate round trip times in milli seconds:
        Minimum = 0ms, Maximum = 0ms, Average = 0ms
```

Figure 2.17 describing available network path between the client and server

2.6.2 Response Time

Response time affects how quickly network applications appear to be working. Slow response times are often magnified by network applications that need to send and receive lots of information across the network, or applications that produce immediate results from the customer entry [54]. In large networks there are many factors that can affect the response times between a client and server. These factors can include:

- ❑ Overloaded network segments
- ❑ Network error
- ❑ Faulty network wiring
- ❑ Broadcast storms
- ❑ Faulty network devices
- ❑ Overloaded network hosts.

Any one or a combination of those factors can contribute to slow network response time. For internal LAN connections, the response time should be well within 1 or 2 milliseconds. For WAN connections, the response times can often be over 200 or 300 milliseconds, depending on connectivity speeds.

2.6.3 Network Utilization

The major factor in network performance is the utilization of each network segment along the path between two endpoints. The network utilization represents the percent of time that the network is in use over a given period. Ethernet segment is either at 100 percent utilization (carrying a packet), or at 0 percent utilization (idle). To determine the total number of bits received on the interfaces, each of the packet byte rates is multiplied by 8. This value is divided by the total interface capacity multiplied by the time interval of the sample (in seconds) [53].

$$Utilization = \frac{((datasent + datarev) * 8)}{(int\ speed * sampletime)} * 100$$

For example, a 10Mb half duplex network interface that sends 800,000 bytes of data and receives 250, 000 bytes over a 5 second period would have a network utilization of:

$$Utilization = \frac{((800000+250000)*8)}{(1000000*5)} * 100$$

$$= 5.6\%$$

The 5.6 percent utilization represents the network utilization only for that 5 second period.

2.6.4 Network throughput

The throughput of a network represents the amount of network bandwidth available for the network application at any given moment, across the network links [55]. Determining network throughput allows the network administration to find network bottlenecks that slow down performance over a given network link between clients and servers. The hardest part of determining the network throughput is calculating the effect that the intermediate link has on the overall end-to-end network connection. Calculating network throughput is a mathematical process, it involves sending periodic streams of packets, and determining the rate at which the server receives the streams. Each stream sample produces data elements used to determine the amount of bandwidth remaining on the network link. The streams are increased until the maximum bandwidth is observed, then quickly backed off so as not to affect the network performance [55]. It is best to calculate network throughput at different times of the day, and on different days of the week. This enables you to gather all of the information on different applications as they are run on the network [55].

2.7 previous works

The use of switch in LAN is an effective technique to increase the throughput of the network. Elsaadany, Singhal, and Lui [1] studied how buffering at the switch affects the performance of the network, in particular, how the buffer size at the ports of the switch affects the packet loss rate and the percentage of delayed packets. To examine the effects of packet loss and delay, they modelled the switch with different buffer sizes and compared their performance merits. The result obtained showed a substantial improvement in packet loss rate and the packet delay. They found the packet delay with a buffer size of 100 recorded 7.78 ms and 0.05% packet loss. With a buffer size of 50 the packet delay was recorded 17.06 ms and 1.06 % packet loss [1].

Jain [56] analyzed the performance of Fiber Distributed Data Interface using a simple analytical model and simulation model. The performance metrics of response time, efficiency, and maximum access delay were considered. The efficiency was defined as the ratio of the maximum obtainable throughput to the nominal bandwidth of the network and the delay was defined as the time it takes to receive usable token. In addition, the performance was affected by a parameter called the Target Token Rotation Time (TTRT), length of the cable, total number of stations, and frame size. The results showed poor efficiency (60%) in 4 ms and large access delay with 165 ms of TTRT. 8 ms value of TTRT was most desirable, it gave 80% efficiency and 1 second maximum delay. The cable length showed lower efficiency and longer access delay. Frame size did not appear in the simple model of efficiency and access delay, because frame size had little impact on FDDI performance. The response time that was obtained by using the simulation model was not significantly affected by TTRT values [56].

Kabir, Abdullah, Khatun, Anas, and Mahdi [57] made a study to increase throughput and reduce average packet delay of CSMA/CD based LAN by reducing collision, and eliminated the packet loss. The study showed performance enhancement of CSMA/CD based on Fast Ethernet single channel optical LAN in terms of throughput, average packet delay and percentage of collision. Here the collision rate was minimized to less than 3% and packet loss eliminated. The average packet delay was also decreased to 1.01 ms [57].

Nahum¹, Rosu¹, Seshan², and Almeida³ [58] examined the effects of Wide-Area condition on World Wide Web (WWW) server performance. They used an experimental testbed which emulates WAN characteristics in a live setting by introducing factors such as delay and packet loss. The purpose was to examine how these factors interacted with the host TCP implementation and what influence they had on the web server performance. The testbed consisted of a cluster of personal computers (PCs) acting as clients, connected via switched LAN to a higher performance web server. The study found that the packet delay and packet loss could have substantial effects on server throughput [57]. There was also another approach used to examine the 1994 California election web site [59] and 1996 summer Olympics web site [60]. The 1994 study identified a performance problem with protocol control block (PCB) lookups and the handling of TCP connections in the wait time state. The suggested fix for these problem was quickly incorporated into operating systems. The 1996 study concentrated on TCP behaviour and made suggestions to

improve the loss recovery. This analysis identified performance problems that were difficult to find with current benchmarks.

Kashorda [61] examined the three main telecommunication options that could be used to interconnect universities in Kenya and for access to the global Internet, namely: digital leased-line local access network, VAST-based network, and terrestrial digital radio network. The cost analysis suggested that digital leased-line and the digital radio networks are the two viable options for creating a national education network. The analysis proved that the private digital radio network has the key advantage of providing high-speed links between the universities [61].

Golmie, Cypher, and Rebala [62] studied performance analysis of low rate wireless technologies for medical application. They evaluated the effect of packet segmentation and bakeoff parameter tuning to improve the overall network performance that is measured in terms of packet loss, throughput, and packet delay. The study found 18% of packet loss and 0.0392 second of delay [62].

Khattab [63] proposed Hybrid Packet Reservation Multiple Accesses (PRMA)-CSMA/CD protocol to study the performance of interconnection between wireless and wired LANs, using a queuing model to investigate the wireless/wired LAN via packet delay and throughput factors. The study showed a good result to maximize the system performance while assuring stability and the parameters can be calculated adaptively during the operation to assure best performance under different load conditions [63].

Gardner [64] studied the performance of VoIP under speech coder G723.1, G729A, and G711, packet delay and packet loss was the key factor to study the performance. R-value was also a main issue to measure the quality of service. The author used two capacity links (256 kbps and 1,544 Mbps) to examine the performance of VoIP. In 256 kbps the optimum coder was G729A with 4.3 call and good quality of service (R-value = 76). The packet delay in the coder was 199 ms and 0.005% packet loss. G723.1 was selected as the optimum coder with 1.544 Mbps capacity link, where recorded under this coder 37.1 calls, 122 ms packet delay, and 0.005 % of packet loss. R-value recorded the same result 76.

Kos, Vugrinice, and Tomazic [11] investigated the possibilities of real time voice transmission over Local Area Networks. The network of choice was 10 Mbit/s Ethernet and the voice coder of choice was G723.1. Delay was identified as the key parameter to real time voice transmission. They measured the total delay between VoIP terminals in an unpreloaded network. The measurement showed the following results:

- 95 ms delay in the Personal Computer (PC) telephone (transmission). This delay comprises hardware delay (sound card and network card) and voice processing on the computer (Netmeeting software and TCP/IP stack of windows operating system);
- 396 ms delay in the Personal Computer (PC) telephone (reception). This delay comprises the hardware delay and jitter buffer delay;
- 1 ms delay on the physical transmission med.

2.8 Summary

In this chapter a review of the technologies used in LAN, WAN, and WLAN has been given. The different topologies in LAN and WAN, the different type of LANs such as (Ethernet, Token Ring, and Fiber Distributed Data Interface), different devices of LAN connection to link more than LAN together, and the different types of cable were discussed. The different type of capacity links such as (ISDN, DSL, ATM, X.25, and Frame Relay) were also discussed. The WLAN technology through (Infrared, Spread Spectrum, and Narrowband microwave), standard technology of WLAN, and WLAN in the market were also discussed. Network protocols such as (TCP, IP, WAP, WTP, and WTLS) and network performance were also illustrated.

Chapter Three

Methods and Tools

3.1 Introduction

This chapter describes the tools and methodology used for analyzing the performance of WAN between Asmara and Massawa. The simulation models examine the performance of LAN, WAN, and WLAN based on the following parameters: network traffic throughput, packet loss, collision, and queue delay. The goal of the simulation study is to characterize the system performance and allow basic changes in the way system elements (components) are configured; additional components may be added and/or system values (parameters) may be subjected to change. All the simulations for network performance characterization have been performed using the discrete event simulator OPNET Modeler version 11.0 [68]. OPNET is commonly used for network simulation and provides a rich library of models for implementing wired and wireless simulation scenarios. This chapter begins with a brief discussion of the tools and simulation methodologies used in OPNET and then discusses other methods with Visual Basic and JavaScript programs.

3.2 Testing the existing traffic of WAN

To examine and compare the performance of the typical WAN traffic (e.g. through Asmara LAN and Massawa LAN) via single path capacity link (19200 bps) and dual path capacity link (9600 bps) we use the following tools and method:

3.2.1 Data collection

The data used for analysis are captured in August and September 2005 between the two LANs. Table 3.1 and Table 3.2 describes the sample of the data from August and September 2005 between the LANs.

Table 3.1 Sample of Data from August 2005 between Asmara LAN and Massawa LAN for single path

Time(hour)	Sent Packet (Packet/hour)	Received Packet(Packet/hour)	Packet Size (bytes)
8	1483	1469	1500
9	1956	1943	1500
10	2224	2219	1500
11	1633	1629	1500
12	850	850	1500
13	0	0	1500
14	634	633	1500
15	1227	1216	1500
16	2653	2644	1500
17	1742	1740	1500
18	475	475	1500
8	549	549	1500
9	1639	1623	1500
10	1546	1541	1500
11	2856	2837	1500
12	1126	1126	1500
13	0	0	1500
14	923	923	1500
15	1849	1842	1500
16	1535	1524	1500
17	2064	2057	1500
18	113	113	1500
8	1861	1848	1500
9	2943	2937	1500
10	2376	2368	1500
11	1345	1345	1500
12	728	728	1500
13	0	0	1500
14	1265	1254	1500
15	1946	1932	1500
16	2530	2524	1500
17	1757	1741	1500
18	523	521	1500

Table 3.2 Sample of Data from September 2005 between Asmara LAN and Massawa LAN for dual path

Time (hour)	Sent Packet (Packet/hour)	Received Packet (Packet/hour)	Packet Size
8	667	667	1500
9	829	828	1500
10	1000	997	1500
11	753	753	1500
12	543	543	1500
13	0	0	1500
14	381	381	1500
15	972	970	1500
16	1231	1227	1500
17	847	845	1500
18	464	464	1500
8	720	720	1500
9	996	994	1500
10	1107	1106	1500
11	1248	1243	1500
12	839	839	1500
13	0	0	1500
14	427	427	1500
15	659	659	1500
16	914	912	1500
17	588	588	1500
18	324	324	1500
8	631	628	1500
9	1044	1041	1500
10	1325	1323	1500
11	951	947	1500
12	726	722	1500
13	0	0	1500
14	357	357	1500
15	549	549	1500
16	833	830	1500
17	1275	1273	1500
18	789	787	1500

3.2.2 Softwares

For simulation and calculation Microsoft Visual Basic and Java script are used. The scripts of both languages are described in detail in Appendix A.

3.2.3 Method

To study the performance of WAN traffic between Asmara LAN and Massawa LAN, M/M/1 single path and M/M/2 for dual path queuing models were used. The key factors of the analysis are:

- Channel utilization (ρ)
- Throughput
- Waiting time in the queue
- Packet loss

Queuing models are usually classified in terms of the service times, the number of servers, the serving process, and the way in which arrivals and servicing occur over time. The simplest of the models is single-line, single server. The further classification of the queue identifies the timing of arrivals for service and their processing. The most fundamental characteristics of the queuing model problems are the average arrival rate and the average service rate [65] [66] [67].

Define: (definition)

λ = Average arrival rate (number of units arriving for service per unit of time)

μ = Average service rate (number of units being serviced per unit of time)

From Little's theorem, we have:

$$N = \lambda \cdot T$$

$$N_q = \lambda \cdot W$$

Where:

N = Average number of packet in the system.

T = Average packet time in the system.

N_q = Average number of packets waiting in the queue.

W = Average packet waiting time in the queue.

The model is described in detail in Appendix B.

To compute the factors that are mentioned in section 3.2.3, the average of the arrival packet (λ) and the average of service rate (μ) are extracted from the traffic data measurement shown in Table 3.1 and Table 3.2 through the single path capacity link (19200 bps) and dual path capacity link (9600 bps) , using the following formula [66]:

$$\lambda = \frac{\text{Total number of packet arriving}}{60 \times 60}$$

For example for the first entry hour on 1 August 2005, we have:

$$\lambda = \frac{1469}{60 \times 60} = 0.408 \text{ packet / second}$$

and

$$\mu = \frac{\text{Capacity link}}{\text{Total number packet size (byte)} \times 8 \text{ bits}}$$

$$\mu = \frac{19200}{1500 \times 8} = 1.6 \text{ packet / second}$$

For dual path capacity link (9600bps), we use the same formula as used for single path

$$\lambda = \frac{\text{Total number of packet arriving}}{60 \times 60}$$

For example for the first entry hour on 1 September 2005, we have:

$$\lambda = \frac{667}{60 \times 60} = 0.185 \text{ packet / second}$$

and

$$\mu = \frac{\text{Capacity link}}{\text{Total number packet size (byte)} \times 8 \text{ bits}}$$

Example

$$\mu = \frac{9600}{1500 \times 8} = 0.8 \text{ packet / second}$$

To analyze the factors that are mentioned in section 3.2.3, a JavaScript program was developed to locate all the analytical formula that are described in detail in Appendix B in one function to compute the factors. The results and the analysis of the factors are described in chapter 4. The program script is given in Appendix C.

3.3 Simulating the Performance of LAN and WLAN

Here we examine the efficiency and effectiveness of LANs and WLANs traffic by studying the following parameters: network traffic throughput (via File Transfer Protocol, Email system, and Database system), packet loss, collision, and queue delay. The study used the following tools and methods.

3.3.1 Simulation Tools: The OPNET Modeler

To study the performance of LAN, WAN, and WLAN through the above parameters, OPNET simulation modeler was used. Different types of connection devices are considered namely: hubs, switches, routers, fiber optic, and 10 Base-T cables.

OPNET Modeler is among the leading discrete event network simulation models used both by commercial and research communities. It provides a comprehensive framework for modeling wired as well as wireless network scenarios [68].

Simulation models are organized in a hierarchy consisting of three main levels: the simulation network, node models and process models. The top level refers to the simulation scenario or simulation network. It defines the network layout, the nodes and the configuration of attributes of the nodes comprising the scenario. The node models are at the second level in the hierarchy and consist of an organized set of modules describing the various functions of the node. The modules in the nodes are implemented using process models, the lowest level in hierarchy. Process models consist of finite state machines, definitions of model functions, and process interface that defines the parameters for interfacing with other process models and configuring attributes. Finite state machine models are implemented using Proto C, which is a discrete event library based on C functions. The hierarchal structure of the models, coupled with support for C language programming allows easy development of communication or networking models [68].

OPNET Modeler also uses an object-oriented approach for the development of models and simulation scenarios. The models can be identified as a CLASS, which can be reused any number of times in the simulation by creating its different instantiations, just like the creation of objects in any object-oriented programming language. OPNET allows the user to extend the functionality of the basic models already available as part of the model library. Thus, by defining the value of attributes of the basic model the user can develop customized models following particular standards or vender specifications [68].

3.3.2 Simulation Methods

To simulate the parameters described above, different scenarios with different devices and configurations were created. To study the LAN performance, three different scenarios were created, namely: scenario 1 used a hub connection; scenario 2 used a switch connection; and scenario 3 used a router connection. The simulation details for the LAN are described in sections 3.3.2.1, 3.3.2.2, and 3.3.2.3 respectively. The performance results of each scenario are described in chapter 4.

3.3.2.1 LAN with hub connection (scenario one)

In this scenario, the LAN is connected with 6 workstations and three arbitrary servers – in this case, an FTP Server, Dbase Server, and Email Server – through a hub connection. The area used for LAN is 100×100 square meters, wherein the star topology is employed with the hub connection. 10-Base-T Cables were used to connect the workstations through the hub. Figure 3.1 describes the simulation model of LAN with hub connection.

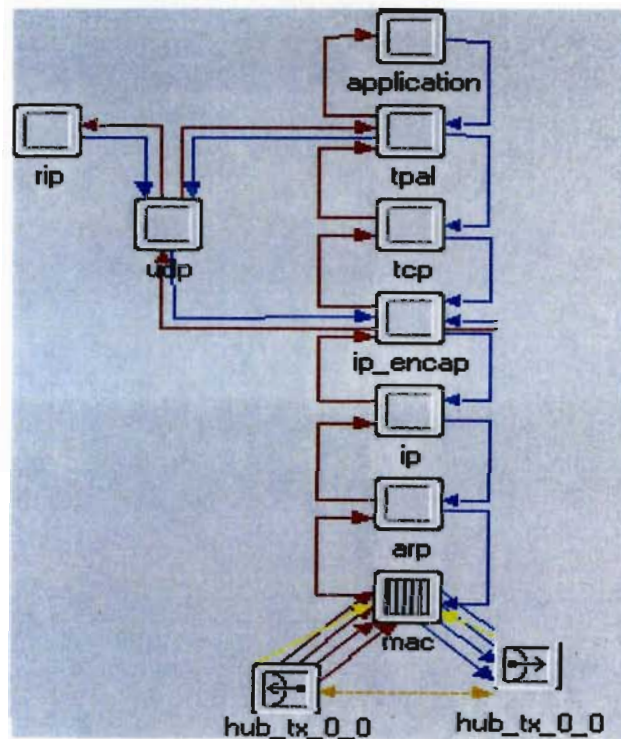


Figure 3.1 LAN Model with hub connection

3.3.2.2 LAN with Switch connection (scenario two)

In this scenario the same method that is described in section 3.3.2.1 was used except for changing the hub connection to switch connection. Figure 3.2 describes the simulation model with switch connection.

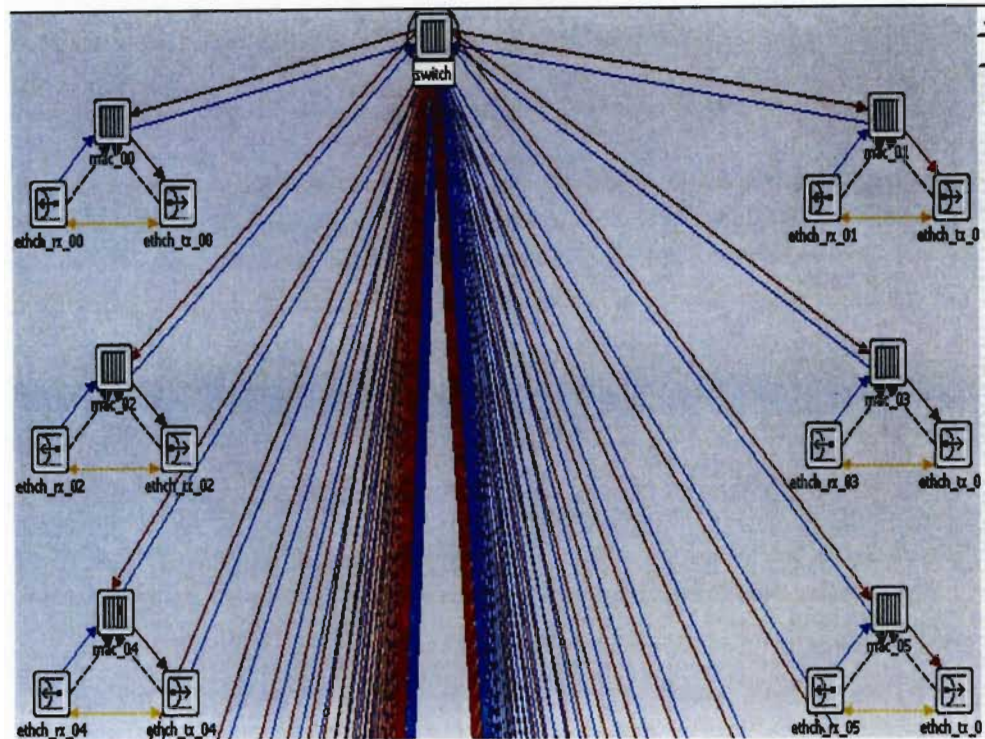


Figure 3.2 LAN Model with Switch connection

3.3.2.3 LAN with router connection (scenario three)

In this scenario five LANs were linked to the Eritrean Information System Agency (EISA), which is the government research centre, via fibre optic cable, to share the important resource such as Email service and file transfer. EISA is considering a centre or a gate for the ministries to exchange information between one another. The distance between EISA and each ministry is 500-600 meters. In the simulation scenario each LAN contained three workstations, one for Email user, the second workstation for FTP user, and the third workstation for Telnet user. The three workstations were connected through a switch and router. EISA contained an Email server, an FTP server, Telnet server, and a default gateway router connection through a switch. The media connection of the LANs is 10Base-T with 10 Mbps of capacity. The fiber optic capacity exceeded 100 Mbps. Figure 3.3 describes the simulation model with router connection

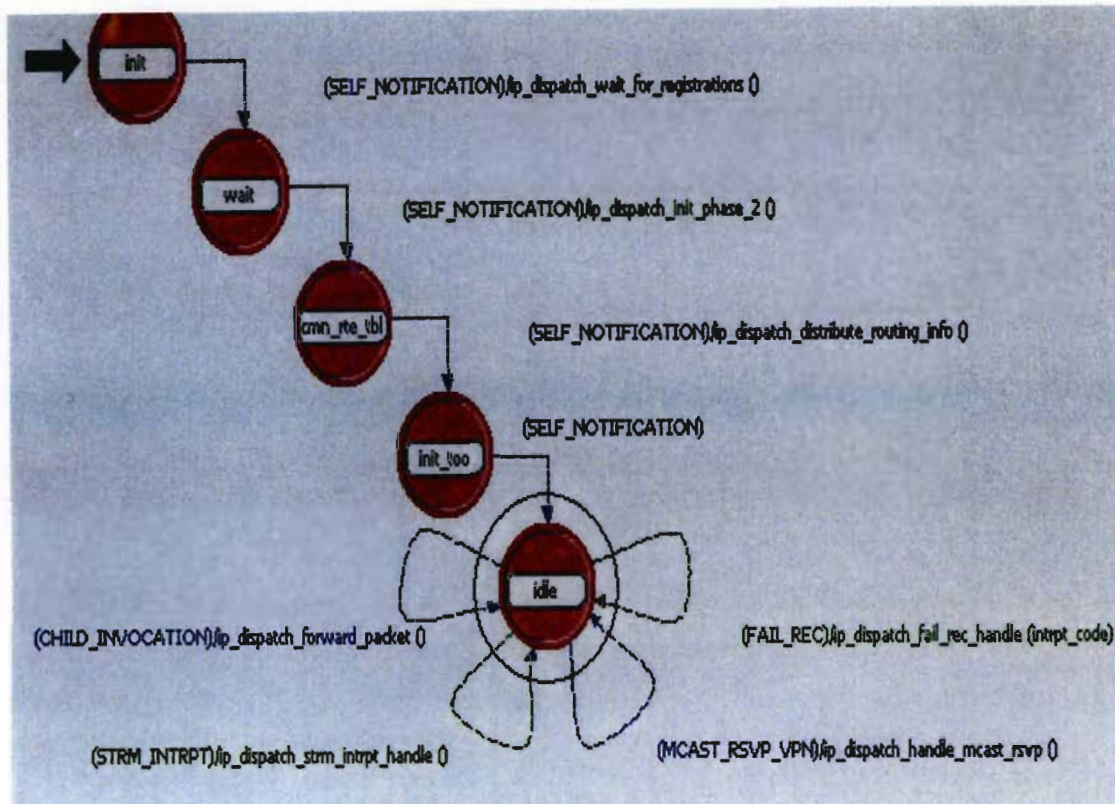


Figure 3.3 Different LANs with Router connection

To study the WLAN performance two scenarios were run; the first scenario (scenario four) used antennae to communicate directly between two stations and the second scenario (scenario five) used an integration between wireline and WLAN network. The simulation details are described in sections 3.3.2.4 and 3.3.2.5. The performance results of each scenario are described in chapter 4.

3.3.2.4 WLAN peer-peer network or directly communicate (scenario four)

This scenario had two wireless LAN-based workstations in a simple network configuration (independent BSS). This was an independent or peer-peer network where stations would directly communicate with each other. This kind of data communication is very important for the country – it can be utilized for military communication in different locations, agriculture and geology or to exchange information between workstations during field survey engineering. Figure 3.4 describes the simulation model of WLAN; peer-to-peer communicate between two independent stations.

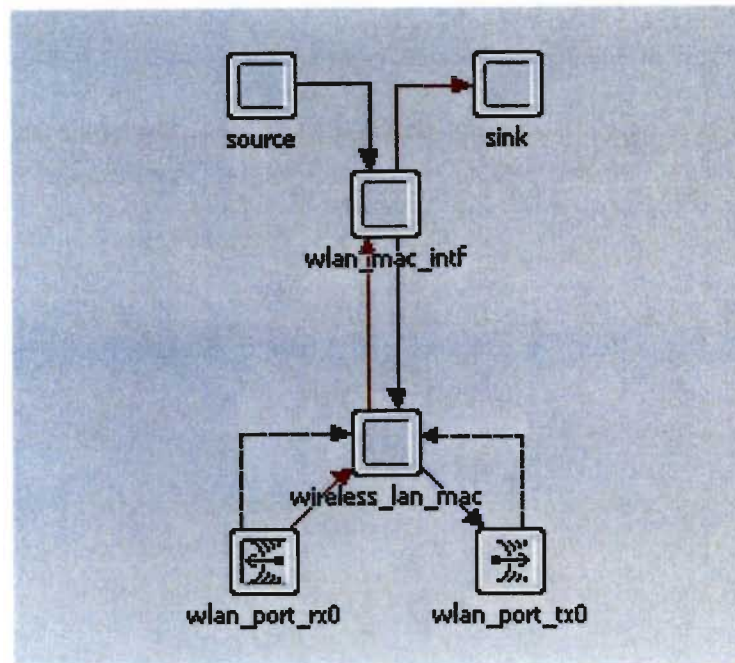


Figure 3.4 WLAN simulation model of two base stations

3.3.2.5 Integration between wireline and WLAN networks (scenario five)

This scenario consisted of wireless and wireline networks. The purpose of the scenario was to show the inter-communication between wireless and wireline networks through the Internet backbone. Site One (Site1) and Site Two (Site2) subnets contained 10 wireless stations; all stations complied with the wireless LAN (802.11) protocol. The access point nodes in Site1 and Site2 connected each subnet to the wireline network. The clients in the wireless LAN communicated with servers at remote sites via IP Cloud. This type of communication is very important in areas such as Massawa, Decmhare and Asmara to solve the wiring problems there. Figure 3.5 describes the integrated simulation model between wireline and WLAN.

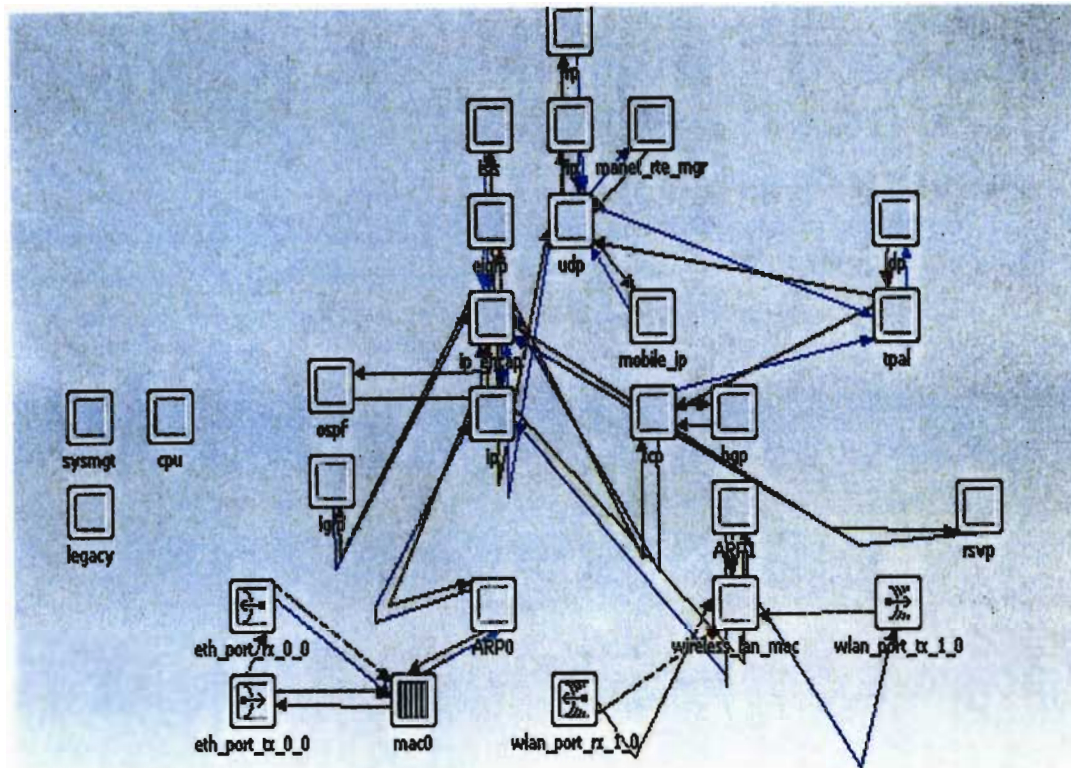


Figure 3.5 Wireline and WLAN simulation model

3.4 Performance of VoIP in Eritrean WAN Environment

We examine the performance of VoIP on the current Eritrean networks through packet loss and delay parameters.

To study the performance of VoIP through packet loss and delay parameters the OPNET Modeler simulation program was used as described in section 3.3.1, the simulation study was done in two areas:

Between Asmara LAN and Massawa LAN using different capacity links: 28 kbps, 33 kbps, 64 kbps, and 128 kbps and the total traffic of WAN between Asmara LANs and Massawa LANs with 256 kbps. A sample of the simulation is described in section 3.4.2

3.4.1 R-value

E-Model defines the “R” value as the measure of voice quality. The International Telecommunication Union (ITU) and the Telecommunication Industries Association (TIA) have both approved the E-Model for use [69] [71].

R is the general parameter that is used to define voice quality. Table 3.3 shows a comparison of R values and Mean Opinion Score (MOS) levels. R is represented in [70] [71] [72] as the following:

Table 3.3 E-Model Values

User Satisfaction	E-Model-R	MOS
Very Satisfied	90	4.3
Satisfied	80	4
Some User Dissatisfied	70	3.6
Many Users Dissatisfied	60	3.1
Nearly All Users Dissatisfied	50	2.6
Not Recommended	0	1

$$R = R_o - I_s - I_d + I_e + A \quad (3.1)$$

Where:

R_o represents the signal to noise ratio at 0 dB point. It should comprise the noise in the network and in the environment at the speaker and listener sides.

I_s is the parameter for simultaneous impairments like excessive loudness ratings, uncomfortable side-tone levels and simple quantisation distortion.

I_d contains impairments due to echoes and delay. For example, talker and listener echoes and excessive delays.

I_e represents the impairments from transmission equipment. Impacts from these impairments are difficult to measure theoretically. Therefore, values for codec's and other frequently used equipment have been measured and printed in recommendation by ITU, and this is from where we assume the value for calculation in this study.

A is a factor that adjusts the quality value in certain situations by considering non technical aspects in the system.

The simultaneous impairments I_s equations are [69]:

$$I_s = I_o l_r + I_{st} + I_q \quad (3.2)$$

Where:

$I_o l_r$ and I_q are impairments that are related to the low values of overall gain and impairments.

I_{st} is the impairment due to non-optimum sidetone.

$$I_{st} = 10 \left[1 + \left(\frac{STMR_o - 12}{5} \right)^6 \right]^{\frac{1}{6}} - 46 \left[1 + \left(\frac{STMR_o}{23} \right)^{10} \right]^{\frac{1}{10}} + 36 \quad (3.3)$$

$$STMR_o = -10 \ln \left[10^{\frac{STMR}{10}} + e^{\frac{T}{4}} 10^{\frac{TEL R}{10}} \right] \quad (3.4)$$

Where:

STMR is the sidetone Masking Rating.

TEL R is the Talker Echo Path Loss.

T is the mean one way delay of the echo path.

Id is defined as (69):

$$Id = Idte + Idle + Idd \quad (3.5)$$

$$Idte = \left[\frac{Roe - Re}{2} + \sqrt{\frac{(Roe - Re)^2}{4} + 100} - 1 \right] (1 - e^r)$$

$$Roe = -1.5(No - RLR) \quad (3.6)$$

$$Re = 80 + 2.5(TErv - 14)$$

$$TErv = TELR - 40 \ln \left[\frac{1 + \frac{T}{10}}{1 + \frac{T}{150}} \right] + 6e^{-0.37^2}$$

Where:

RLR is receiving Loudness Rating.

Idel represents Listener Echo.

$$Idle = \frac{Ro - Rle}{2} + \sqrt{\frac{(Ro - Rle)^2}{4} + 16} \quad (3.7)$$

$$Rle = 10.5(WEPL + 7)(r + 1)^{-0.25}$$

Where **WEPL** is Weighted Echo Path Loss and **Tr** is round trip delay.

$$Idd = 25 \left\{ \left(1 + x^6 \right)^{\frac{1}{6}} - 3 \left(1 + \left[\frac{x}{3} \right]^6 \right)^{\frac{1}{6}} \right\} + 2$$

(3.8)

$$x = \frac{\ln\left(\frac{Ta}{100}\right)}{\ln(2)}$$

We take all the analytical formulas that are mentioned above, and create a simple program through visual basic to calculate R-Value. The source code of the program is listed in Appendix C.

3.4.2 Simulation of VoIP between Asmara LAN and Massawa LAN

The purpose of this scenario was to study the performance of Voice over Internet Protocol (VOIP) with only voice on WAN between Asmara LAN and Massawa LAN through OPNET Modeler simulation program. The distance between Asmara LAN and Massawa LAN is 115 km. The technologies used for this purpose were star topology with switch connection and Gateway Router, 10BaseT for each LAN, 28 kbps, 33 kbps, 64 kbps, and 128 kbps capacity links between Asmara LAN and Massawa LAN and speech coder parameters as shown in Table 3.4. Figure 3.6 described the simulation model. Once the delay parameter value and packet loss parameter value were obtained it was easy to measure the quality of service (QOS) using R-value Model as described in section 3.4.1. The performance study results and R-values are described in chapter 5.

Table 3.4 Voice Coder Compression Parameters

Coding Standard	Bit Rate (kbps)	Frame Length (ms)	Codec Delay (ms)	MOS
G723.1	5.3/6.3	30	37.5	3.9/3.7
G729A	8	10	15	4
G711	64	0.125	0.125	4.2

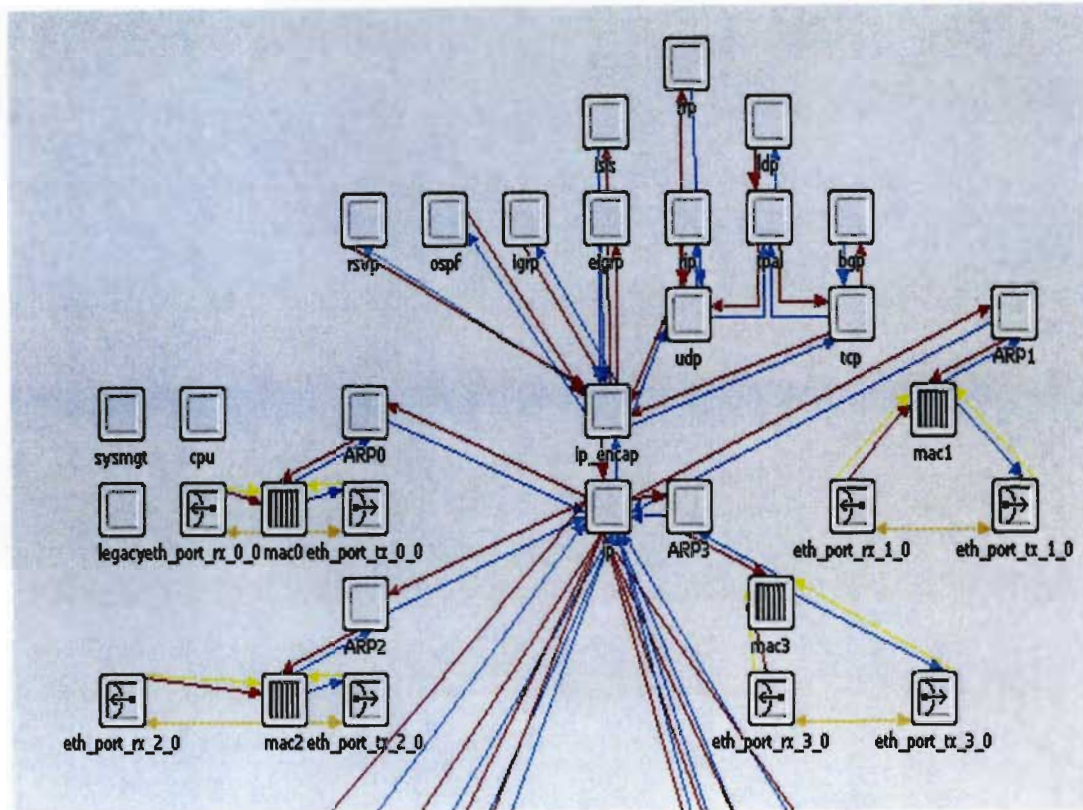


Figure 3.6 Model for VoIP between Asmara LAN and Massawa LAN

3.4.3 Simulation of VoIP between Asmara LANs and Massawa LANs (Total traffic)

The purpose of this scenario was to study the performance of Voice over Internet Protocol (VOIP) with voice and data on WAN between Asmara LANs and Massawa LANs total traffic through OPNET Modeler simulation program. The distance between Asmara LAN and Massawa LAN is 115 km. The technologies used for this purpose were star topology with switch connection and Gateway Router, 10BaseT for each LAN, 256 kbps capacity link between Asmara LANs and Massawa LANs with speech coder parameters as shown in Table 3.4. Figure 3.7 described the simulation model of the campus. Once the delay parameter value and packet loss parameter value were obtained it was easy to measure the quality of service (QOS) using R-value model as described in section 3.4.1. The performance study results and R-values are described in chapter 5.

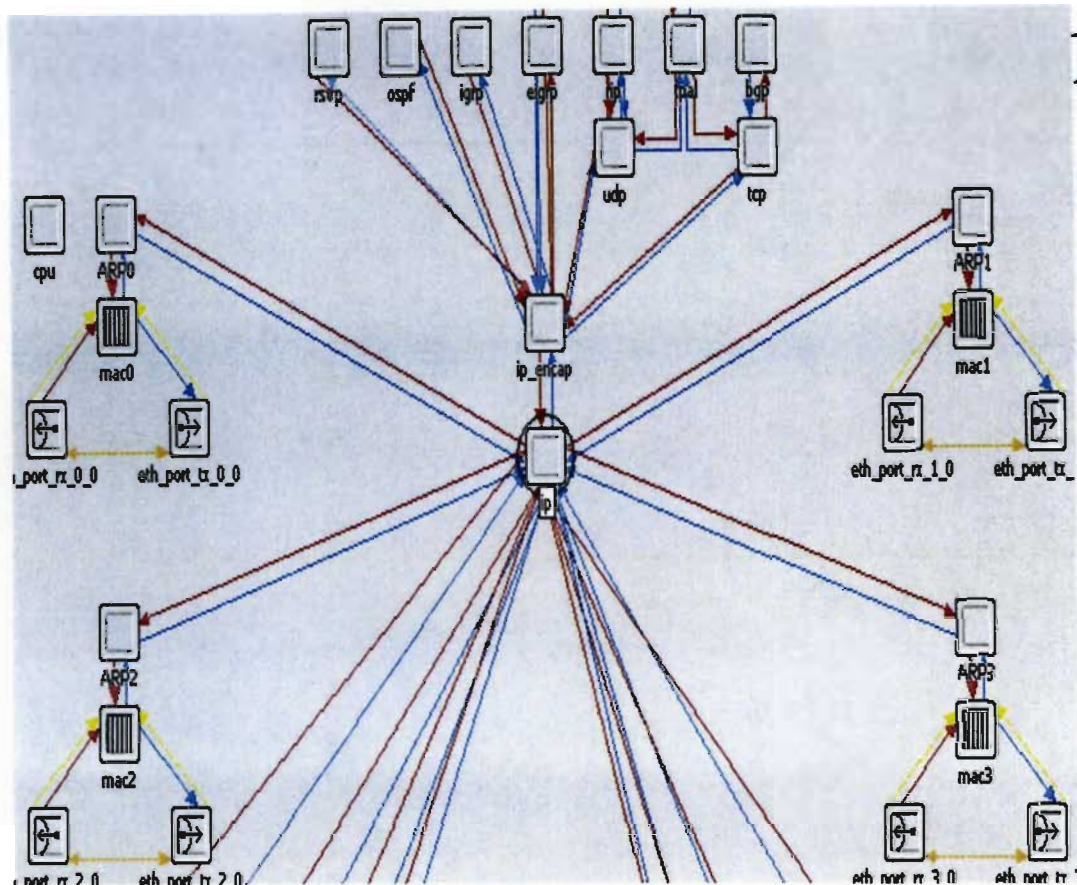


Figure 3.7 Total traffic of WAN simulation model between Asmara LANs and Massawa

3.5 Summary

The chapter discusses the tools and methods used to examine the measured traffic of WAN between Asmara LAN and Massaw LAN using Markovian M/M/1 queue model to determine the queue delay in single link and M/M/2 queue model for dual path to determine the queue delay. JavaScript program was employed to compute the analytical parameters as shown in section 3.2.3.

The chapter also describes OPNET Modeler simulation program and the methods used to examine the performance of LAN and WLAN using different connection devices for LAN scenarios and two scenarios for WLAN peer-to-peer and integration of wireline and WLAN communication.

Finally, the same OPNET Modeler simulation program was employed to examine the performance of VoIP with voice through WAN between Asmara LAN and Massawa LAN using 28 kbps, 33 kbps, 64 kbps, and 128 kbps capacity links with G723.1, G729A, and G711 speech coders. Then the performance of VoIP with voice and data through different cases, was examined using the total traffic of Asmara LANs and Massawa LANs via 256 kbps WAN capacity link with G723.1, G729A, and G711 speech coders.

Chapter Four

Network Performance Results and Analysis

4.1 Introduction

This chapter has two main goals. The first goal is to examine the performance of WAN between Asmara and Massawa LANs. This is achieved from the analysis of single path (19200 bps) and dual path (9600 bps) traffic, the throughput, and packet loss parameters. The traffic analysis results and discussion of single and dual path are described in sections 4.2 and 4.3. The second goal is to select the best connection of the LAN from the simulation analysis results that can support the performance of the network and eliminate all the problems that were mentioned in chapter one. Collision, packet loss, and delay parameters are the key factors to analyze the simulation result. The simulation and the analysis of LAN are described in section 4.5. This chapter also examines the performance of WLAN in old areas (where cables is not possible) through peer-to-peer communication and an integrated wireless and wireline network using the same parameters that are mentioned above to analyze the simulation result. The simulation results and discussion are described in section 4.6.

4.2 Single path traffic analysis (from measurement)

In order to analyze the single link measured traffic for August 2005 we highlight three typical days (1st, 10th and 20th of August 2005) and at the daily busy hour in August 2005, to study the performance of the WAN between Asmara and Massawa. The traffic is measured based on 20 users as the size of LAN in both areas (Asmara and Massawa). From the analysis of the single path capacity we found the average of the service rate (μ) is equal to 1.6 packet/sec. Figure 4.1 and Table 4.1 describe the traffic on different days and the daily busy hour in August 2005. In figure 4.1 the graphs show the busy hour arrival rate (λ) and utilization (ρ) in the busy hour of each day. The busy hour on 1st of August was 4 pm with 0.734 Packet/sec arrival rate (λ) and 0.459 channel utilization (ρ). It can be seen from the same graphs that the busy hour on 10 of August 2005 was 11 am with 0.788 Packet/sec arrival rate (λ) and 0.492 channel utilization (ρ). We also observe from the graph that 11 am was the busy hour on 20 August 2005 with 0.714 Packet/sec arrival rate (λ) and 0.446 channel utilization (ρ). The graphs also show zero traffic in figure 4.1 during lunch break. Table 4.1 shows the typical daily busy hour arrival rate at 11 am on 29 August 2005 with 1.067 Packet/sec arrival rate (λ) and 0.667 channel utilization.

The table also shows 0.693 Packet/sec as the average of the daily busy hour arrival rate (λ) and 0.424 as the average of the daily channel utilization (ρ). 11 am 12 noon were the typical busy hours during August 2005. The analysis found that the channel utilization (ρ) result in the typical busy hour on 29 August 2005 was above 0.6 Packet/sec, this means that the delay and packet loss were very high on that day. In other words, there is a problem in the performance of WAN. The analysis also found that the customers preferred to start the service process 11-12 am and 4 pm. The throughput, packet loss, and delay parameters will be discussed in sections 4.2.1, 4.2.2, and 4.2.3.

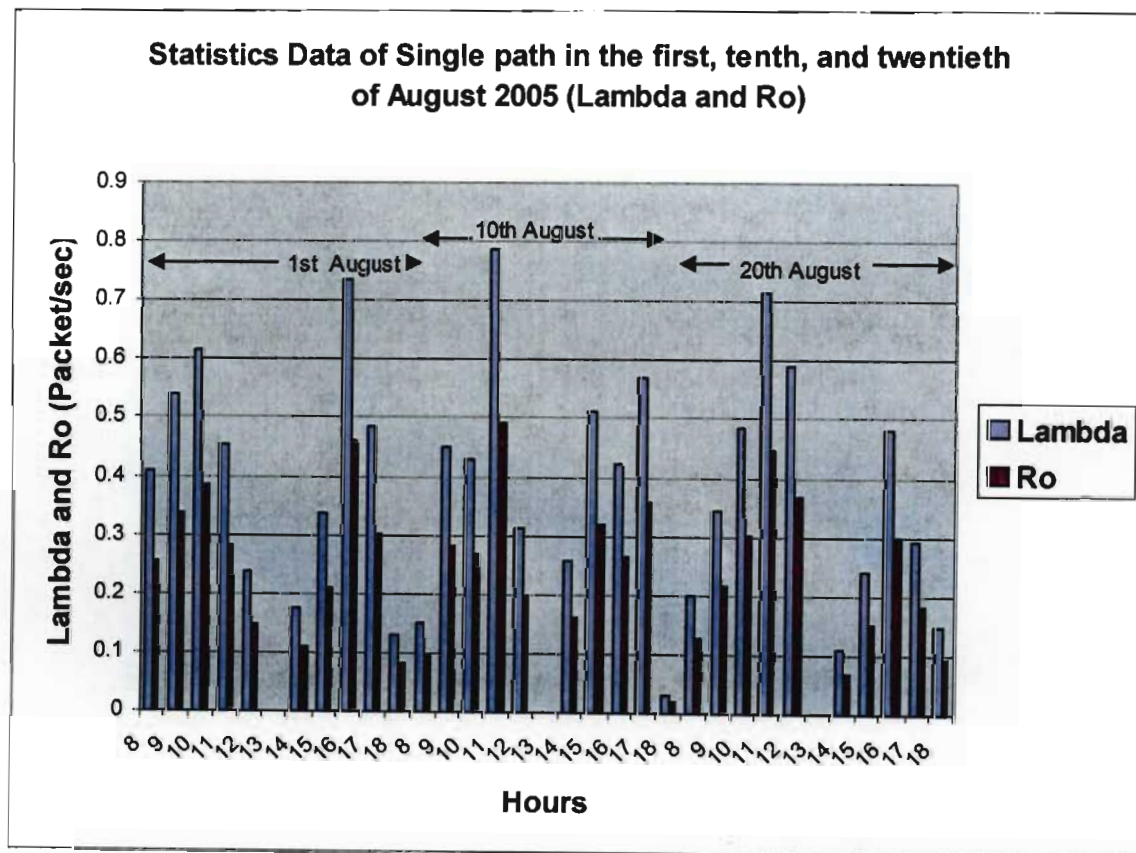


Figure 4.1 Traffic analysis of single path 1st , 10th , and 20th August 2005

Table 4.1 traffic in single path at busy hour of each day during August 2005

Day	Busy Hour (Am/Pm)	Arrival Rate (λ) (Packet/sec)	Utilization (ρ)
1	16 pm	0.734	0.459
2	11 am	0.853	0.494
3	9 am	0.815	0.411
4	12 noon	0.884	0.552
5	16 pm	0.624	0.39
6	12 noon	0.763	0.477
8	9 am	0.66	0.355
9	12 noon	0.62	0.392
10	11 am	0.788	0.492
11	11 am	0.821	0.513
12	16 pm	0.802	0.561
13	12 noon	0.665	0.409
15	16 pm	0.675	0.41
16	11 am	0.68	0.429
17	16 pm	0.439	0.274
18	11am	0.76	0.475
19	10 am	0.448	0.28
20	11 am	0.714	0.446
22	12 noon	0.4	0.246
23	10 am	0.806	0.504
24	12 noon	0.445	0.278
25	10 am	0.424	0.265
26	12 noon	0.483	0.301
27	12 noon	0.496	0.31
29	11 am	1.067	0.667
30	10 am	0.883	0.552
31	11 am	0.983	0.58
Average		0.6937	0.427

4.2.1 Throughput analysis results in single path

From the analysis of the typical days that are discussed in section 4.2, we are looking at the throughput during 1st, 10th and 20th of August 2005. As expected the graphs in figure 4.2, show high throughput in the same busy hour of each day that is discussed in section 4.2. The throughput on the 1st of August 2005 was 0.731 Packet/sec at the expected busy hour of 4 pm. We observe from the graph 0.785 Packet/sec throughput on 10 August 2005 at the expected busy hour of 11 am. It can be seen from the graph that the throughput on 20 August 2005 was 0.712 Packet/sec at the expected busy hour of 11 am. Table 4.2 shows the daily busy hour throughput during August 2005. We observe from the table the typical busy hour throughput on 29 August 2005 at the expected busy hour of 11 am with 1.064 Packet/sec throughput, and the average of the

daily busy hour throughput during August was 0.68 Packet/sec throughput. The table also shows 11 am and 12 noon as the typical busy hours during August 2005.

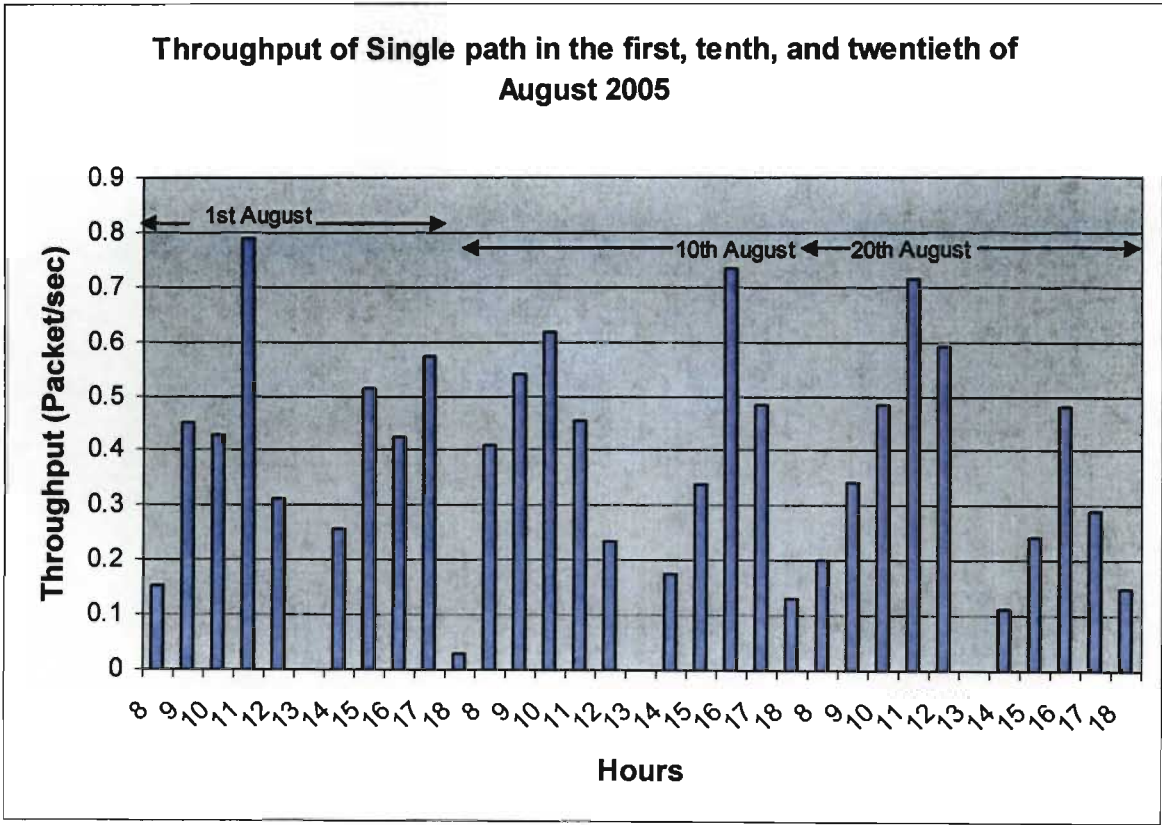


Figure 4.2 describes the throughput in 1st, 10th, and 20th of August 2005

Table 4.2 throughputs in single path at busy hour of each day during August 2005

Day	Busy Hour (Am/Pm)	Throughput (Packet/sec)
1	16 pm	0.731
2	11 am	0.852
3	9 am	0.813
4	12 noon	0.881
5	16 pm	0.622
6	12 noon	0.761
8	9 am	0.658
9	12 noon	0.618
10	11 am	0.785
11	11 am	0.819
12	16 pm	0.8
13	12 noon	0.663
15	16 pm	0.672
16	11 am	0.678
17	16 pm	0.437
18	11 am	0.757
19	10 am	0.446
20	11 am	0.712
22	12 noon	0.388
23	10 am	0.803
24	12 noon	0.443
25	10 am	0.422
26	12 pm	0.481
27	12 noon	0.493
29	11 am	1.064
30	10 am	0.88
31	11 am	0.979
Average		0.68

4.2.2 Packet loss analysis result in single path

Through the analysis result of the typical days discussed in section 4.2, we are looking at the packet loss ratio during 1st, 10th and 20th of August 2005. The graphs in figure 4.3 show unexpected high packet loss at 8 am, 9 am and 3 pm; because those hours were not busy during the traffic hours as shown in figure 4.1 and there was not any high delay at 8 am, 9 am, and 3 pm as shown in figure 4.4. There is no logical reason for this loss (0.00305-0.00388 packet/sec) at these hours. High packet loss ratio was expected at 4 pm, since this hour was the busy hour during the traffic hours and the delay was also high in the mentioned hour (see figure 4.4). This could be reasonable for this packet loss ratio 0.0038 packet/sec on the 1st of August 2005. Figure 4.3 also shows unexpected packet loss ratio at 10, 12 am and 2 and 3 pm on the 10th of August

2005 because those hours were not busy during the traffic time and the delay was normal in the mentioned hours (see figure 4.4). Therefore, there is no logical reason for this loss (0.0016-0.0025 packet/sec) at the hours of 10, 12 am and 2 and 3 pm as shown in figure 4.3. Finally, on the 20th of August 2005, we expected more packet loss ratio at 11 am because this hour was the busy hour during the traffic hours between Asmara and Massawa as shown in figure 4.1. However, the graphs in figure 4.3 show an unexpected high packet loss ratio at 8 am while 8 am was not a busy hour during the traffic hours and there is no high delay at 8 am (see figure 4.4), which results in the loss of (0.0036 packet/sec). Therefore, again, there is no logical reason for this loss. The high packet loss ratio that occurred at 11 am on 20 August 2005, as shown in figure 4.3, was expected because this hour was the busy hour during the traffic hours and the delay was also high in the mentioned hour (see figure 4.4). Therefore, it is logical to obtain this loss (0.0033 – 0.0036 packet/sec) during the said traffic hours. Table 4.3 shows a daily busy hour packet loss ratio during August 2005. We observe from the table, the typical busy hour packet loss ratio on 29 August 2005 at the expected busy hour of 11 am with 0.00912 Packet/sec, and the average of the daily busy hour packet loss ratio during August, was 0.00449 Packet/sec. The table also shows 11 am and 12 noon as the typical busy hour during August 2005. The analysis concludes that the reason for the illogical packet loss ratio at unexpected hours could be from the dial up line or from the modem.

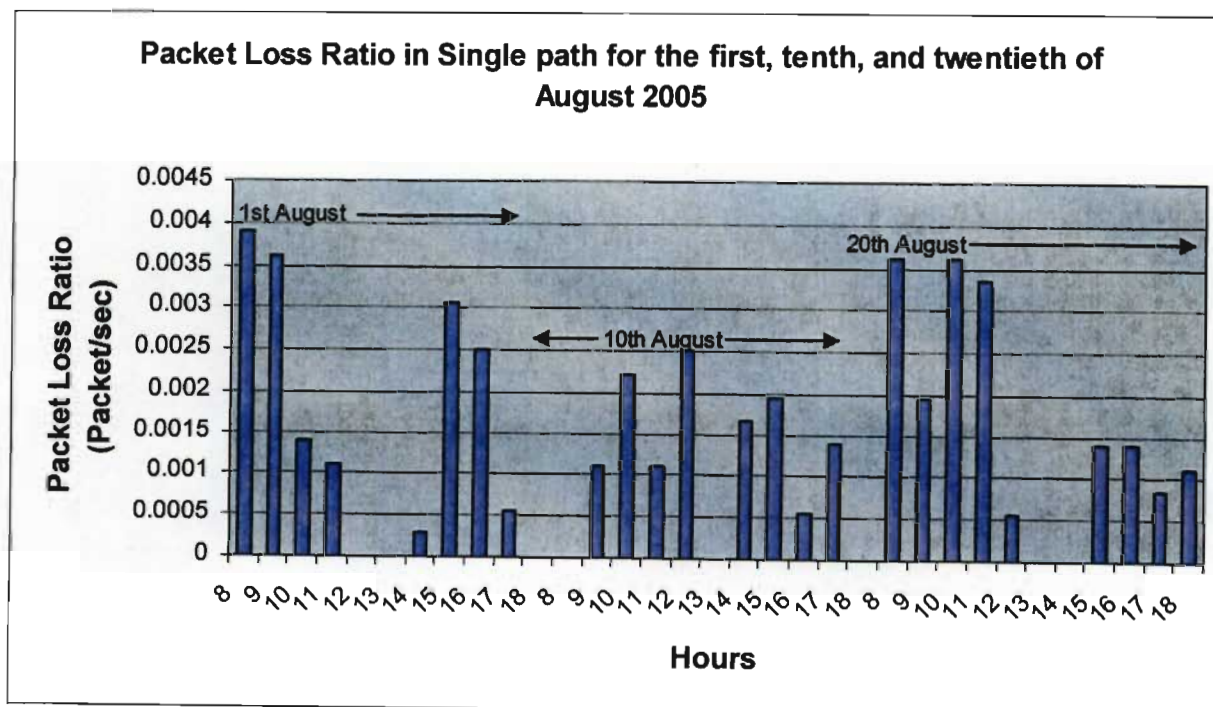


Figure 4.3 describes the packet loss ratio in 1st, 10th, and 20th of August 2005

Table 4.3 packet loss in single path at busy hour of each day during August 2005

Day	Busy Hour (Am/Pm)	Packet Loss Ratio
1	8 am	0.0036
2	11am	0.0066
3	9 am	0.00203
4	12 noon	0.005
5	16 pm	0.00531
6	12 noon	0.00109
8	9 am	0.00291
9	12 noon	0.00264
10	12 noon	0.00566
11	11 am	0.00135
12	16 pm	0.00492
13	12 noon	0.00235
15	16 pm	0.00504
16	11 am	0.00562
17	16 pm	0.0012
18	11 am	0.0076
19	10 am	0.00554
20	11 am	0.0033
22	12 noon	0.0028
23	10 am	0.00445
24	12 noon	0.00372
25	10 am	0.00585
26	12 noon	0.00113
27	12 noon	0.0066
29	11 am	0.00912
30	10 am	0.0072
31	11 am	0.0086
Average		0.00449

4.2.3 Delay analysis result in single path

To determine the delay in the queue, we assume a Markovian (M/M/1) queue. From the analysis of the typical days discussed in section 4.2, we are looking at the delay during 1st, 20th and 20th of August 2005. As expected the graphs in figure 4.4 show high delay in the same busy hours as discussed in section 4.2. We observe from the graphs that the delay on the 1st of August was 0.530 second at the expected busy hour of 4 pm. It can also be seen from the graphs that the delay on the 10th of August was 0.659 second at the expected busy hour of 11 am. The graphs also show

that the delay on the 20th of August was 1.18 second at the expected busy hour, 11 am. Table 4.4 shows the daily busy hour delay during August 2005. We observe from the table that the typical busy hour delay on 29 August 2005, at the expected busy hour of 11 am, was 1.336 second and the average of the daily busy hour delay during August was 0.403 second. The table also shows 11am and 12 noon as the typical busy hours during August 2005. From analyzing the traffic results in the busy hour of the typical days (1st, 10th and 20th of August 2005) and the daily busy hour in August 2005, the study found the performance of WAN with the single link showed a weak performance based on 1.336 second of high delay in the queue and 0.00912 of high packet loss ratio.

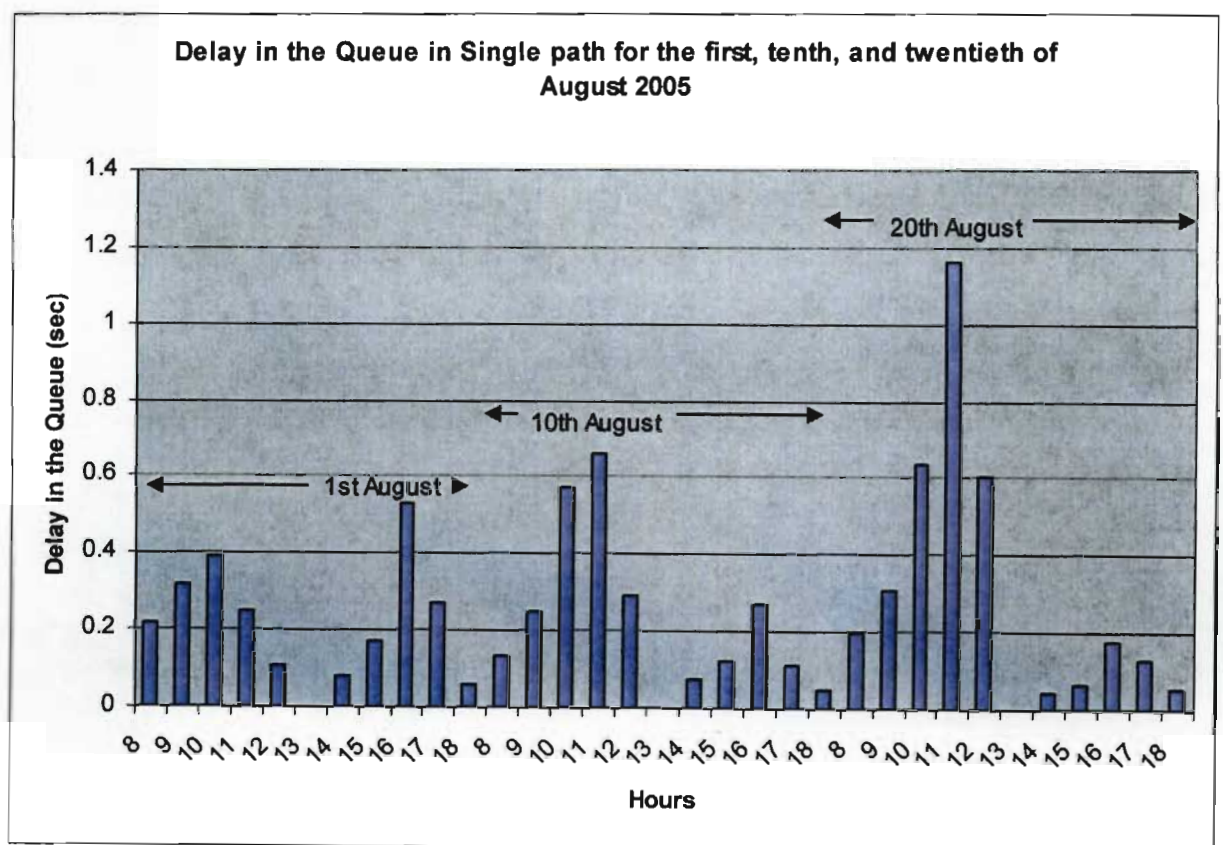


Figure 4.4 describes the delay in Queue for 1st, 10th, and 20th of August 2005

Table 4.4 delays in the queue in single path at busy hour of each day during August 2005

Day	Busy Hour (Am/Pm)	Delay in the Queue (sec)
1	16 pm	0.530
2	11 am	0.478
3	9 am	0.53
4	12 noon	0.682
5	16 pm	0.5
6	12 noon	0.435
8	9 am	0.289
9	12 noon	0.253
10	11 am	0.659
11	11 am	0.542
12	16 pm	0.718
13	12 noon	0.284
15	16 pm	0.286
16	11 am	0.0323
17	16 pm	0.104
18	11am	0.43
19	10 am	0.109
20	11 am	1.18
22	12 noon	0.216
23	10 am	0.513
24	12 noon	0.107
25	10 am	0.095
26	12 noon	0.13
27	12 noon	0.139
29	11 am	1.336
30	10 am	0.681
31	11 am	0.981
Average		0.403

4.3 Dual path traffic analysis (from measurement)

From the analysis results of dual link measured traffic for September 2005 we again select three typical days (1st, 10th and 20th of September 2005) and the daily busy hour of September 2005 to study the performance of the WAN between Asmara and Massawa. The traffic is measured based on 20 users as the size of LAN in both areas (Asmara and Massawa). From the analysis of the dual path capacity we found the average of the service rate (μ) is equal to 0.8 packet/sec. Figure 4.5 and Table 4.5 describe the traffic on different days and the daily busy hour in September

2005. In figure 4.5 the graphs show the busy hour arrival rate (λ) and utilization (ρ) in the busy hour of each day. The busy hour on 1st of September was 4 pm with 0.340 Packet/sec arrival rate (λ) and 0.213 Packet/sec channel utilization (ρ). It can be seen from the same graphs that the busy hour on 10 September 2005 is 11 am with 0.524 Packet/sec arrival rate (λ) and 0.327 Packet/sec channel utilization (ρ). We also observe from the graph that 4 pm was the busy hour on 20 September 2005 with 0.414 Packet/sec arrival rate (λ) and 0.258 Packet/sec channel utilization (ρ). The graphs also show zero traffic in figure 4.5 during lunch break. Table 4.5 shows the typical busy hour arrival rate at 11 am on 15 September 2005 with 0.597 Packet/sec arrival rate (λ) and 0.375 Packet/sec channel utilization. The table also shows 0.446 Packet/sec as the average of the daily busy hour arrival rate (λ) and 0.277 Packet/sec as the average of the daily channel utilization (ρ). The table also shows 11am and 12 noon as the typical busy hours during August 2005. The analysis concludes that the customers preferred 11-12 am to start service process. The throughput, packet loss, and delay parameters will be discussed in sections 4.3.1, 4.3.2, and 4.3.3.

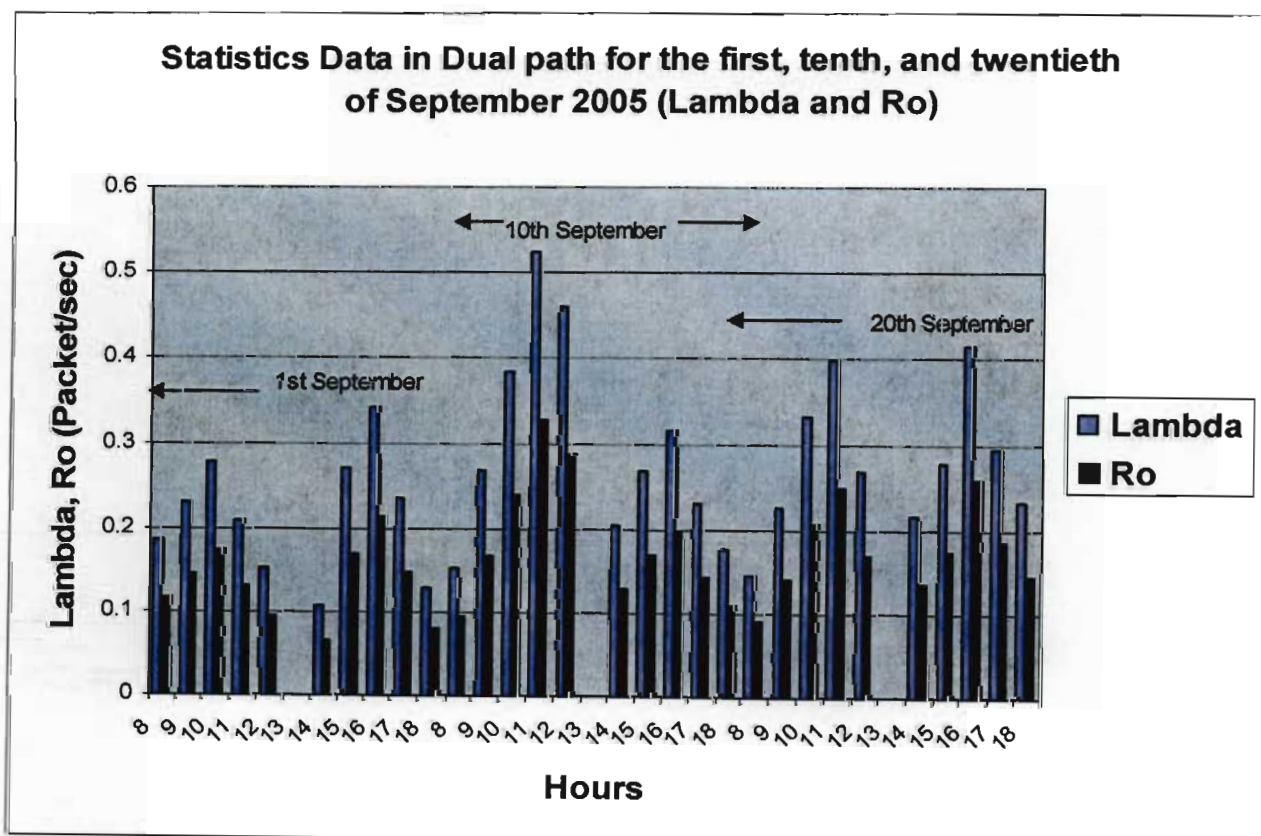


Figure 4.5 Traffic analysis of dual path during 1st, 10th, and 20th September 2005

Table 4.5 traffic in dual path at busy hour of each day during September 2005

Day	Busy Hour (Am/Pm)	Arrival Rate (λ) (Packet/sec)	Utilization (ρ)
1	16 pm	0.34	0.213
2	11 am	0.345	0.215
3	17 pm	0.353	0.221
5	12 noon	0.475	0.285
6	17 pm	0.441	0.275
7	12 noon	0.313	0.195
8	12 noon	0.473	0.295
9	12 noon	0.505	0.315
10	11 am	0.524	0.327
12	11 am	0.478	0.299
13	12 noon	0.476	0.297
14	12 noon	0.317	0.198
15	11 am	0.597	0.375
16	12 noon	0.492	0.307
17	11 am	0.435	0.275
19	11 am	0.457	0.285
20	16 pm	0.414	0.258
21	11 am	0.323	0.201
22	11 am	0.596	0.372
23	11 am	0.483	0.301
24	16 pm	0.509	0.318
26	11 am	0.505	0.314
27	11 am	0.449	0.28
28	11 am	0.394	0.218
29	10 am	0.483	0.302
30	10 am	0.435	0.272
Average		0.446	0.277

4.3.1 Throughput analysis result in dual path

From the analysis of the typical days that are discussed in section 4.3, we are looking at the throughput during 1st, 10th and 20th of September 2005. As expected the graphs in figure 4.6, show high throughput in the same busy hour of each day as discussed in section 4.3. The throughput on the 1st of September was 0.337 Packet/sec at the expected busy hour of 4 pm. We observe from the graph 0.521 Packet/sec throughput on 10 September 2005 at the expected busy hour of 11 am. It can be seen from the graph that the throughput on 20 September 2005 was 0.411

Packet/sec at the expected busy hour of 4 pm. Table 4.6 shows the daily busy hour throughput during September 2005. We observe from the table, the typical busy hour throughput on 15 September 2005 at the expected busy hour of 11 am with 0.593 Packet/sec throughput. The average of the daily busy hour throughput during September was 0.464 Packet/sec throughput. The table also shows 11am and 12 noon as the typical busy hours during August 2005

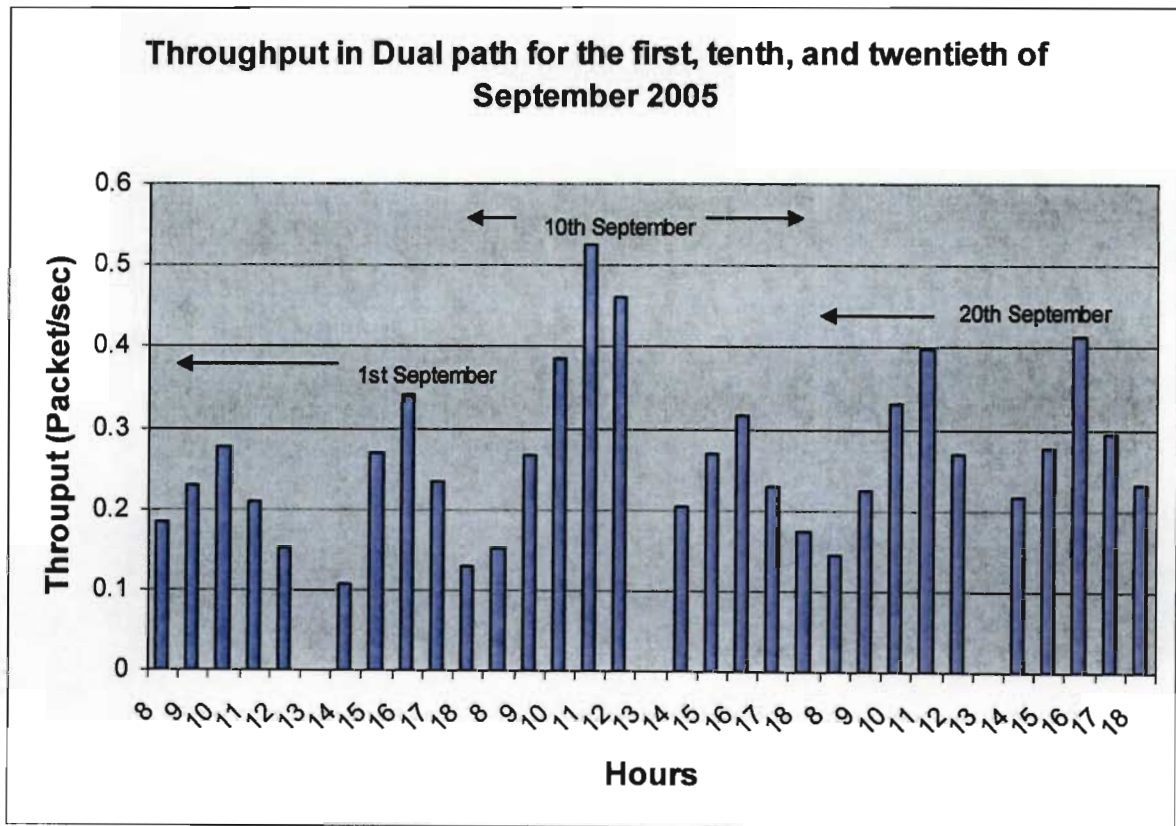


Figure 4.6 describes the throughput in 1st, 10th, and 20th of September 2005

Table 4.6 throughputs in dual path at busy hour of each day during September 2005

Day	Busy Hour (Am/Pm)	Throughput (Packet/sec)
1	16 pm	0.337
2	11 am	0.342
3	17 pm	0.351
5	12 noon	0.472
6	17 pm	0.439
7	12 noon	0.312
8	12 noon	0.471
9	12 noon	0.501
10	11 am	0.521
12	11 am	0.475
13	12 noon	0.473
14	12 noon	0.315
15	11 am	0.593
16	12 noon	0.49
17	11 am	0.432
19	11 am	0.455
20	16 pm	0.411
21	11 am	0.322
22	11 am	0.591
23	11 am	0.481
24	16 pm	0.505
26	11 am	0.502
27	11 am	0.446
28	11 am	0.392
29	10 am	0.481
30	10 am	0.433
Average		0.464

4.3.2 Packet loss analysis result in dual path

Through the analysis of the typical days discussed in section 4.3, we consider the packet loss during the 1st, 10th and 20th of September 2005. The graphs in figure 4.7 show high packet loss at 4 pm; and this loss is expected because 4 pm was the busy hour during the dual traffic hours between Asmara and Massawa as shown in Figure 4.5. Therefore, it is logical to expect this loss (0.00324 packet/sec) at 4 pm on the 1st of September 2005. We observe from the graphs more packet loss ratio on 10 September 2005 at the expected busy hour of 11 am. Therefore, it is logical to expect this loss (0.00364 packet/sec). It can be seen from the graphs high packet loss

ratio (0 .00351 Packet/sec) on 20 September 2005 at the expected busy hour of 11 am. We observe from the graph a high packet loss ratio in the unexpected hour of 12 am because this hour was not busy during the traffic hours and there is no high delay at this hour (see figure 4.8). Therefore, we did not expect to get this illogical loss at this hour. Table 4.7 shows the daily busy hour packet loss ratio during September 2005. We observe from the table that the typical busy hour packet loss ratio on 15 September 2005 at the expected busy hour of 11 am, with 0.00418 Packet/sec, and the average of the daily busy hour packet loss ratio during September was 0.00332 Packet/sec. The table also shows 11am and 12 noon as the typical busy hours during August 2005. The analysis concludes that the reason for the illogical packet loss ratio at unexpected hours could be from dial up line or from the modem.

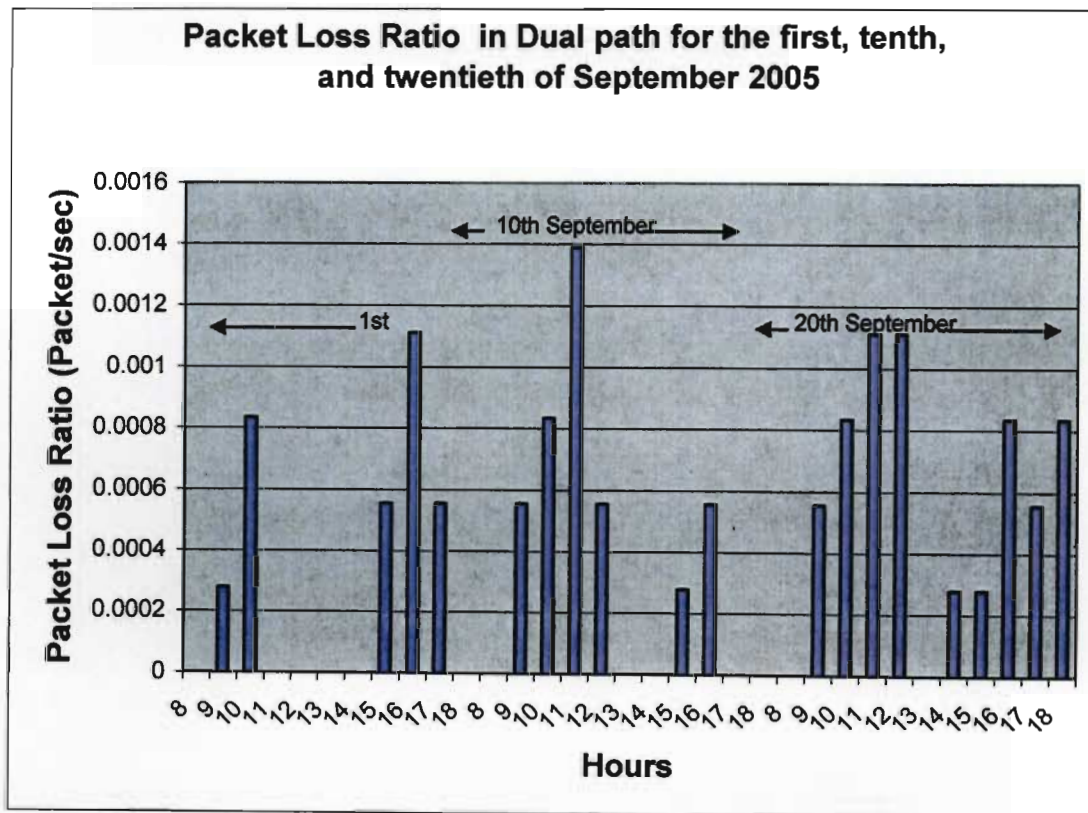


Figure 4.7 describes the packet loss in 1st, 10th, and 20th of September 2005

Table 4.7 packet loss in dual path at busy hour of each day during September 2005

Day	Busy Hour (Am/Pm)	Packet Loss Ratio
1	16 pm	0.00324
2	11 am	0.004
3	17 pm	0.00456
5	12 noon	0.0051
6	17 pm	0.00188
7	12 noon	0.0013
8	12 noon	0.00375
9	12 noon	0.00221
10	11 am	0.00364
12	11 am	0.00371
13	12 noon	0.00332
14	12 noon	0.00217
15	11 am	0.00418
16	12 noon	0.00321
17	11 am	0.00362
19	11 am	0.00411
20	11 am	0.00351
21	11 am	0.00185
22	11 am	0.00412
23	11 am	0.00372
24	16 pm	0.00471
26	11 am	0.00471
27	11 am	0.00361
28	11 am	0.00237
29	10 am	0.00286
30	10 am	0.00254
Average		0.00332

4.3.3 Delay analysis result in dual path

To determine the delay in the queue, we assume a Markovian (M/M/2) queue. From the analysis of the typical days discussed in section 4.3, we are looking at the delay during 1st, 10th and 20th of August 2005. As expected the graphs in figure 4.8 show high delay in the same busy hours as discussed in section 4.3. We observe from the graphs that the delay on the 1st of September was 0.110 second at the expected busy hour of 4 pm. It can also be seen from the graphs that the delay on the 10th of September was 0.215 second at the expected busy hour of 11 am. The graphs also

show that the delay on the 20th of September was 0.219 second at the expected busy hour of 4 pm. Table 4.8 shows the daily busy hour delay during September 2005. We observe from the table that the typical busy hour delay on 15 September 2005 at the expected busy hour of 11 am, with 0.336 second and the average of the daily busy hour delay during September, was 0.192 second. The table also shows 11am and 12 noon as the typical busy hours during September 2005. From analyzing the traffic result in the busy hours of the typical days (1st, 10th and 20th of September 2005) and the daily busy hour of September 2005, the study found that the performance of WAN with dual link shows a weak performance based on 0.336 second of high delay in the queue and 0.00418 of high packet loss ratio.

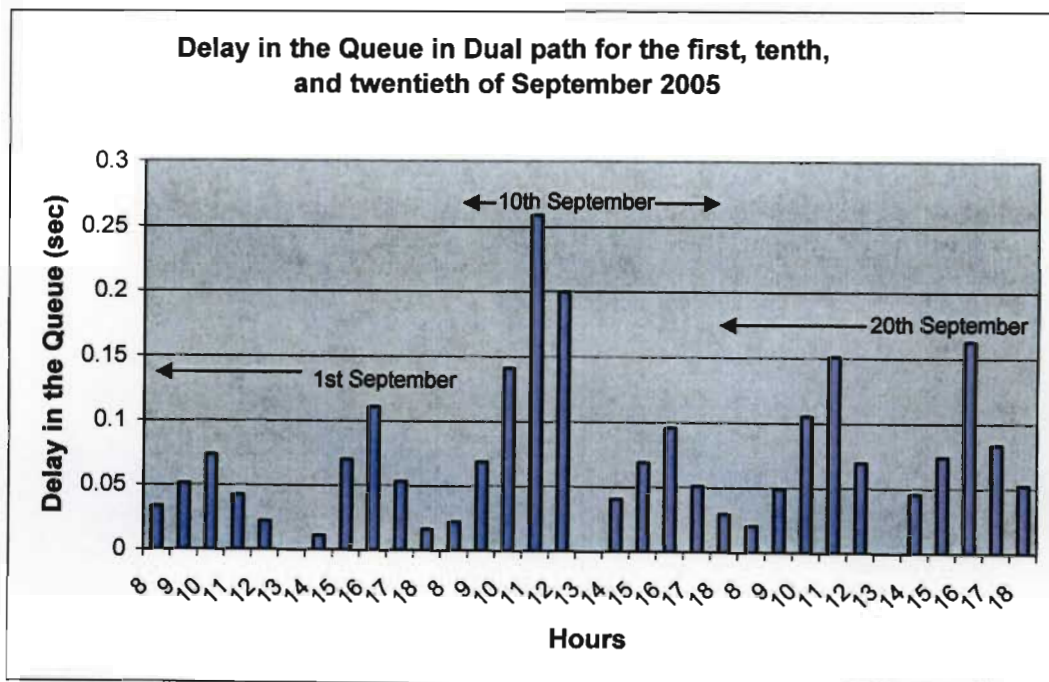


Figure 4.8 describes the delay in Queue for 1st, 10th, and 20th of September 2005

Table 4.8 delay in the queue in dual path at busy hour of each day during September 2005

Day	Busy Hour (Am/Pm)	Delay in the Queue (sec)
1	16 pm	0.11
2	11 am	0.113
3	17 pm	0.118
5	12 noon	0.197
6	17 pm	0.183
7	12 noon	0.093
8	12 noon	0.21
9	12 noon	0.24
10	11 am	0.215
12	11 am	0.258
13	12 noon	0.213
14	12 noon	0.096
15	11 am	0.336
16	12 noon	0.228
17	11 am	0.179
19	11am	0.197
20	16 pm	0.219
21	11 am	0.099
22	11 am	0.335
23	11 am	0.162
24	16 pm	0.244
26	11 am	0.244
27	11 am	0.19
28	11 am	0.116
29	10 am	0.22
30	10 am	0.179
Average		0.192

4.4 Comparing the single with dual path performance

This section compares the performance of WAN through the parameters analysis, which are the results of single and dual path as discussed in sections 4.2.1-4.2.3 and 4.3.1-4.3.3. Table 4.9 shows the performance results of single and dual path in the typical busy hours of August and September 2005. It can be seen from table 4.9 that the traffic in single path scores 1.067 Packet/sec. However, the high packet loss ratio reduces the performance of the WAN. The table also shows high traffic with high loss ratio and high delay in dual link. However, these results are considerably better than the single link results and this is obviously because the dual link has half

the capacity of the single link. Therefore, it is logical to find the result of the dual link better than the single link result.

Table 4.9 describes the comparison performance results between single and dual path

Parameters	Single Path	Dual Path
Traffic	1.067 packet/sec	0.597 packet/sec
Packet Loss	0.00912 packet/sec	0.00418 packet/sec
Delay in the Queue	1.336 second	0.336 second

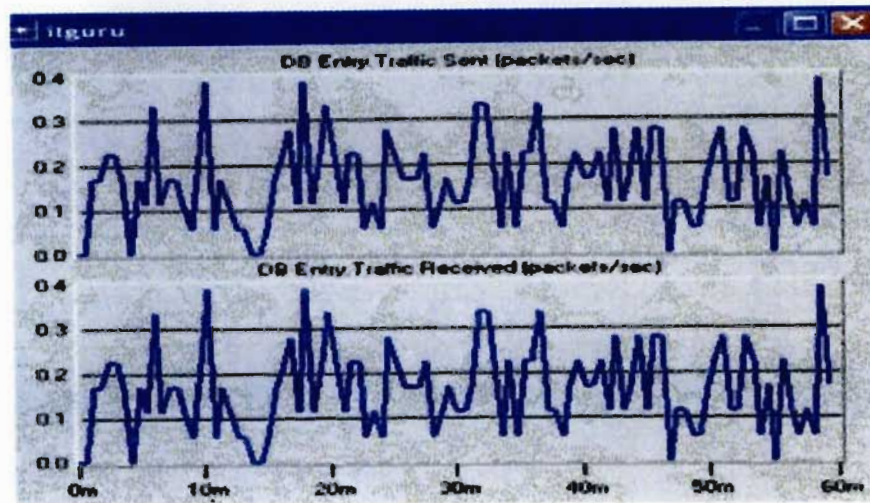
4.5 LAN Performance simulation results

This section discusses the simulation results of LAN performance through star topology (which is the topology used in Eritrean LANs) with hub connection, switch connection, and router connection using Database application and downloading files for the traffic between the users and the server. We compare the performance results that are obtained from hub, switch, and router connection to select the best performance for the LAN from the three connection performance results. Collision, packet loss, and queue delay were the key parameters to study the performance of LAN with different connections. The simulation results are described in sections 4.5.1, 4.5.2, and 4.5.3. Note that this simulation is necessary since there was no access to measured LAN traffic in Eritrea.

4.5.1 Simulation result of LAN with hub connection

Figures 4.9 and 4.10 describe the traffic simulation results between the LAN workstations and the server using database application and file transfer protocol to request information and download files from the server (note: these are the two major contributions to LAN traffic). The average rate of transmission and reception of the packets between the workstations and the server was 0.398 Packet/sec as shown in figure 4.9. The graphs in figure 4.10 also show little traffic during the 60 minutes of simulation, and the average rate of transmission and reception of the packets between the workstations, was between (0.055-0.320 packet/sec). This means the files that were transferred between the users were very few during the simulation hour. Therefore, it was logical to observe little traffic in figure 4.10. Collision, packet loss, and queue delay will be discussed in section 4.5.1.1.

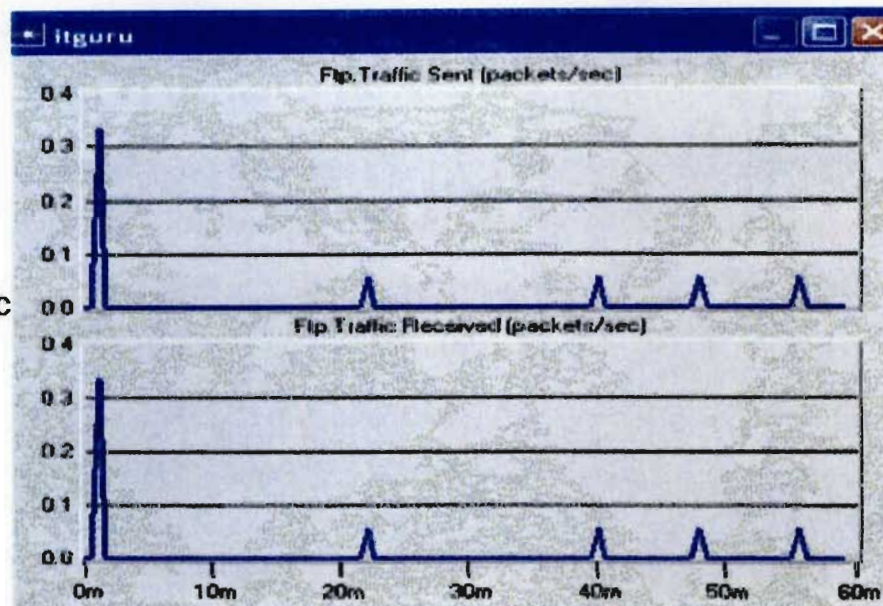
Packet/sec



Simulation Time

Figure 4.9 Traffic between workstations and server using database application with hub connection

Packet/sec



Simulation Time

Figure 4.10 Traffic between workstations and server using Ftp with hub connection

4.5.1.1 Collision, packet loss, and queue delay results (hub connection)

Through the traffic hour between the users and the server that are discussed in section 4.5.1, we are looking at collision, packet loss, and queue delay to study the performance of LAN through hub connection during the traffic hour, as shown in Figures 4.9 and 4.10. The graphs in figure

4.11 show the collisions that occurred in each workstation during the traffic hour. From the same graphs in figure 4.11, we observe a high collision ratio (0.075, 0.0638, and 0.0527) in workstations 4, 5, 6 and low collision is recorded once in the networks. The main reason for these high collisions could be from the hub because the hub repeats the incoming packets to all its ports and this repeat create a collision. The graphs in figure 4.12 show the packet loss ratio in each workstation during the traffic hour. It can also be seen from the graphs in figure 4.12. More packet loss (0.0186- 0.0230) in workstation 4, 5 and 6. The main reason of these losses could be from the collision during the traffic hour, as shown in figure 4.11. The graphs in figure 4.13 show the delay in each workstation during traffic hour. The graphs also show more delay (0.320 – 0.410 ms) in workstation 1,2,3 and 6.

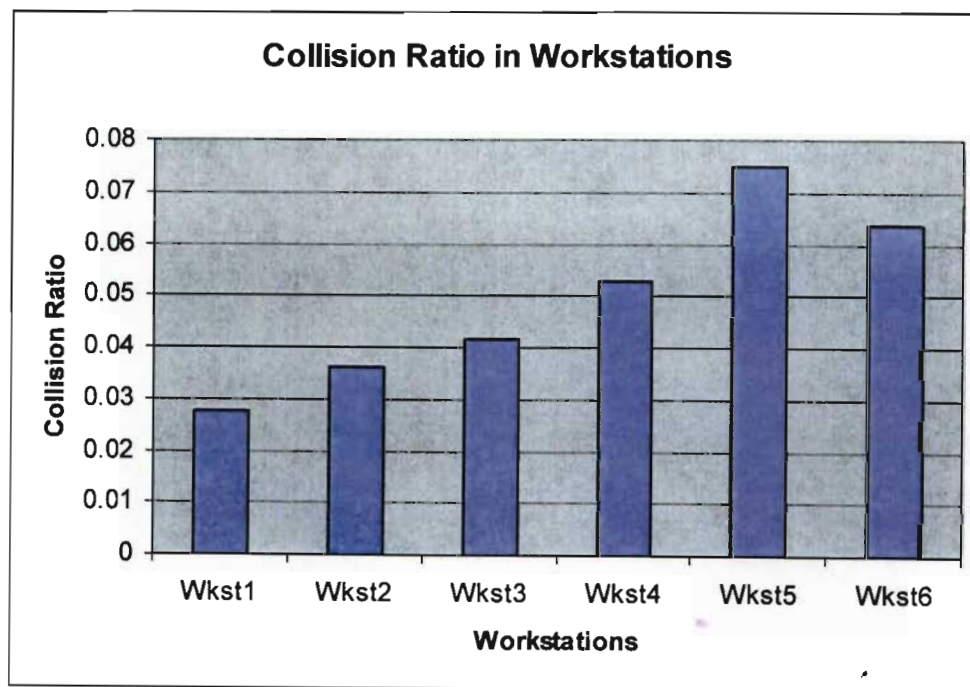


Figure 4.11 Results of collision in each workstation

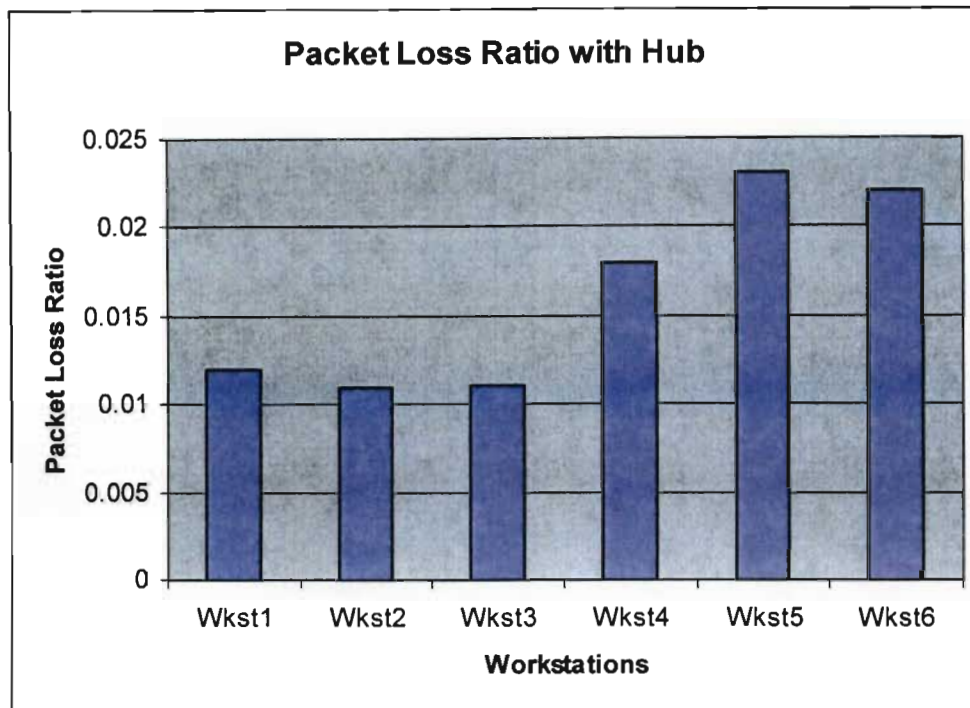


Figure 4.12 Result of packet loss in each workstation

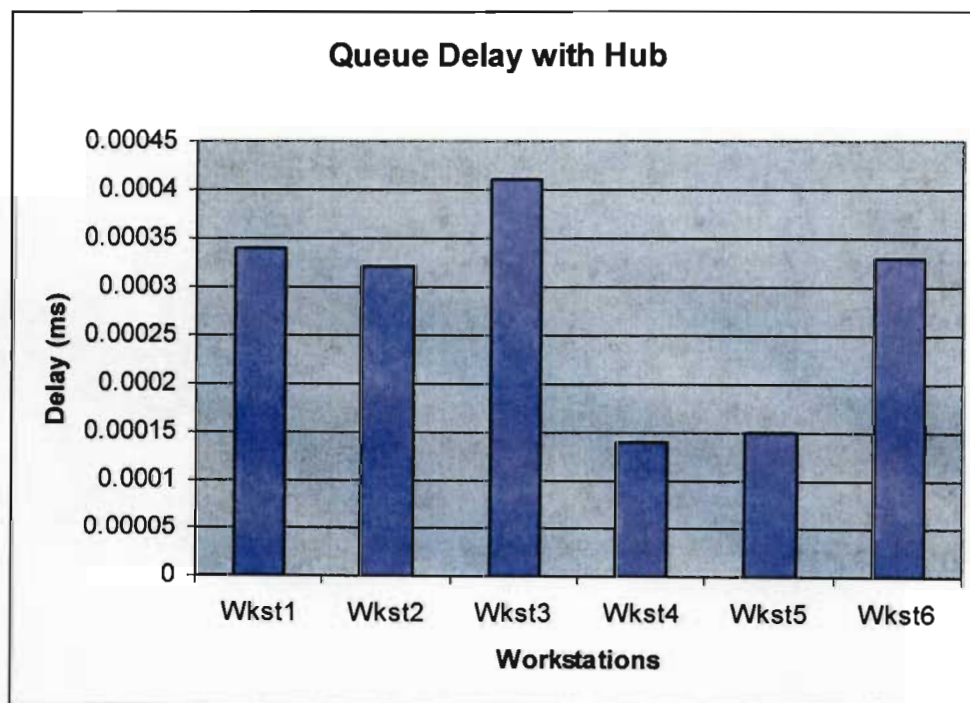


Figure 4.13 Result of queue delay in each workstation

4.5.2 Simulation results of LAN with switch connection

Figures 4.14 and 4.15 describe the simulation traffic results of the LAN with switch connection using database application and file transfer protocol to request information and download files from the server. The average rate of transmission and reception of the packets between the workstations and the server was 0.45 Packet/sec as shown in figure 4.14. The graphs in figure 4.15 also show little traffic during the 60 minutes of simulation and the average rate of transmission and reception of the packets between the workstations was between 0.055-0.350 Packet/sec. During the simulation traffic hour, there was not much demand for file transfer between the users in the LAN. Therefore, it was logical to observe a little traffic during an hour of traffic as shown in figure 4.15. Collision, packet loss, and queue delay will be discussed in section 4.5.2.1.

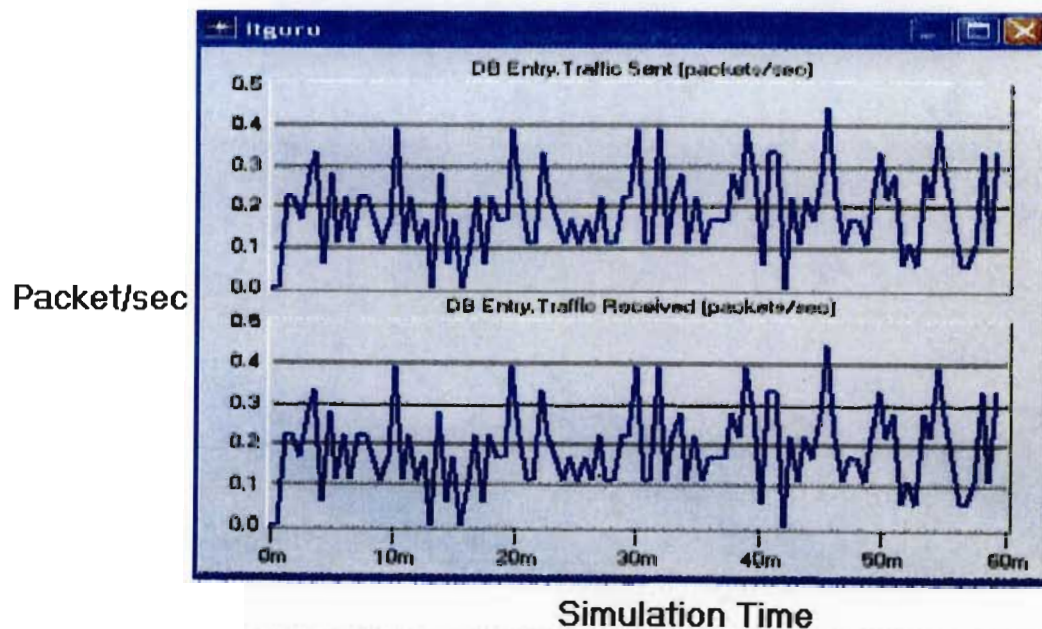


Figure 4.14 traffic between workstations and server using database application with switch connection

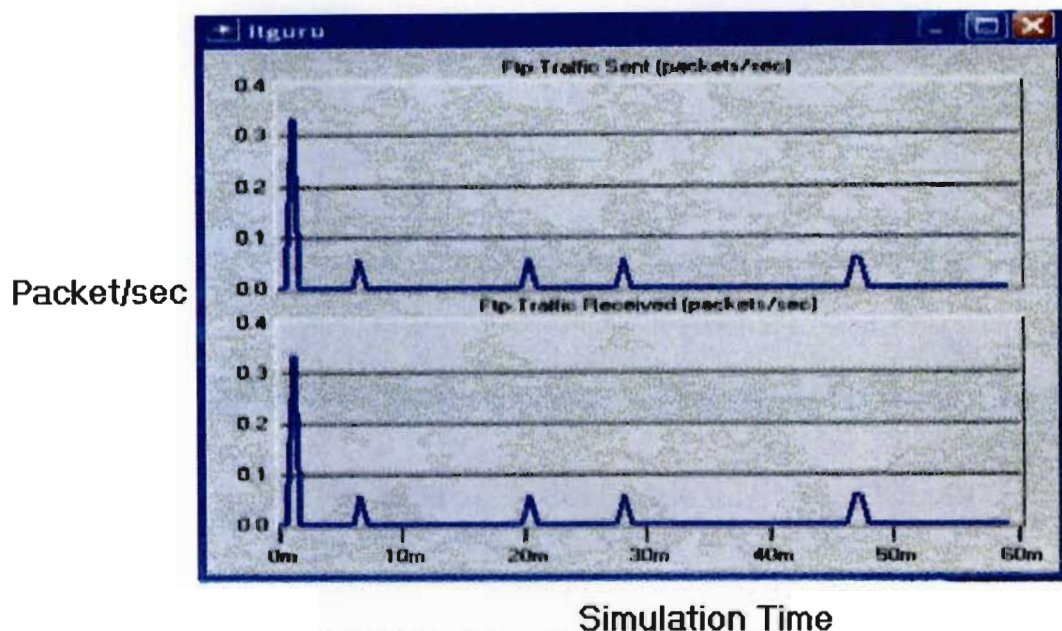


Figure 4.15 traffic between workstations and server using Ftp with switch connection

4.5.2.1 Collision, packet loss, and queue delay results (switch connection)

Through the traffic hour between the users and the server that are discussed in section 4.5.2, we are looking at collision, packet loss, and queue delay to study the performance of LAN through switch connection during the simulation traffic hour. Analyzing the simulation traffic results as shown in figures 4.14 and 4.15, there is no collision during the traffic hour. This is because the switch automatically eliminates the collisions. The analysis also shows that no packet loss occurred during the simulation traffic hour, which means the result is good compared with Elsaadany, Singhal, and Lui [1] packet loss results of 0.05 – 1.06 %. The graphs in figure 4.16 show the delay in each workstation during the simulation traffic hour. From the same graphs in figure 4.16, we observe the delay is more in workstation 2,3 and 4 and these results of the delay 0.0386 – 0.0480 ms are higher than the delay results obtained by Elsaadany, Singhal, and Lui [1] 7.78 – 17.06 ms.

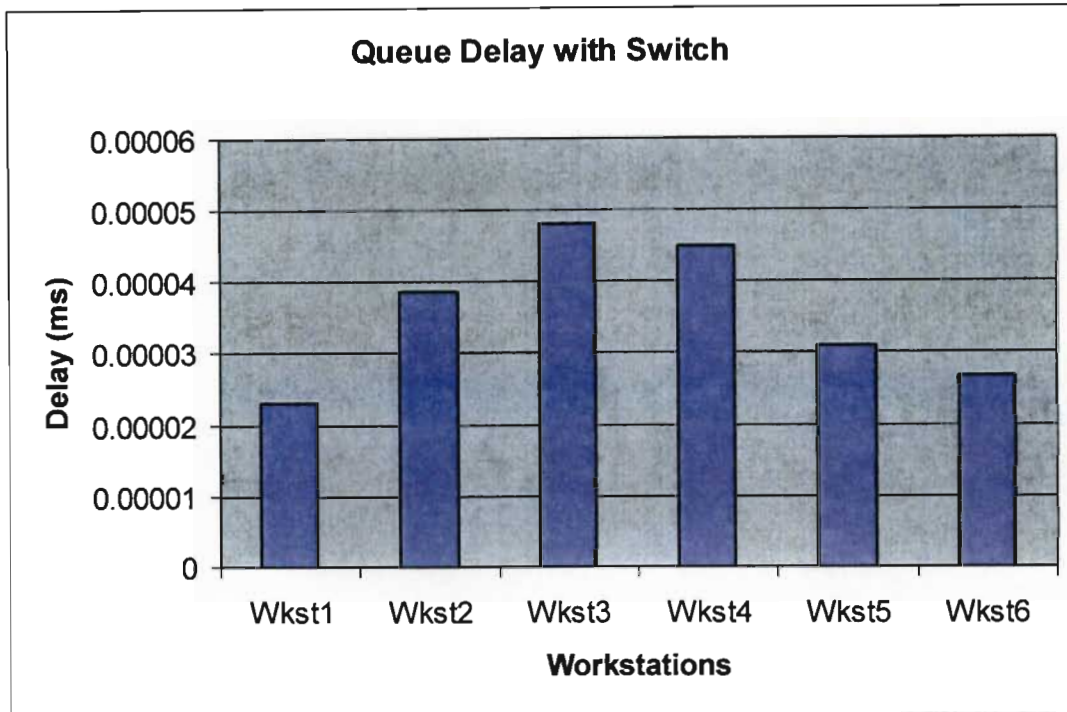


Figure 4.16 Result of queue delay in each workstation with switch

4.5.3 Simulation results of LAN with router connection

Figures 4.17 and 4.18 describe the simulation traffic results between the ministries' LANs using email system and file transfer protocol that are used to exchange messages between the users and to download files. In figure 4.17 the graphs show some high traffic in the first 5 minutes between the ministries LANs. The average rate of transmission and reception of the packets between the LANs is 0.222 Packet/sec. The graphs in figure 4.18 show little traffic in the first 15 minutes between the ministries' LANs. The average rate of transmission and reception of the packets between the ministries' LANs in the first 15 minutes is 0.055 Packet/sec. The graphs also show zero traffic in the remaining minutes because of no files transfer between the ministries' LANs in the remaining minutes. Therefore, it is logical to obtain zero traffic in the remaining minutes as shown in figure 4.18. Collision, packet loss, and queue delay will be discussed in section 4.5.3.1.

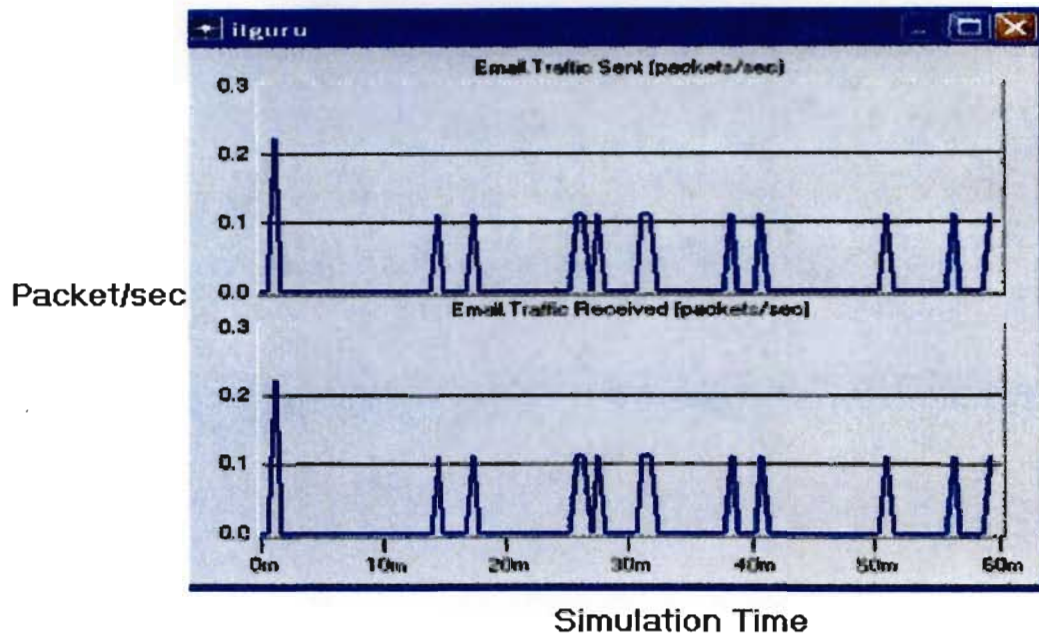


Figure 4.17 traffic between the ministries LANs using email system with router connection

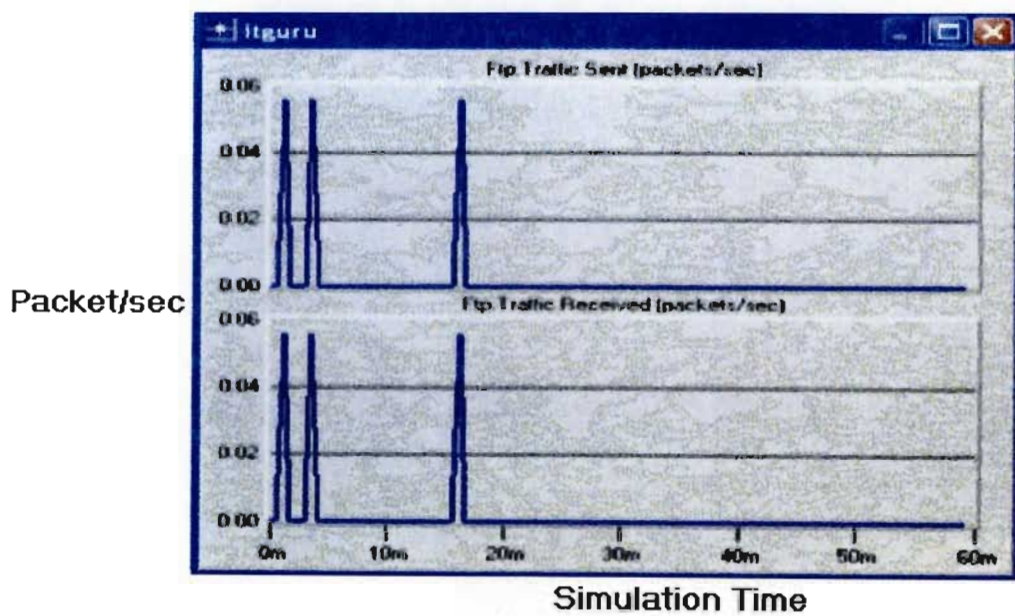


Figure 4.18 traffic between the ministries LANs using Ftp with router connection

4.5.3.1 Collision, packet loss, and queue delay results (router connection)

Through the traffic hour between ministries' LANs that are discussed in section 4.5.3, we are looking at collision, packet loss, and queue delay to study the performance of LAN through router connection during the simulation traffic hour. From analyzing the simulation traffic results as

shown in figures 4.17 and 4.18, there is no collision during the traffic hour because of the router. The analysis also shows that no packet loss occurred during the simulation hour and this result agrees with Kabir, Abdullah, Khatun, Anas and Mahdi [57] study to eliminate packet loss. The graphs in figure 4.19 show the delay in each workstation during the simulation hour. From the graphs, we observe the delay is more in Eritrean Information System Agency (EISA) LAN and Foreign Affairs LAN, and these delays (0.0176 – 0.0200 ms) are more than the result obtained by Kabir, Abdullah, Khatun, Anas and Mahdi [57]. The result obtained from that study was 1.01 ms. The target of study in (section 4.5) was to examine the performance of LAN through star topology with hub connection, switch connection, and router connection using collision, packet loss, and queue delay as key parameters to analyze the simulation traffic results. The study found the performance of LAN with hub connection shows a weak performance based on 0.075 of high collision ratio, 0.0186-0.0230 Packet/sec of high packet loss ratio, and (0.328- 428 ms) of high delay in the queue. The study also found the performance of LAN with switch connection records better performance based on zero collision, zero packet loss ratio and (0.0386-0.0480 ms) of low delay in the queue when compared to a hub connection result. Finally the study found the performance of LAN with router connection records a good performance based on zero collision, zero packet loss ratio, and (0.0176 – 0.0200 ms) of low delay in the queue and compared to a switch connection result.

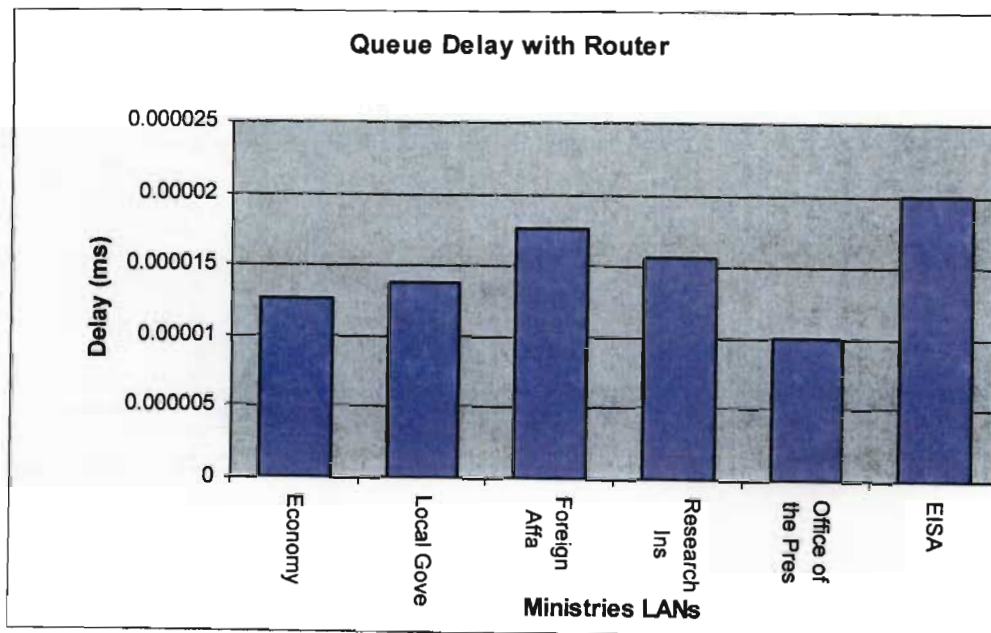


Figure 4.19 Result of queue delay in ministries' LANs

reception of the packets between the two stations was 1.65 packet/sec for downloaded files. Collision, packet loss, and queue delay will be discussed in section 4.6.1.1.

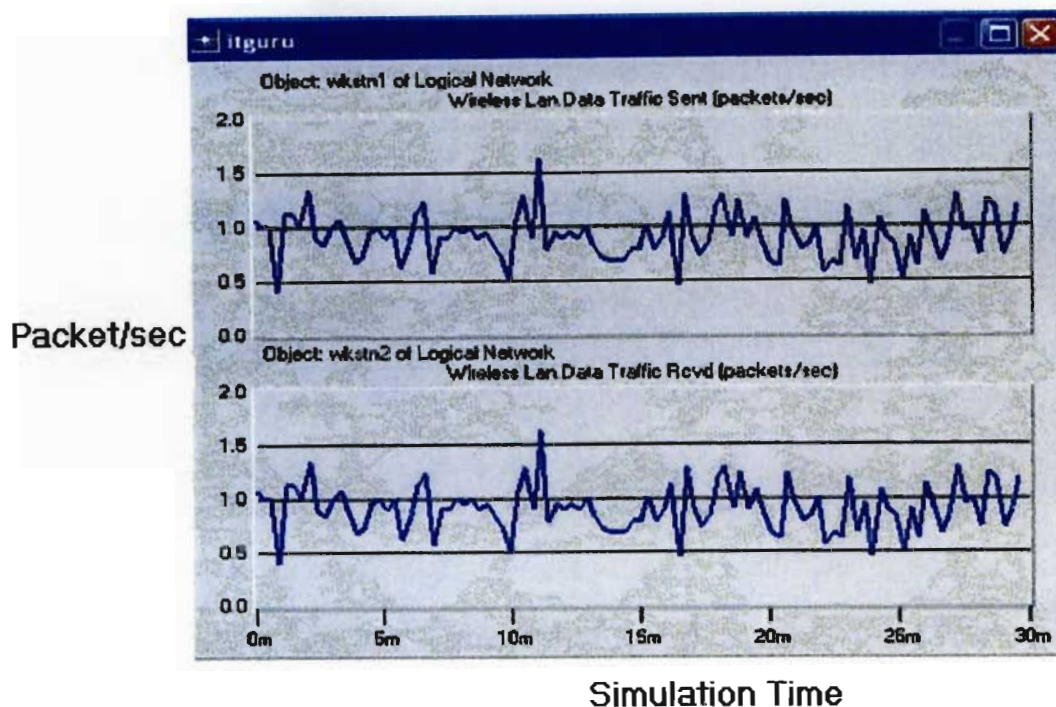


Figure 4.20 traffic between two WLAN base stations

4.6.1.1 Collision, packet loss and queue delay with peer-to-peer WLAN connection

During the half hour simulation traffic between two WLAN stations using file transfer protocol to download files between the two stations as discussed in section 4.6.1, we are looking at collision, packet loss, and queue delay to study the performance of WLAN through peer-to-peer connection. From analyzing the simulation traffic results as shown in figure 4.20, we found 0.0555 packet loss ratio and zero collision. These results are better than the result obtained by Golmie, Cypher, and Rebala [62] in their analysis of low rate wireless technologies for medical applications. Their results, showed 18 % of packet loss. The graphs in Figure 4.21 show the highest delay result of 0.00115 second compared with Golmie, Cypher, and Rebala [62] delay result of 0.0392 second.

4.5.4 Comparing the performance results of LAN with hub, switch, and router

In this section, we compare the performance results of LAN using a star topology with hub, switch, and router connection. Collision, packet loss, and delay in the queue were the key parameters to study the performance of LAN and select the best connection. We conducted an analysis on the performance results of each connection and compared the value of the parameters value between them, as shown in Table 4.10. We found the performance of LAN with router (government networks) connection to record a better performance than the other connections through its minimum delay in the queue, zero packet loss, and zero collision. From this analysis of the study of LAN performance, we suggest that the LAN with router or switch connection can minimize the congestion problem and improve the performance of LAN in each different section of the country.

Table 4.10 Analysis results of LAN performance with different connection

Parameters	Star topology with Hub connection	Star topology with Switch connection	Star topology with Router connection
Collision	0.075	0	0
Packet Loss	0.0186- 0.0230	0	0
Queue Delay	0.328 – 0.428 ms	0.0386 – 0.0480 ms	0.0176 – 0.0200 ms

4.6 WLAN Performance simulation results

This section discusses the simulation results of peer-to-peer WLAN and integrated wireless and wireline networks via Internet backbone, using FTP to download files, HTTP to browse the WebPages, and database application to request information for the traffic. Collision, packet loss, and queue delay are again the key parameters to study the performance of WLAN. The simulations results of WLAN with peer-to-peer connection between two base stations and integrated between wireless and wireline networks via Internet backbone, are described in section 4.6.1 and 4.6.2.

4.6.1 Simulation results of WLAN with peer-to-peer connection

Figure 4.20 describes the simulation traffic results between two WLAN stations using file transfer protocol to download files between the two stations. The average rate of transmission and

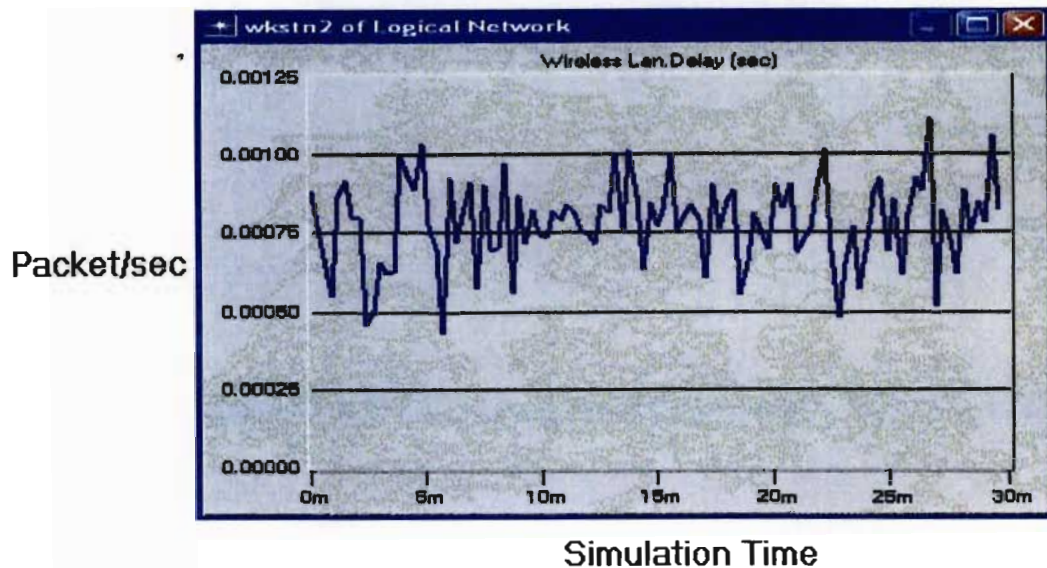


Figure 4.21 Delays in WLAN peer-to-peer communication

4.6.2 Simulation results of WLAN with wireless and wireline

Figures 4.22 and 4.23 describe the simulation results between two Remote sites of WLAN connection with Ftp and HTTP servers, using file transfer protocol to download files and browsing the web sites. The average rate of transmission and reception of the packets between the two sites was 0.850 Packet/sec for downloading files. We observe from the graph in Figure 4.23 that the average rate of transmission and reception of packets between the two sites was 6.5 Packet/sec for browsing the web sites. Collision, packet loss, and queue delay will be discussed in section 4.6.2.1.

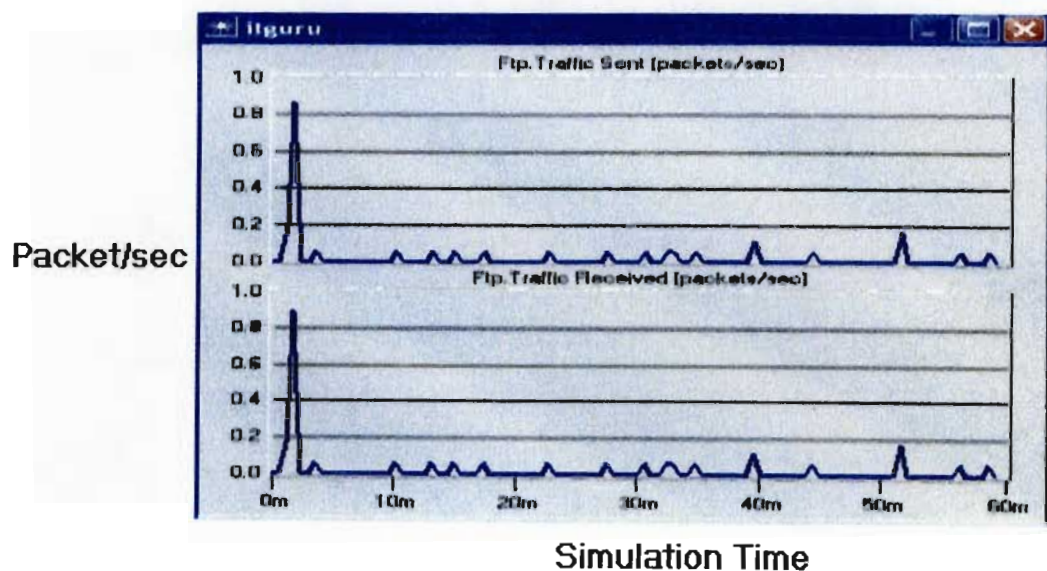


Figure 4.22 traffic between wireless and wired network with Ftp

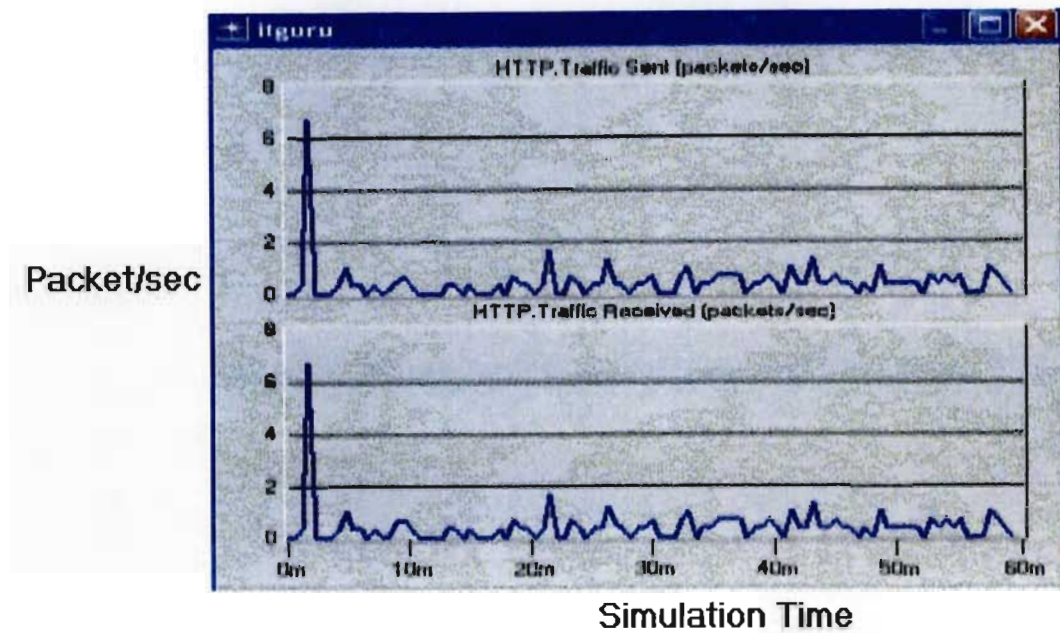


Figure 4.23 traffic between wireless and wired network with HTTP

4.6.2.1 Collision, packet loss, and queue delay with wireless and wireline

We are looking at collision, packet loss, and queue delay to study the performance of WLAN during the simulation hour. Through the traffic hour between two Remote sites of WLAN connected with Ftp and HTTP servers, using file transfer protocol to download files and browsing the web sites as discussed in section 4.6.2, From the results shown in figures 4.22 and 4.23, there is no collision and packet loss during the simulation traffic hour. Table 4.11 shows the delay between the sites and the remote servers (FTP and HTTP). It is observed from the table that the maximum delay (0.00264 ms) occurred between Site2 and remote servers (HTTP) and the minimum delay (0.00568 ms) occurred between Site1 and the remote server (FTP). These results of delay between the two sites are acceptable and better when compared to peer-to-peer connection delay results as discussed in section 4.6.1.1. The target of study in section 4.6 was to examine the performance of WLAN through peer-to-peer connection and integration between wireless and wireline using collision, packet loss, and queue delay as key parameters to analyze the simulation traffic results. The study found that the performance of WLAN with peer-to-peer communication shows a weak performance based on 0.0555 of packet loss ratio, and 0.00115 second of delay in the queue. The study also found the performance of WLAN and integration between wireless and wireline acceptable based on zero packet loss ratio, and 0.000568 - 0.000576 ms of low delay in the queue.

Table 4.11 Queue delay simulation results between wireless and wireline network

Delay		Value	
From	To	Maximum	Minimum
Site1	Remote Servers (FTP)	0.0022 ms	0.000568 ms
Site 2	Remote Server (HTTP)	0.00264 ms	0.000576 ms

4.7 Summary

In this chapter, we discussed the performance of WAN between Asmara and Massawa using measured traffic of single and dual link by highlighting three typical days and daily busy traffic hours from August to September 2005. The study found that the performance of WAN with both single and dual links shows a weak performance based on 0.00912 high packet loss ratio and 1.336 second high delay in the queue for single link, and 0.00418 high packet loss ratio and 0.336 second high delay in the queue for dual link.

In the second part of the chapter, we used OPNET Modeler program to simulate the performance of LAN. In order to select the best connection of LAN, the study found the performance of LAN with hub connection showed a weak performance based on 0.075 of high collision ratio, 0.0186-0.0230 Packet/sec packet loss ratio, and 0.328- 428 ms delay in the queue. The study also found the performance of LAN with switch connection records a better performance based on zero collision, zero packet loss ratio, and 0.0386-0.0480 ms delay in the queue as compared to the hub connection result. Finally, the study found that the performance of LAN with router connection records a good performance based on zero collision, zero packet loss ratio, and 0.0176 – 0.0200 ms delay in the queue compared to the switch connection result. From the comparison results between the three connections, the study found the performance of LAN with router (government networks) connection to record a better performance than the other connections through its minimum delay in the queue, zero packet loss, and zero collision.

Finally, we discussed the performance of WLAN using an OPNET Modeler program to simulate the performance of WLAN. The target of study was to examine the performance of WLAN through peer-to-peer connection and integration between wireless and wireline. The study found that the performance of WLAN with peer-to-peer communication showed a weak performance based on 0.0555 packet loss ratio and 0.00115 second of delay in the queue delay. The study also

found the performance of WLAN as integrated between wireless and wireline was acceptable based on zero collision, zero packet loss ratio, and 0.000568 - 0.000576 ms delay in the queue.

Chapter Five

Performance of VoIP in the Eritrean Environment

5.1 Introduction

In chapter four we discussed the performance of the networks through data communication using measured traffic of single and dual links for the WAN and the simulation traffic for the LAN and the WLAN. In this chapter we will look at the performance of VoIP in the Eritrean environment using an OPNET Modeler program to simulate the performance of VoIP via delay and packet loss parameters to calculate R-value using E-Model for the quality of service (QoS). The details of E-Model are described in section 3.4.1 in chapter 3. In this chapter, will first discuss the performance of VoIP with voice using the WAN between the Asmara LAN and the Massawa LAN with 28 kbps, 33 kbps, 64 kbps, and 128 kbps to determine the cheapest and highest quality acceptable link that can support the speech coder with most calls. The second goal of the study is to determine the best speech coder with 256 kbps through the performance of VoIP with voice and data using the total traffic of WAN between the Asmara LANs and the Massawa LANs. One of the key components of the E-Model and voice quality is latency. The simulation results and discussion of VoIP performance are described in sections 5.2 and 5.3

5.1.1 Latency

Latency is one of the most crucial limitations that affect voice over packet systems. This is because when people have a conversation they interact with each other. This interactivity dictates the delay by determining the amount of time that a sender will wait before assuming that his / her last response was not heard and repeats his/her words. If the delay is too high, the receiver will have just received the first response and will be responding at the same time. This creates confusion and unclear conversation and thus poor interactivity [73]. Both TIA116 [71] and Hersent [72] address the topic of delay. Interestingly enough, what is considered unacceptable varies significantly. For example, Hersent [73] noted an acceptable delay of up to 250 ms. This is due to the fact that when a person finishes speaking, there is an approximate 200 ms break before the other person starts speaking. This is called turn taking [71]. When an extra 150 ms is added, the turn taking rules begin to fail and the conversation rhythm must change. Figure 5.1 shows the impact that delay has on the voice quality according to the E-Model. The algorithms to complete these calculations are in ITU Recommendation [70].

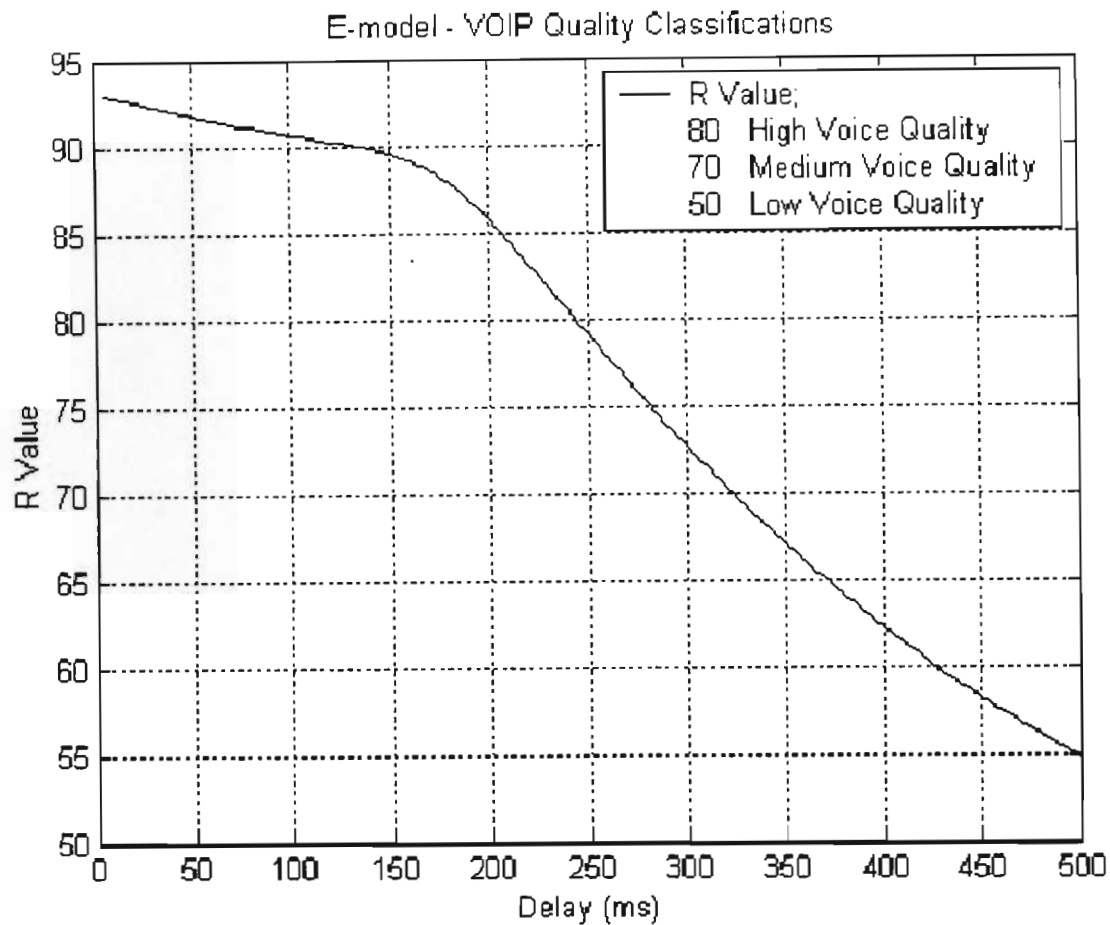


Figure 5.1 E-model- VoIP Quality Classification [71]

5.1.2 Packet Loss

Packet loss is unavoidable on the Internet. Reliable protocols like TCP cannot be used due to the fact that retransmission can take multiple round trip times and voice transmission typically cannot withstand the delay [74]. Packet loss may or may not be related to latency. If packet loss is caused by bit errors, these losses would be unrelated to latency. However, if a maximum delay for a voice stream is exceeded, typically those packets are dropped. In this case, there exists a tradeoff between delay and packet loss. If the maximum delay allowed by a voice stream is increased, packet loss will decrease.

5.2 Performance analysis of VoIP between Asmara LAN and Massawa LAN

In this section, we study the performance of VoIP with voice through a WAN between the Asmara LAN and the Massawa LAN using 28 kbps, 33 kbps, 64 kbps, and 128 kbps capacity links with G723.1, G729A, and G711 as described in sections 5.2.1, 5.2.2, and 5.2.3. The purpose of the study is to determine the cheapest high quality link that can support the speech coder with the most calls. Delay and packet loss are the key factors to measure the quality of service and the performance of VoIP.

5.2.1 G.723.1.

G723.1 is a Dual rate speech coder that performs compression and decompression of 8 kHz speech signals. It encodes 16-bit PCM samples into 16-bit code-words yielding 10 or 12 code-words per 240 sample frames for the 5.3 Kbps and 6.3 Kbps channels respectively. G.723.1 is used for compressing the speech or other audio signal components of multimedia services at a very low bit rate as part of the overall H.324 family of standards. This coder was optimised to represent high quality speech at the above rates using a limited amount of complexity. It encodes speech or other audio signals in frames using linear predictive analysis-by-synthesis coding. The excitation signal for the high rate coder is Multipulse Maximum Likelihood Quantization (MP-MLQ) and for the low rate coder it is Algebraic-Code-Excited Linear-Prediction (ACELP) [70].

5.2.2 G729A

The G.729A ITU-T standard for voice coding compresses 8 kHz linear audio signals and encodes them for transmission at 8 Kbps with reduced complexity [71].

5.2.3 G711

The G.711 standard uses pulse code modulation (PCM) to compress, decompress, encode, and decode analog speech, which can then be transmitted and received as binary data [71].

5.2.4 Performance of VoIP with 28kbps

Table 5.1 describes the simulation results of delay and packet loss with 28 kbps capacity link for three coders used for VoIP. We can see from the table that G729A has better results than the other speech coders as shown in table 5.1. It can be seen from the table that the delay in all speech coders (G723.1, G729A, and G711) is above 250 ms, which means that the result of delay is unacceptable based on the limitation of delay (250 ms) as noted by Hersent [73].

Table 5.1 the simulation result between Asmara and Massawa
with 28 kbps capacity link

Coder	Delay (ms)	Packet Loss Ratio
G723.1	529	0.202
G729A	430	0.168
G711	498	0.187

From the above results, the simulation determines the corresponding R-values as shown in table 5.2 and we can observe from the table that the results of R-values are less than 70 for all the speech coders. That means the quality of the voice is poor for all the speech coders (G723.1, G729A, and G711) and is unacceptable compared to the ITU and ITA category for R-value level (see figure 5.1). We can also see from table 5.2 that G723.1 records the worst quality compared to the other speech coders. The study found the performance of VoIP between the Asmara LAN and the Massawa LAN with 28 kbps to be unacceptable based on above 400 ms of high delay and high packet loss ratio with all speech coders and poor quality of service based on R-value less than 70 with all speech coders.

Table 5.2 R-Value results with 28 kbps

Coder	R-Value	Number of Calls	Delay (ms)
G723.1	52	5.2	529
G729A	62.5	3.5	430
G711	56.5	0.4	498

5.2. 5 Performance of VoIP with 33 kbps

Table 5.3 introduces the delay and packet loss with 33 kbps capacity link for three coders used for VoIP. The table shows that the G711 encoder records a good performance based on 125 ms of delay and 0.0013 low packet loss ratio. We also observe from the table that the delay for all speech coders is below 200 ms, which means the result of delay with all speech coders is acceptable.

Table 5.3 the simulation result between Asmara and Massawa
with 33 kbps capacity link

Coder	Delay (ms)	Packet Loss Ratio
G723.1	172	0.003
G729A	142	0.002
G711	125	0.0013

Using the above results, the simulation determines the corresponding R-values as shown in table 5.4, we can observe from the table that the results of the R-values are greater than 80 for all the speech coders. This means the quality of the voice is high for all the speech coders (G723.1, G729A, and G711) and is acceptable (see figure 5.1). We can also see, from Table 5.4, that the G711 speech coder records an excellent result of voice quality of 89.2 with 0.5 call, however, the G723.1 speech coder gives a better number of calls with a very good voice quality over the simulation period as shown in table 5.4. The study found that the performance of VoIP between the **Asmara** LAN and the Massawa LAN with 33 kbps was acceptable based on below 200 ms of **low delay** and low packet loss ratio with all speech coders and high quality of service with all speech coders based on R-value results (greater than 80) with all speech coders.

Table 5.4 R-Value results with 33 kbps

Coder	R-Value	Number of Calls	Delay (ms)
G723.1	87.5	6.2	172
G729A	88.2	4.1	142
G711	89.2	0.5	125

5.2. 6 Performance of VoIP with 64 kbps

Table 5.5 describes the simulation results of delay and packet loss with 64 kbps capacity link for the three coders used for VoIP. The table shows, G711 record a good performance based on 120 ms of minimum delay and 0.00122 low packet loss ratio. It can also be seen from the table that the delay in all speech coders (G723.1, G729A, and G711) is below 200 ms, which means the result of delay with all speech coders is acceptable. As expected, the delay values decrease for each coder as the link capacity increases.

Table 5.5 The simulation result between Asmara and Massawa
with 64 kbps capacity link

Coder	Delay (ms)	Packet Loss (%)
G723.1	155	0.0025
G729A	130	0.0016
G711	120	0.00122

From the above results, the simulation determines the corresponding R-values as shown in table 5.6. We can observe from the table that the results of the R-values are greater than 80 for all speech coders, which means the quality of the voice is high with all speech coders (G723.1, G729A, and G711) and acceptable compared to the ITU and ITA category level of R-value (see figure 5.1). We can also see from table 5.6 that the G711 speech coder recorded an excellent result of voice quality of 89.4 with 1 call. However, the G723.1 speech coder gives a better number of calls with a high voice quality over the simulation period as shown in table 5.6. The study found the performance of VoIP between the Asmara LAN and the Massawa LAN with 64 kbps acceptable based on below 200 ms of low delay and low packet loss ratio for all speech coder's and high quality of service based on R-value greater than 80.

Table 5.6 R-Value results with 64 kbps

Coder	R-Value	Number of Calls	Delay (ms)
G723.1	88	12	155
G729A	88.7	8	130
G711	89.4	1	120

5.2. 7 Performance of VoIP with 128 kbps

Table 5.7 introduces the delay and packet loss with 128 kbps capacity link for the three coders used for VoIP. The table shows that the G729A records a good performance based on 110 ms of low delay and 0.00118 low packet loss ratio. We also observe from the table that the delay for all the speech coders is below 150 ms, which means the result of delay is acceptable for them all. As expected, the delay values drop for each coder as the link capacity increases.

Table 5.7 The simulation result between Asmara and Massawa with 128 kbps capacity link

Coder	Delay (ms)	Packet Loss Ratio
G723.1	122	0.00128
G729A	110	0.00115
G711	115	0.00118

From the above results, the simulation determines the corresponding R-values as shown in table 5.8. We can observe from the table that the results of the R-values are greater than 80 for all speech coders, which means that the quality of voice is high with all speech coders (G723.1, G729A, and G711) and is acceptable compared to the ITU and the ITA category level of R-value (see figure 5.1). We can also see from table 5.8 that the G729A speech coder records an excellent result of voice quality of 91.3 with 16 calls, however, the G723.1 speech coder gives a better number of calls with a high voice quality over the simulation period as shown in table 5.8. The study found the performance of VoIP between the Asmara LAN and the Massawa LAN with 128 kbps acceptable based on below 150 ms of low delay and low packet loss ratio with all speech coders and high quality of service based on R-value greater than 80 for all speech coders.

Table 5.8 R-Value results with 128 kbps

Coder	R-Value	Number of Calls	Delay (ms)
G723.1	89	24.1	122
G729A	91.3	16	110
G711	91	2	115

From the simulation results of the VoIP performance under Eritrean environment, using different capacity links and three speech coders (G723.1, G729A, and G711), the study found that the 33 kbps to 128 kbps links all give R values that exceed 87; hence, the speech quality is very good in each case (See figure 5.1). However, we note that while G711 gives the highest R value for each link capacity, it supports the lowest number of calls. On the other hand, the G723.1 encoder supports about 6.2 calls with 33 kbps, 12 calls with 64 kbps, and 24.1 calls with 128 kbps. Based on the performance (delay and packet loss), quality of service (R-value), and the cost, the study concludes that the 128 kbps link with the G723.1 encoder has the most calls (24.1). The study also found the 28 kbps link gives R-value lower than 70, hence it is unacceptable.

5.3 Performance analysis of VoIP between Asmara LANs and Massawa LANs

In chapter 4, we studied the performance of the WAN between the Asmara LAN and the Massawa LAN using measured traffic. In this section, we are looking at the performance of VoIP with voice and data through different cases as shown in Table 5.9, using the total traffic of Asmara LANs and Massawa LANs with 256 kbps WAN capacity link to determine the best speech coder with 256 kbps. Delay and packet loss are the key factors to measure the quality of service and the performance of VoIP. The simulation results of each case are described in sections 5.3.1, 5.3.2, 5.3.3, 5.3.4, and 5.3.5. Based on the traffic measurements in chapter 4, the maximum design load of 244 kbps has been determined (during the busy hour) between the Asmara and Massawa WAN.

Table 5.9 VoIP performance study with different cases

Case No	Total Load	Voice	Data
1	244 kbps	10%	90%
2	244 kbps	20%	80%
3	244 kbps	30%	70%
4	244 kbps	40%	60%
5	244 kbps	50%	50%

5.3.1 Case 1 (Performance of VoIP with 10 % Voice and 90 % Data)

Table 5.10 introduces the delay and packet loss with 256 kbps capacity link for the three coders used for VoIP. The table shows that the G729A encoder records a good performance based on 192 ms of low delay and 0.00618 low packet loss ratio. We also observe from the table that the delay for all speech coders is below 250 ms, which means the result of delay for all speech coders is acceptable.

Table 5.10 the simulation result of 10 % voice and 90 % data

Coder	Load (kbps)	Delay (ms)	Packet Loss Ratio
G723.1	244	200	0.00675
G729A	244	192	0.00618
G711	244	196	0.00622

Using the above results, the simulation determines the corresponding R-values shown in table 5.11, and we can observe from the table that the results of the R-values are greater than 80 for all speech coders. That means the quality of voice and data is high with all speech coders (G723.1, G729A, and G711) and is acceptable compared to ITU and ITA category level of R-value (see figure 5.1). We can also see from table 5.11 that the G729A speech coder recorded an excellent result of 86.1 with 3.05 calls. However, the G723.1 speech coder gives a better number of calls with a higher quality over the simulation period as shown in table 5.11. The study found that the performance of VoIP is acceptable based on delay values below 250 ms and low packet loss ratio, with all speech coders, as well as high quality of service based on R-value greater than 80 with all speech coders.

Table 5.11 R-Value result of 10 % voice and 90 % data

Coder	R-Value	Number of Calls	Delay (ms)
G723.1	84.2	4.6	200
G729A	86.1	3.05	192
G711	85.4	0.38	196

5.3.2 Case 2 (Performance of VoIP with 20 % Voice and 80 % Data)

Table 5.12 introduces the delay and packet loss with a 256 kbps capacity link for the three coders used for VoIP. The table shows that the G729A records a good performance based on 223 ms of low delay and 0.00746 low packet loss ratio. We also observe from the table that the delay for all speech coders is below 250 ms, which means that the result of delay for all speech coders is acceptable'.

Table 5.12 the simulation result of 20 % voice and 80 % data

Coder	Load (kbps)	Delay (ms)	Packet Loss Ratio
G723.1	244	230	0.0086
G729A	244	223	0.00746
G711	244	225	0.00758

From the above results, the simulation determines the corresponding R-values as shown in table 5.13. We can thus observe from the table that the results of the R-values are greater than 80 with all speech coders. This means that the quality of voice and data is high for all speech coders (G723.1, G729A, and G711) and is acceptable compared to the ITU and ITA category level of R-value (see figure 5.1). We can also see from table 5.13 that the G729A speech coder recorded an excellent result of 84.7 with 6.1 calls. However, the G723.1 speech coder gives a better number of calls with a higher quality over the simulation period as shown in table 5.13. The study found that the performance of VoIP is acceptable based on delay values below 250 ms and low packet loss ratio with all speech coders. Also, we find a high quality of service based on R-value results with all speech coders as shown in table 5.13, as compared to Gardner [64] with an R-value result of 76 with G729A coder.

Table 5.13 R-Value result of 20 % voice and 80 % data

Coder	R-Value	Number of Calls	Delay (ms)
G723.1	83.6	9.2	230
G729A	84.7	6.1	223
G711	84	0.76	225

5.3.3 Case 3 (Performance of VoIP with 30 % Voice and 70 % Data)

Table 5.14 introduces the delay and packet loss with 256 kbps capacity link for the three coders used for VoIP. The table shows that the G729A encoder records a good performance based on 238 ms of low delay and 0.00891 low packet loss ratio. We also observe from the table that the delay for all speech coders is below 250 ms, which means that the result of delay for all speech coders is acceptable.

Table 5.14 the simulation result of 30 % voice and 70 % data

Coder	Load (kbps)	Delay (ms)	Packet Loss Ratio
G723.1	244	242	0.00956
G729A	244	238	0.00891
G711	244	240	0.00923

Using the above results, the simulation determines the corresponding R-values as shown in table 5.15 and we can observe from the table that the result of the R-values are greater than 70 for all speech coders. This means that the quality of voice and data is medium for all speech coders (G723.1, G729A, and G711) and is acceptable compared to the ITU and ITA category level of R-value (see figure 5.1). We can also see from table 5.15 that the G729A speech coder recorded an excellent result of 82.3 with 9.1 calls. However, the G723.1 speech coder gives a better number of calls with a higher quality over the simulation period as shown in table 5.15. The study found that the performance of VoIP is acceptable based on delay values below 250 ms and low packet loss ratio with all speech coders. We also observe a high quality of service based on R-value results with all speech coders as shown in table 5.15 (compared to Gardner [64] R-value result of 76 with G729A coder.

Table 5.15 R-Value result of 30 % voice and 70 % data

Coder	R-Value	Number of Calls	Delay (ms)
G723.1	80	13.8	242
G729A	82.3	9.1	238
G711	80.6	1.1	240

5.3.4 Case 4 (Performance of VoIP with 40 % Voice and 60 % Data)

Table 5.16 introduces the delay and packet loss with 256 kbps capacity link for the three coders used for VoIP. We can see from the table that the G723.1 has better results compared to the results of other speech coders as shown in table 5.16. We also observe from the table that the delay for all speech coders is above 250 ms.

Table 5.16 the simulation result of 40 % voice and 60 % data

Coder	Load (kbps)	Delay (ms)	Packet Loss Ratio
G723.1	244	276	0.0108
G729A	244	289	0.0116
G711	244	283	0.0111

From the above results, the simulation determines the corresponding R-values as shown in table 5.17 and we can observe from the table that the result of the R-values are greater than 70 for all speech coders. This means that the quality of voice and data is medium for all speech coders (G723.1, G729A, and G711) and is acceptable compared to the ITU and ITA category level of R-value (see figure 5.1). We can also see from table 5.17 that the G723.1 speech coder recorded a good result of 77.2 with a high number of 18.4 calls. The study found that the performance of VoIP is acceptable based on medium quality of service for all speech coders as shown in table 5.17.

Table 5.17 R-Value result of 40 % voice and 60 % data

Coder	R-Value	Number of Calls	Delay (ms)
G723.1	77.2	18.4	276
G729A	73.4	12.2	289
G711	75.6	1.5	283

5.3.5 Case 5 (Performance of VoIP with 50 % Voice and 50 % Data)

Table 5.18 shows the delay and packet loss with 256 kbps capacity link for the three coders used for VoIP. We can see from the table that the G711 encoder has better results compared to the

results of other speech coders as shown in table 5.18. We also observe from the table that the delay for all the speech coders is above 250 ms.

Table 5.18 the simulation result of 50 % voice and 50 % data

Coder	Load (kbps)	Delay (ms)	Packet Loss Ratio
G723.1	244	320	0.0132
G729A	244	298	0.126
G711	244	286	0.018

Using the above results, the simulation determines the corresponding R-values as shown in table 5.19. We can observe from the table that the results of the R-values are less than 70 for all speech coders which means that the quality of voice and data is low for all speech coders (G723.1, G729A, and G711) and is unacceptable compared to the ITU and ITA category level of R-value (see figure 5.1). We can see also from table 5.19 that the G711 speech coder recorded a low result of 67.8 with a low number of 1.9 calls. However, the G723.1 speech coder gives a better number of calls with low quality over the simulation period as shown in table 5.19. The study found that the performance of VoIP as unacceptable based on above 250 ms of delay and high packet loss ratio for all speech coders and poor quality of service based on R-value less than 70 for all speech coders as shown in table 5.19. In order to determine the best speech coder with 256 kbps, the study found G729A achieves the quality target of study based on the quality of service (R-value grater **than 80**) with Case 1, Case 2, and Case3. The study also found that the R-value decreases **with** the case of 50% voice and 50 % data and gave a low quality of service as shown in table 5.19

Table 5.19 R-Value result of 50 % voice and 50 % data

Coder	R-Value	Number of Calls	Delay (ms)
G723.1	65.3	23	320
G729A	66.2	15.2	298
G711	67.8	1.9	286

5.4 Increased WAN Traffic

In section 5.3, we have assumed an offered traffic of 244 kbps during the peak hours. The link used has a capacity of 256 kbps, giving $\rho = 0.953$. This is high. With an assumed annual traffic growth rate of 10 %, after 1 year, the traffic offered would be 270.4 kbps, which is greater than the channel capacity. Normally, a value of $\rho \sim 0.6-0.7$ is recommended. Thus we should have a link of 512 kbps up to year 4 when the traffic is projected to be 375 kbps giving $\rho = 0.6977$. For years 5-10 the traffic will range 395- 633 kbps, and therefore the link capacity should be raised to 1 Mbps ($\rho \sim 0.39$ for year 5 and $\rho \sim 0.63$ for year 10). In this section, we simulate the performance of the link for a capacity of 512 kbps for mixed traffic ratios of 40 % voice and 60 % data and 50 % voice and 50 % data. The simulation results of Case 6 and Case 7 are described in sections 5.4.1 and 5.4.2.

5.4.1 Case 6 (Performance of VoIP with 40 % Voice and 60 % Data via 512 kbps)

Table 5.20 introduces the delay and packet loss with 512 kbps capacity link for the three coders used for VoIP. The table shows that the G729A recorded a good performance based on 185 ms of low delay and 0.00361 low packet loss ratio. We also observe from the table that the delay for all speech coders is below 250 ms, which means that the result of delay for all speech coders is acceptable'.

Table 5.20 the simulation result of 40 % voice and 60 % data

Coder	Load (kbps)	Delay (ms)	Packet Loss Ratio
G723.1	395	217	0.00721
G729A	395	185	0.00361
G711	395	205	0.00662

Using the above results, the simulation determines the corresponding R-values as shown in table 5.21. We can observe from the table that the results of the R-values are greater than 80 for all speech coders which means that the quality of voice and data is high for all speech coders (G723.1, G729A, and G711) and is acceptable compared to the ITU and ITA category level of R-value (see figure 5.1). We can also see from table 5.21 that G729A speech coder recorded an

excellent result of 86.5 with 19.7 calls. However, the G723.1 speech coder gives a better number of calls with a higher quality over the simulation period as shown in table 5.21. The study found that the performance of VoIP is acceptable based on delay values below 250 ms and low packet loss ratio for all speech coders; in addition, we have high quality of service based on R-value greater than 80 for all speech coders.

Table 5.21 R-Value result of 40 % voice and 60 % data

Coder	R-Value	Number of Calls	Delay (ms)
G723.1	83.1	29.8	217
G729A	86.5	19.7	185
G711	84.3	2.4	205

5.4.2 Case 7 (Performance of VoIP with 50 % Voice and 50 % Data via 512 kbps)

Table 5.22 introduces the delay and packet loss with 512 kbps capacity link for the three coders used for VoIP. The table shows that the G729A records a good performance based on 229 ms of low delay and 0.0084 low packet loss ratio. We also observe from the table that the delay for all speech coders is below 250 ms, which means this result is acceptable for all speech coders .

Table 5.22 the simulation result of 50 % voice and 50 % data

Coder	Load (kbps)	Delay (ms)	Packet Loss Ratio
G723.1	395	248	0.00983
G729A	395	229	0.0084
G711	395	235	0.00863

From the above results, the simulation determines the corresponding R-values as shown in table 5.23. We can observe from the table that the results of the R-values are greater than 70 for all speech coders. This means that the quality of voice and data is high for all speech coders (G723.1, G729A, and G711) and is acceptable compared to the ITU and ITA category level of R-value (see figure 5.1). We can also see from table 5.23 that the G729A speech coder recorded an excellent

result of 82.5 with 24.6 calls. However, the G723.1 speech coder gives a better number of calls with a medium quality over the simulation period as shown in table 5.23. The study found that the performance of VoIP is acceptable based on delay values below 250 ms and low packet loss ratio with all speech coders, and high quality of service based on R-value greater than 70 for all speech coders.

Table 5.23 R-Value result of 50 % voice and 50 % data

Coder	R-Value	Number of Calls	Delay (ms)
G723.1	79	37.2	248
G729A	82.2	24.6	229
G711	81	3	235

5. 5 Summary

In this Chapter, we discussed the performance of VoIP with voice through a WAN between the Asmara LAN and the Massawa LAN using 28 kbps, 33 kbps, 64 kbps, and 128 kbps capacity links with G723.1, G729A, and G711. The study found the performance of VoIP between the Asmara LAN and the Massawa LAN with 28 kbps to be unacceptable based on high delay above 400 ms and high packet loss ratio for all speech coders and low quality of service with speech coders based on R-value less than 70 with all speech coders. The study also found the performance of VoIP between Asmara LAN and Massawa LAN with 33 kbps to 128 kbps acceptable based on low delay (below 200 ms) and low packet loss ratio for all speech coders and high quality of service based on R-value results greater than 80 for all speech coders. Finally the study found the 128 kbps links with the G723.1 had the most calls (24.1). The study also found that the 28 kbps link gives a R-value lower then 70, hence it is unacceptable.

We also studied the performance of VoIP with voice and data through different cases as shown in table 5.9, using the total traffic of the Asmara LANs and the Massawa LANs with 256 kbps capacity link of WAN to determine the best speech coder with 256 kbps. The study found that the G729A coder achieves the target of study based on the quality of service (R-value greater than 80) with Case 1, Case 2, and Case 3. The study also found that the R-value decreases with 50% voice and 50 % data and gave low quality of service. Based on the simulation result of increasing

the traffic to 395 kbps between Asmara LANs and Massawa LANs with 512 kbps capacity links of WAN, the study found that the G729A coder achieves the target of study based on the quality of service (R-value greater than 80) with Case 6 and Case 7. On the other hand, the G723.1 coder gives 37.2 calls for an R value of 79 (a delay of 248 ms) for the same traffic ratio. Thus, for the Eritrea network, the G723.1 encoder would be preferred for 256 kbps and 512 kbps trunks, since it offers the highest number of calls for the loading ratio considered, while also offering good quality of service.

Chapter Six

Conclusion and Recommendation

A review of the technologies used in LAN, WAN, and WLAN has been given. The different topologies in LAN and WAN, the different type of LANs such as Ethernet, Token Ring, and Fiber Distributed Data Interface, different devices of LAN connection to link LANs together, and different types of cable, were discussed. The different types of capacity links (such as ISDN, ADSL, ATM, X.25, and Frame Relay) were also discussed. The WLAN technology (infrared, spread spectrum, and narrowband microwave), and standard technologies of WLAN, and WLAN in the market, were also discussed. Network protocols such as TCP, IP, WAP, WTP, and WTLS and network performance were also illustrated.

We discussed the tools and methods used to examine the measured traffic of the WAN between the Asmara LAN and the Massawa LAN using the Markovian M/M/1 queue model to determine the queue delay in single link, and M/M/2 queue model for the dual link delay. A JavaScript program was employed to compute the analytical parameters. OPNET Modeler simulation program is used to examine the performance of LAN and WLAN using different connection devices **for LAN** scenarios and two scenarios for WLAN peer-to-peer and integration of wireline and WLAN communication.

An **OPNET Modeler** simulation program was employed to examine the performance of VoIP **with voice through** WAN between the Asmara LAN and the Massawa LAN using 28 kbps, 33 kbps, 64 kbps, and 128 kbps capacity links with G723.1, G729A, and G711 speech coders. The **performance** of VoIP with different ratios of voice and data for total traffic of 244 kbps between Asmara and Massawa LANs via 256 kbps. WAN capacity link was simulated using G723.1, G729A, and G711 speech coders.

We discuss the performance of WAN between the Asmara LAN and the Massawa LAN using measured traffic of single and dual link by highlighting three typical days and daily busy traffic hours from August to September 2005. The study found that the **performance** of the WAN with both single and dual links showed weak performance based on 0.00912 high packet loss ratio and

1.336 second high delay in the queue for single and 0.00418 high packet loss ratio and 0.336 second high delay in the queue for dual link.

We used OPNET Modeler program to simulate the performance of the LAN. In order to select the best connection of LAN, the study found that the performance of LAN with hub connection shows a weak performance based on a collision ratio of 0.075, a high packet loss ratio of 0.0186-0.0230 Packet/sec, and a high delay of 0.328- 428 ms in the queue. The study also found that the performance of the LAN with switch connection records gave a better performance based on zero collision, zero packet loss ratio, and 0.0386-0.0480 ms delay in the queue, compared to the hub connection results. Finally the study found that the performance of the LAN with router connection records was better still, based on zero collision, zero packet loss ratio, and 0.0176 – 0.0200 ms delay in the queue. From the comparison of results between the three connections, the study found the LAN with router (government networks) connection recorded a better performance than the other connections through its low delay in the queue, zero packet loss, and zero collision.

We discuss also the performance of the WLAN using OPNET Modeler program to simulate the performance of the WLAN. The target of study was to examine the performance of the WLAN through peer-to-peer connection and integration between wireless and wireline. The study found that the performance of the WLAN with peer-to-peer communication was weak, based on 0.0555 packet loss ratio, and a 0.00115 second queue delay. The study also found that the performance of WLAN integrated between wireless and wireline was also acceptable based on zero collision, zero packet loss ratio, and a low 0.000568 - 0.000576 ms delay in the queue.

We discuss the performance of VoIP with voice through a WAN between the Asmara LAN and the Massawa LAN using 28 kbps, 33 kbps, 64 kbps, and 128 kbps capacity links with G723.1, G729A, and G711 coders. The study found that the performance of VoIP between the Asmara LAN and the Massawa LAN with 28 kbps was unacceptable based on high delay above 400 ms and high packet loss ratio for all speech coders, and a low quality of service based on R-value less than 70 for all three speech coders. The study also found the performance of VoIP between the Asmara LAN and the Massawa LAN with 33 kbps to 128 kbps was acceptable based on delay below 200 ms and low packet loss ratio and high quality of service based on R-value results (greater than 80) for all speech coders. The study found the 28 kbps link gave an R-value lower

then 70, hence it is unacceptable. On the other hand, the G723.1 encoder gives the highest number of calls with R values greater than 80 for link capacities of 33 kbps to 128 kbps.

Finally, we discuss the performance of VoIP with voice and data through different cases using the load of 244 kbps between the Asmara and the Massawa LANs, with a 256 kbps capacity link of a WAN. The study found that the G729A coder achieves the target of study based on the quality of service (R-value grater than 80) with Case 1, Case 2, and Case 3; on the other hand, the G723.1 coder gives the highest number of calls. The study also found that the R-value decreases with the case of 50% voice and 50 % data and gave a low quality of service. Based on the simulation result of increasing the traffic 395 kbps between the Asmara LANs and the Massawa LANs with 512 kbps capacity links of WAN, the study found that the G729A achieves the target of study based on the quality of service (R-value grater than 80) with Case 6 and Case 7. However, the G723.1 encoder would still be recommended in both cases when both number of calls and R value are taken into considration.

Recommendation

- Based on the weak performance results obtained from measured traffic between Asmara LAN and Massawa LAN, we recommend that the government of Eritrea updates the capacity link from 19.2 kbps for single link, and 9.6 kbps for dual link, to higher capacity link such as: ISDN and ADSL, to improve the performance of WAN between the LAN of the two cities.
- Based on the simulation results obtained from the performance of LAN with different connections (hub, switch, and router), we recommend that the Eritrean government avoids the use of hub connections with the LANs.
- Based on the poor quality of service obtained from the performance of VoIP with 28 kbps, we recommend that the Eritrean government avoids the use of 28 kbps channels.
- From the simulation, VoIP is achievable in Eritrea with a G723.1 encoder for link capacities exceeding 33 kbps. However, with an annual traffic growth of 10%, a 512 kbps link should be installed between Asmara and Massawa to support good quality VoIP.

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Appendix A

**/Single Link (VB Script)*

**/ this function is used to calculate the average of the arrival packet (λ) and the average of the service rate (μ) with a single capacity link (19200 bps)*

Private Sub SingleLink_Click ()

Dim sConnect As String

Dim sSQL As String

Dim dfwConn As ADODB.Connection

sSQL = "select ArrPacket, Date, PackSize,(ArrPacket)/(8*3600) as Lambda,
(Singlelink)/(PackSize*8) as mu from simdatab WHERE Singlelink = "" & datlink & """

‘ open connection

Set dfwConn = New Connection

dfwConn.Open sConnect

‘ create a recordset using the provided collection

Set datPrimaryRS = New Recordset

datPrimaryRS.CursorLocation = adUseClient

datPrimaryRS.Open sSQL, dfwConn, adOpenForwardOnly, adLockReadOnly

Set MSHFlexGrid1.DataSource = datPrimaryRS

With MSHFlexGrid1

.Redraw = False

‘ set grid's column widths

.ColWidth (0) = -1

.ColWidth (1) = -1

.ColWidth (2) = -1

.ColWidth (3) = -1

```
.ColWidth (4) = -1
```

```
‘ set grid’s style
```

```
.AllowBigSelection = True
```

```
.FillStyle = flexFillRepeat
```

```
‘ make header bold
```

```
.Row = 0
```

```
.Col = 0
```

```
.RowSel = .FixedRows - 1
```

```
.ColSel = .Cols - 1
```

```
.CellFontBold = True
```

```
.AllowBigSelection = False
```

```
.FillStyle = flexFillSingle
```

```
.Redraw = True
```

```
End With
```

```
End Sub
```

```
‘ Close function
```

```
Private Sub Close_Click ()
```

```
Unload Me
```

```
End Sub
```

**/Dual Link (VB Script)*

**/ this function is used to calculate the average of the arrival packet (λ) and
the average of the service rate (μ) with a dual capacity link (9600 bps)*

Private Sub DualLink_Click ()

Dim sConnect As String

Dim sSQL As String

Dim dfwConn As ADODB.Connection

sSQL = "select ArrPacket, Date, PackSize, (ArrPacket)/(8*3600) as Lambda,
(Duallink)/ (PackSize*8) as mu from simdatab WHERE Duallink = "" & datlink & """

‘ open connection

Set dfwConn = New Connection

dfwConn.Open sConnect

‘ create a recordset using the provided collection

Set datPrimaryRS = New Recordset

datPrimaryRS.CursorLocation = adUseClient

datPrimaryRS.Open sSQL, dfwConn, adOpenForwardOnly, adLockReadOnly

Set MSHFlexGrid1.DataSource = datPrimaryRS

With MSHFlexGrid1

.Redraw = False

‘ set grid’s column widths

.ColWidth(0) = -1

.ColWidth(1) = -1

.ColWidth (2) = -1

```
.ColWidth (3) = -1
```

```
.ColWidth (4) = -1
```

```
‘ set grid’s style
```

```
.AllowBigSelection = True
```

```
.FillStyle = flexFillRepeat
```

```
‘ make header bold
```

```
.Row = 0
```

```
.Col = 0
```

```
.RowSel = .FixedRows - 1
```

```
.ColSel = .Cols - 1
```

```
.CellFontBold = True
```

```
.AllowBigSelection = False
```

```
.FillStyle = flexFillSingle
```

```
.Redraw = True
```

```
End With
```

```
End Sub
```

Appendix B

M/M/1 and M/M/2 Markov Queue Model

The simplest single-server model having both input and service process described probabilistically is one with exponential inter arrival and service time (denoted by M/M/1) [62] [63] [64].

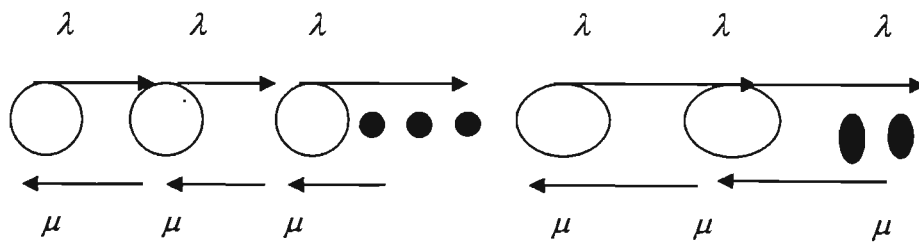
Exponential inter arrival density: $\lambda e^{-\lambda t}$ (1)

Exponential service time density: $\mu e^{-\mu t}$

We define the number of packets n in the system at any point in time as including the packets waiting in line plus those in service. Let $t = 0$ represent the system starting point in time, and we define:

$P_n(T) \equiv$ Probability that n packets are in the system at Time T . (2)

Actually $P_n(T)$ depends on the member of packets i in the system at Time 0, but this dependence is expressed in the notation. Let $h > 0$ be a very small interval of time. If there are $n > 0$ packets in the system at time $t + h$, then we consider only possibilities that there were $n-1$, n , or $n + 1$ packet at time t . Any other possibilities are of relatively insignificant probability.



State diagram for M/M/1 Queue

Consequently, for $n > 0$:

$$P_n(t+h) = P_n(t)[1 - \lambda h][1 - \mu h] + \quad (n \text{ Packet at } t, \text{ no packet came, no packet left})$$

$$\begin{aligned}
P_n(t)[\lambda h][\mu h] + & \quad (\text{n Packet at t, one packet come, one packet left}) \\
P_{n-1}(t)[\lambda h][1 - \mu h] + & \quad (\text{n - 1 packet at t, one packet come, no packet left}) \\
P_{n+1}(t)[1 - \lambda h][\mu h] & \quad (\text{n + 1 packet at t, no packet came, one packet left}) \\
P_0(t+h) = P_0(t)[1 - \lambda h] + & \quad (\text{no packet at t, no packet came}) \\
P_1(t)[1 - \lambda h][\mu h] & \quad (\text{one packet at t, no packet come, one packet left})
\end{aligned}$$

The above equation may be written in the following manner, where the terms with h^2 are omitted because they are relatively very small ($h \rightarrow 0$):

$$\begin{aligned}
\frac{P_n(t+h) - P_n(t)}{h} &= -(\lambda + \mu)P_n(t) + \lambda P_{n-1}(t) + \mu P_{n+1}(t) \\
\frac{P_0(t+h) - P_0(t)}{h} &= -\lambda P_0(t) + \mu P_1(t)
\end{aligned} \tag{3}$$

Because the value of h is small, the terms at the left side of (3) may be considered as derivative:

$$\frac{dP_n}{dt} = -(\lambda + \mu)P_n(t) + \lambda P_{n-1}(t) + \mu P_{n+1}(t) \quad \text{For } n > 0. \tag{4}$$

This expression is exact because the terms neglected in (3) becomes 0 as $h \rightarrow 0$:

$$\frac{dP_0}{dt} = -\lambda P_0(t) + \mu P_1(t) \quad \text{For } n = 0 \tag{5}$$

With moderately advanced analysis, it is possible to solve the system of linear different equations (4) and (5) for each $P_n(t)$. (To do so, we must also state the number of packets I in the system at time 0.) The result is called transient solution, since it depends directly on the value of t .

$$\rho = \frac{\lambda}{\mu} < 1 \quad (\text{traffic intensity assumption}) \quad (6)$$

We can find the equilibrium solution $P_n(t) = P_n$ for all t by using the consequence that each $\frac{dP_n}{dt}$ must equal 0, if the solution P_n is indeed independent of t . Hence, to obtain P_n , all we need to do is set the time derivation in (4) and (5) equal to 0:

$$0 = \lambda P_{n-1} - (\lambda + \mu)P_n + \mu P_{n+1} \quad \text{for } n = 1, 2, 3, \dots \quad (7)$$

$$0 = -\lambda P_0 + \mu P_1 \quad \text{for } n = 0 \quad (8)$$

The system of difference equations (7) and (8) is easily solved recursively starting with (8):

$$0 = \lambda P_{n-1} - (\lambda + \mu)P_n + \mu P_{n+1}$$

$$\mu P_{n+1} = (\lambda + \mu)P_n - \lambda P_{n-1}$$

$$0 = -\lambda P_0 + \mu P_1$$

$$\lambda P_0 = \mu P_1$$

By dividing the equation by μ we get this set of equations that contains the only parameter ρ - the traffic rate:

$$P_{n+1} = \left(\frac{\lambda + \mu}{\mu} \right) P_n - \left(\frac{\lambda}{\mu} \right) P_{n-1}$$

$$P_{n+1} = \left(\frac{\lambda}{\mu} + \frac{\mu}{\mu} \right) P_n - \rho P_{n-1}$$

$$P_{n+1} = (1 + \rho)P_n - \rho P_{n-1}$$

$$P_1 = \left(\frac{\lambda}{\mu}\right) P_0 = \rho P_0 \quad (9)$$

In the set (9) P_1 is expressed by P_0 . By inserting P_1 and P_0 to the equation for P_2 we get:

$$P_2 = (1 + \rho)P_1 - \rho P_0 = (1 + \rho)\rho P_0 - \rho P_0 = \rho P_0 + \rho^2 P_0 - \rho P_0 = \rho^2 P_0$$

$$P_3 = (1 + \rho)P_2 - \rho P_1 = (1 + \rho)\rho^2 P_0 - \rho(\rho P_0) = \rho^2 P_0 + \rho^3 P_0 - \rho^2 P_0 = \rho^3 P_0$$

Similarly P_3 and P_2 may be used to express P_4 , etc which gives the general formula for P_n .

$$P_n = P_0 \rho^n \quad (10)$$

The value P_0 can be computed by using the obvious requirement, that the sum of all probabilities must be equal to 1:

$$\sum_{n=0}^{\infty} P_n = P_0 \sum \rho^n = \frac{P_0}{1 - \rho} = 1$$

$$\text{That is giving } P_0 = 1 - \rho \quad (11)$$

So that

$$P_n = (1 - \rho) \rho^n \quad \text{for } n = 0, 1, 2.$$

Now we can calculate the average number of packets in the system in steady state.

$$N = \lim_{t \rightarrow \infty} E \{N(t)\} = \sum_{n=0}^{\infty} n \cdot P_n = \sum_{n=0}^{\infty} n \cdot \rho^n \cdot (1 - \rho)$$

$$\begin{aligned}
&= \rho(1-\rho) \sum_{n=0}^{\infty} n \cdot \rho^{n-1} = \rho(1-\rho) \cdot \frac{\partial}{\partial \rho} \left[\sum_{n=0}^{\infty} \rho^n \right] \\
&= \rho(1-\rho) \cdot \frac{\partial}{\partial \rho} \left(\frac{1}{1-\rho} \right) = \rho(1-\rho) \left(\frac{1}{(1-\rho)^2} \right) \\
&= \frac{\rho(1-\rho)}{(1-\rho)(1-\rho)} = \frac{\rho}{1-\rho}
\end{aligned}$$

And formally using $\frac{\lambda}{\mu}$

$$\begin{aligned}
N &= \frac{\rho}{1-\rho} = \frac{\frac{\lambda}{\mu}}{1-\frac{\lambda}{\mu}} = \frac{\frac{\lambda}{\mu}}{\frac{\mu-\lambda}{\mu}} = \frac{\lambda}{\mu} \times \frac{\mu}{\mu-\lambda} \\
N &= \frac{\lambda}{\mu-\lambda}
\end{aligned}$$

The average delay per packet (waiting time in the queue plus service time) is given by Little's theorem:

$$\begin{aligned}
T &= \frac{N}{\lambda} = \frac{\rho}{\lambda(1-\rho)} \text{ using } \rho = \frac{\lambda}{\mu} \\
&= \frac{\frac{\lambda}{\mu}}{\lambda(1-\frac{\lambda}{\mu})} = \frac{\frac{\lambda}{\mu}}{\lambda(\frac{\mu-\lambda}{\mu})} = \frac{\lambda}{\mu} \times \frac{\mu}{\lambda(\mu-\lambda)} = \frac{1}{\mu-\lambda} \\
T &= \frac{1}{\mu-\lambda}
\end{aligned}$$

The average waiting time in queue, W , is average delay T less the average service time $\frac{1}{\mu}$,

$$\begin{aligned} W &= \frac{1}{\mu - \lambda} - \frac{1}{\mu} \\ &= \frac{\mu - (\mu - \lambda)}{\mu(\mu - \lambda)} = \frac{\mu}{\mu(\mu - \lambda)} - \frac{\mu(1 - \frac{\lambda}{\mu})}{\mu(\mu - \lambda)} \\ &= \frac{1}{\mu - \lambda} - \frac{1 - \rho}{\mu - \lambda} = \frac{\rho}{\mu - \lambda} \end{aligned}$$

$$W = \frac{\rho}{\mu - \lambda}$$

Using again Little's theorem we can find the average number of packets in queue is:

$$\begin{aligned} N_q &= \lambda W = \frac{\lambda \rho}{\mu - \lambda} \\ &= \frac{\lambda \cdot \frac{\lambda}{\mu}}{\mu - \lambda} = \frac{\frac{\lambda^2}{\mu}}{\mu(1 - \frac{\lambda}{\mu})} = \frac{\lambda^2}{\mu} \times \frac{1}{\mu(1 - \frac{\lambda}{\mu})} \\ &= \frac{\lambda^2}{\mu^2(1 - \frac{\lambda}{\mu})} = \frac{\rho^2}{1 - \rho} \\ N_q &= \frac{\rho^2}{1 - \rho} \end{aligned}$$

M/M/m- the m-server case

The M/M/m queuing system is identical to the M/M/1 system except that there are m servers. A packet at the head of the queue is routed to any server that is available. By writing down the global balance equations for the steady probabilities P_n we obtain:

$$\begin{aligned}\lambda.P_{n-1} &= n.\mu.P_n & n \leq m \\ \lambda.P_{n-1} &= n.\mu.P_n & n > m\end{aligned}$$

From these equations we obtain:

$$P_n = \begin{cases} P_0 \left(\frac{m.\rho}{n!} \right)^n, & n \leq m \\ P_0 \frac{m^m.\rho^n}{m!}, & n > m \end{cases} \quad (1)$$

Where ρ is given by

$$\rho = \frac{\lambda}{m\mu} < 1$$

We can calculate P_0 using (1) and the condition $\sum_{n=0}^{\infty} P_n = 1$. we obtain

$$P_0 = \left[1 + \sum_{n=1}^{m-1} \frac{(m\rho)^n}{n!} + \sum_{n=m}^{\infty} \frac{(m\rho)^n}{m!} \cdot \frac{1}{m^{n-m}} \right]^{-1} \quad (2)$$

And finally,

$$P_0 = \left[1 + \sum_{n=1}^{m-1} \frac{(m\rho)^n}{n!} + \frac{(m\rho)^m}{m!(1-\rho)} \right]^{-1}$$

The probability that an arrival will find all servers busy and will force to wait in queue is an important measure of performance of the M/M/m system. We have:

$$\begin{aligned}
 P\{Queueing\} &= \sum_{n=1}^{\infty} P_n = \sum_{n=m}^{\infty} \frac{P_0 m^m \rho^n}{m!} = P_0 \frac{(m\rho)^m}{m!} \sum_{n=m}^{\infty} \rho^{n-m} \\
 &= \frac{P_0 (m\rho)^m}{m!(1-\rho)} \quad (3)
 \end{aligned}$$

The expected number of packets waiting in the queue is given by

$$N_Q = \sum_{n=0}^{\infty} n.P_{n+m}$$

Using the expression for P_{m+n} in (1), we obtain

$$N_Q = \sum_{n=0}^{\infty} n.P_0 \frac{m^m \rho^{m+n}}{m!} = \frac{P_0 (m\rho)^m}{m!} \sum_{n=0}^{\infty} n.\rho^n$$

Using (3) we obtain

$$N_Q = P_Q \frac{\rho}{1-\rho} \quad (4)$$

Note that

$$\frac{N_Q}{P_Q} = \frac{\rho}{1-\rho}$$

Using Little's theorem and expression (4) for N_Q , we obtain the average time W a packet has to wait in the queue:

$$W = \frac{N_Q}{\lambda} = \frac{\rho \cdot P_Q}{\lambda \cdot (1 - \rho)}$$

The average delay per packet is, therefore,

$$T = \frac{1}{\mu} + W = \frac{1}{\mu} + \frac{\rho \cdot P_Q}{\lambda \cdot (1 - \rho)}$$

And using $\rho = \frac{\lambda}{m \cdot \mu}$, we obtain

$$T = \frac{1}{\mu} + W = \frac{1}{\mu} + \frac{P_Q}{m \cdot \mu - \lambda}$$

Using Little's theorem again, the average number of packets in the system is:

$$N = \lambda \cdot T = \frac{\lambda}{\mu} + \frac{P_Q}{m \cdot \mu - \lambda}$$

And using once more $\rho = \frac{\lambda}{m \cdot \mu}$, we finally obtain

$$N = m\rho + \frac{\rho \cdot P_Q}{1 - \rho}$$

Appendix C

**/ JavaScript program for M/M/1 Queue model*

<HTML>

<HEAD>

<SCRIPT LANGUAGE="JavaScript">

<!--

//----- General functions -----

Function TestStability()

{

 la = document.myForm.arrivalRate.value;

 m = document.myForm.serviceRate.value;

 if (la <= 0)

 {

 alert("Arrival rate is not a positive value ! Correct it and try again.");

 return false

 }

 else

 {

 if (m <= 0)

 {

 alert ("Service rate is not a positive value ! Correct it and try again.");

 return false

 }

 else

 {

 x = la/m;

 if (x>=1)

```

{
    alert("Your system is not stable ! For a stable system the Arrival rate must be smaller
        than the Service rate. Enter correct values and try again.");
    return false
}
else
{
    return true
}
}
}
}

```

//----- End -----

```

function testPositiveValue(textElement)
{
    if (textElement <= 0)
    {
        alert("Please enter a positive value !")
    }
    else
        return true
}

```

//----- End -----

```

function testExper()
{
    if (document.myForm.expduration.value <= 0)
    {
        alert("Experiment duration must be a positive number. I give it the length 1000 !");
        document.myForm.expduration.value = 1000;
    }
}

```

```

    }
    else
        return true
    }

//----- End -----

function testArray()
{
    if (document.myForm.Qsize.value <= 1000)
    {
        alert("Your array seems too small, I give it the size 1000 !");
        document.myForm.Qsize.value = 100;
    }
    else return true
}

```

```

//----- End -----

```

```

function Random()
{
    return Math.random()
}

```

```

----- End -----

```

```

function Results()      // Analytical results
{
    if (TestStability())
    {
        lambda = document.myForm.arrivalRate.value;
        mu = document.myForm.serviceRate.value;
        ro = lambda/mu;
    }
}

```

```

        document.myForm.p0Comp.value = 1-ro;
        document.myForm.LsComp.value = ro/ (1-ro);
        document.myForm.LqComp.value = ro*ro/ (1-ro);
        document.myForm.WsComp.value = 1/ (mu-lambda);
        document.myForm.WqComp.value = ro/ (mu-lambda);
    }

}

//-----End of Analytical Simulation -----

function Initialisation() // Model initialisation
{
    lambda = document.myForm.arrivalRate.value;
    mu    = document.myForm.serviceRate.value;
    time  = 0;           // { simulated time }
    runlength = document.myForm.expduration.value; // { length of the experiment }

    interval = 0;           //{ next interval }
    intmean = 1/lambda;     //{ mean interval between arrivals }
    intmin = runlength;     //{ minimum arrival interval }
    intmax = 0;             //{ maximum arrival interval }
    nextarr = 0;            //{ time of next arrival }
    numoarr = 0;            //{ number of arrivals }
    nowait = 0;             //{ number of not waiting arrivals }
    service = 0;            //{ next service duration }
    sermean = 1/mu;         //{ mean service duration }
    sermin = runlength;     //{ minimum service duration }
    sermax = 0;             //{ maximum service duration }
    numoser = 0;            //{ number of completed services }
    nexteos = 0;            //{ time of next end of service }
    arrserved = 0;          //{ arrival time of the served customer }
    qsize = document.myForm.Qsize.value;    //{ queue array size }
    queue.length = 0;       //{ truncating the array }
}

```

```

queue[qsize-1] = 0;           //{ fixing the array size }
qhead = 0;                   //{ points to the first item }
qtail = 0;                   //{ points to first free slot behind the last item}
qlength = 0;                 //{ queue length }
maxlength = 0;               //{ maximum queue length }
qlsum = 0;                   //{ queue length integral }
qlsumt = 0;                  //{ queue length integral last update }
sbusy = 0;                   //{ service busy: 0=no,1=yes }
susum = 0;                   //{ service busy integral }
susumt = 0;                  //{ service busy integral last update }
wtime = 0;                   //{ waiting time }
maxwtime = 0;                //{ maximum waiting time }
swtime = 0;                  //{ sum of waiting times }
stime = 0;                   //{ time in system }
maxstime = 0;                //{ maximum time in system }
sstime = 0;                  //{ sum of times in system }
}

```

//----- End -----

```

function Accumulate()        //{ Updating integrals }
{
    qlsum = qlsum + qlength*(time-qlsumt);
    qlsumt = time;
    susum = susum + sbusy*(time-susumt);
    susumt = time;
}

```

//----- End -----

```

function Evaluation()        //{ Displays results }
{
    document.myForm.WsSim.value = sstime/numoser;
    n = numoser;
}

```



```

if (sbusy == 1) n--;
document.myForm.WqSim.value = swtime/n;
document.myForm.LqSim.value = qlsum/time;
document.myForm.LsSim.value = (qlsum + susum)/time;
document.myForm.p0Sim.value = nowait/numoarr;

}

//----- End -----

function Simulation()    //{ Simulation run }
{
if (TestStability())
{
    Initialisation();

// Scheduling first arrival:
nextarr = -intmean * Math.log(Random());
nexteos = nextarr + 1;    //{ this means no service scheduled }

    while (time < runlength)
    {
if (nextarr < nexteos)
{
            //{ arrival to occur }
time = nextarr;    //{ advance time }
Accumulate();
numoarr++;
interval = -intmean * Math.log(Random());
nextarr += interval;    //{ scheduling the next arrival }
if (interval > intmax) intmax = interval;
if (interval < intmin) intmin = interval;
if (sbusy==0)
{
            //{ service not busy }
nowait++;

```

```

sbusy = 1;
service = -sermean * Math.log(Random());
nexteos = time + service; //{ scheduling the end of service }
if (service > sermax) sermax = service;
if (service < sermin) sermin = service;
arrserved = time;
}
else
{
    //{ queueing necessary }
    qlength++;
    queue[qtail++] = time;        //{ record arrival time in queue }
    if (qtail == qsize) qtail = 0;    //{ wrap around }
    if (qlength > maxqlength) maxqlength = qlength;
}
}
else
{
    //{ end of service to occur }
    time = nexteos;    //{ advance time }
    Accumulate();
    numoser++;
    stime = time - arrserved;
    sstime += stime;
    if (stime > maxstime) maxstime = stime;
    sbusy = 0;
    nexteos = nextarr + 1; //{ this means no service scheduled }
    if (qlength > 0)
    {
        //{ start the next service }
        sbusy = 1;
        service = -sermean * Math.log(Random());
        nexteos = time + service; //{ scheduling the end of service }
        if (service > sermax) sermax = service;
        if (service < sermin) sermin = service;
        wtime = time - queue[qhead]; //{ waiting time }
        swtime += wtime;
    }
}

```

```

    if (wtime > maxwtime) maxwtime = wtime;
    arrserved = queue[qhead++];
    if (qhead == qsize) qhead = 0; // { wrap around }
    qlength--;
}
}
status = "Running - Time: " + Math.round(time) + " of " + runlength + " Queue: " + qlength
+ " Arrivals: " + numoarr;
}
Evaluation();
}
}

```

//----- Head Code -----

```

var queue = new Array();    //{ creating the queue array }
var lambda = 0;
var mu    = 0;
var time  = 0;              //{ simulated time }
var runlength = 0;          //{ length of the experiment }

var interval = 0;           //{ next interval }
var intmean = 0;            //{ mean interval between arrivals }
var intmin = 0;             //{ minimum arrival interval }
var intmax = 0;             //{ maximum arrival interval }
var nextarr = 0;            //{ time of next arrival }
var numoarr = 0;            //{ number of arrivals }
var nowait = 0;             //{ number of not waiting arrivals }

var service = 0;            //{ next service duration }
var sermean = 0;            //{ mean service duration }
var sermin = 0;            //{ minimum service duration }
var sermax = 0;            //{ maximum service duration }
var numoser = 0;           //{ number of completed services }

```

```

var nexteos = 0;      //{ time of next end of service }
var arrserved = 0;    //{ arrival time of the served customer }

var qsize = 0;        //{ queue array size }
var qhead = 0;        //{ points to the first item }
var qtail = 0;        //{ points to first free slot behind the last item }
var qlength = 0;      //{ queue length }
var maxqlength = 0;   //{ maximum queue length }
var qlsum = 0;        //{ queue length integral }
var qlsumt = 0;       //{ queue length integral last update }

var sbusy = 0;        //{ service busy: 0=no,1=yes }
var susum = 0;        //{ service busy integral }
var susumt = 0;       //{ service busy integral last update }

var wtime = 0;        //{ waiting time }
var maxwtime = 0;     //{ maximum waiting time }
var swtime = 0;       //{ sum of waiting times }

var stime = 0;        //{ time in system }
var maxstime = 0;     //{ maximum time in system }
var sstime = 0;       //{ sum of times in system }

</SCRIPT>
</HEAD>

```

```

<BODY bgcolor="#bC0bC0bC0">
<CENTER><h1>M/M/1 Simulator</h1>
</CENTER>

<FORM NAME="myForm">
<CENTER><h2>Data Entry Form</h2></CENTER>

```

```

<CENTER>
<TABLE BRCOLOR=yellow BORDER CELLPADDING="8" CELLSPACING="4" >
<TR BGCOLOR=cyan><TH> Parameter </TH><TH> Value </TH><TH> Description
</TH></TR>
<TR>

<TD> Arrival Rate (<FONT FACE="Symbol">&#108;</FONT>) </TD>
<TD> <INPUT TYPE="text" NAME="arrivalRate" SIZE=10 VALUE="1"
onBlur="testPositiveValue(arrivalRate.value)"> </TD>
<TD ALIGN="CENTER"> <INPUT TYPE="button" VALUE="Help" NAME="help1Button"
onClick="alert('Arrival rate is the average number of arrivals per time unit. It is the inverted
value of the average interval between two adjacent arrivals.')"></TD>
</TR>
<TR>
<TD> Service Rate (<FONT FACE="Symbol">&#109;</FONT>) </TD>
<TD> <INPUT TYPE="text" NAME="serviceRate" SIZE=10 VALUE="2"
onBlur="testPositiveValue(serviceRate.value)"></TD>
<TD ALIGN="CENTER"> <INPUT TYPE="button" VALUE="Help" NAME="help2Button"
onClick="alert('Service rate is the average number of completed services per time unit,
provided there are always customers in the queue. It is the inverted value of the average service
duration.')"></TD>
</TR>
<TR>
<TD> Experiment duration </TD>
<TD> <INPUT TYPE="text" NAME="expduration" SIZE=10 VALUE="1000"
onBlur="testExper()"></TD>
<TD ALIGN="CENTER"> <INPUT TYPE="button" VALUE="Help" NAME="help3Button"
onClick="alert('Experiment duration should be considerably higher than the maximum service
duration and the maximum interval between arrivals. In the M/M/1 theoretical model both can be
very large, so make sure that the experiment duration is at least much longer than both average
values.')"></TD>
</TR>
<TR>

```

```

<TD> Maximum queue length </TD>
<TD> <INPUT TYPE="text" NAME="Qsize" SIZE=10 VALUE="100"
    onBlur="testArray()"></TD>
<TD ALIGN="CENTER"> <INPUT TYPE="button" VALUE="Help" NAME="help4Button"
    onClick="alert('The queue is implemented as a circular list in an array. This number is the size
of the array. After simulation check, that the array size (this number) is bigger than the maximum
queue length reached. If not, repeat the experiment with a longer array.')"></TD>
</TR>
</TABLE>
</CENTER>

```

```

<P><CENTER>
<TABLE BORDER CELLPADDING="2" CELLSPACING="2" >
<TR BRCOLOR=magenta>

<TD ALIGN=Center WIDTH=120>
<INPUT TYPE="button" VALUE="Compute Results" NAME="resultsButton"
    onClick="Results()">
</TD>
<TD ALIGN=Right WIDTH=120>
<INPUT TYPE="button" VALUE="Run Simulation" NAME="simulateButton"
    onClick="Simulation()"><br>
</TD>
</TR>
</TABLE>
</CENTER>

```

```

//<p><hr>
<CENTER><h2>Basic Results of the simulation</h2></CENTER>
<CENTER>
<TABLE BGCOLOR= ORANGE BORDER CELLPADDING="8" CELLSPACING="4" >

```



```

<TR BGCOLOR=cyan><TH> Result </TH><TH> Computed value </TH><TH> Simulated
value </TH><TH> Explanation </TH></TR>
<TR>
<TD> packet number in system (L<sub>s</sub>) </TD>
<TD> <INPUT TYPE="text" NAME="LsComp" SIZE=10> </TD>
<TD> <INPUT TYPE="text" NAME="LsSim" SIZE=10> </TD>

<TD ALIGN="CENTER"> <INPUT TYPE="button" VALUE="Help" NAME="help5Button"
onClick="alert(' Average number of packet in the system (including the one being
served).')"></TD>
</TR>
<TR>
<TD> packet number in queue (L<sub>q</sub>) </TD>
<TD> <INPUT TYPE="text" NAME="LqComp" SIZE=10> </TD>
<TD> <INPUT TYPE="text" NAME="LqSim" SIZE=10> </TD>

<TD ALIGN="CENTER"> <INPUT TYPE="button" VALUE="Help" NAME="help6Button"
onClick="alert(' Average number of packets in the queue (average queue length including
empty queue intervals).')"></TD>
</TR>
<TR>
<TD> Time in system (W<sub>s</sub>) </TD>
<TD> <INPUT TYPE="text" NAME="WsComp" SIZE=10> </TD>
<TD> <INPUT TYPE="text" NAME="WsSim" SIZE=10> </TD>

<TD ALIGN="CENTER"> <INPUT TYPE="button" VALUE="Help" NAME="help7Button"
onClick="alert(' Average time spent in the system (average of waiting + service
time).')"></TD>
</TR>
<TR>
<TD> Time in queue (W<sub>q</sub>) </TD>
<TD> <INPUT TYPE="text" NAME="WqComp" SIZE=10> </TD>
<TD> <INPUT TYPE="text" NAME="WqSim" SIZE=10> </TD>

```

```

<TD ALIGN="CENTER"> <INPUT TYPE="button" VALUE="Help" NAME="help7Button"
onClick="alert('Average time spent in the queue (average waiting time including zero - no
waiting - values).')"></TD>
</TR>
<TR>
<TD> Idle probability ( $p_0$ ) </TD>
<TD> <INPUT TYPE="text" NAME="p0Comp" SIZE=10> </TD>
<TD> <INPUT TYPE="text" NAME="p0Sim" SIZE=10> </TD>
<TD ALIGN="CENTER"> <INPUT TYPE="button" VALUE="Help" NAME="help8Button"
onClick="alert('Probability, that the arriving packet does not wait = Probability, that the system
is empty = 1 - Probability, that the server is busy.')"></TD>

</FORM>
</BODY>
</HTML>
//----- End the Simulation program of M/M/1 -----

```

****/ JavaScript program for M/M/2***

```

<HTML>
<HEAD>

<SCRIPT LANGUAGE="JavaScript">
<!--
//----- General functions -----

function TestStability()
{

```

```

la = document.myForm.arrivalRate.value;

mu = document.myForm.serviceRate.value;
m = document.myForm.totalNumber.value;
if (la <= 0)
{
    alert("Arrival rate is not a positive value ! Correct it and try again.");
    return false
}
else
{
    if (mu <= 0)
    {
        alert("Service rate is not a positive value ! Correct it and try again.");
        return false
    }
    else
    {
        if (m <= 0)
        {
            alert("the total server numbers are not a positive value ! Correct it and try again.");
            return false
        }
        else
        {
            x = la / (m * mu);
            if (x >= 1)
            {
                alert("Your system is not stable ! For a stable system the Arrival rate must be smaller than the Service rate. Enter correct values and try again.");
                return false
            }
            else
            {

```

```

        return true
    }
}
}
}
}
//----- End -----

```

```

function testPositiveValue(textElement)
{
    if (textElement <= 0)
    {
        alert("Please enter a positive value !")
    }
    else return true
}

//----- End -----

```

```

function testExper()
{
    if (document.myForm.expduration.value <= 0)
    {
        alert("Experiment duration must be a positive number. I give it the length 1000 !");
        document.myForm.expduration.value = 1;
    }
    else
        return true
}

//-----End-----

```

```

function testArray()
{

```

```

if (document.myForm.Qsize.value < 1)
{
    alert("Your array seems too small, I give it the size 1000 !");
    document.myForm.Qsize.value = 1;
}
else
    return true
}

```

```

//-----End-----

```

```

function Random()
{
    return Math.random()
}

```

```

function Results()           // Analytical results
{
    if (TestStability())
    {

        sum1 =0;
        sum2 =0;
        p=0;
        p1=0;
        N=0;
        Nq=0;
        Ws=0;
        T=0;

```

```

        lambda = document.myForm.arrivalRate.value;
        mu = document.myForm.serviceRate.value;
        m = document.myForm.totalNumber.value;

```

```
ro = lambda/(mu*m);
```

```
if (m== 2) {
```

```
    sum1 = 1 + m*ro;
```

```
    sum2 = (m*ro)*(m*ro)/(1-ro);
```

```
    p = 1/(sum1 + sum2);
```

```
    p1 = p*sum2;
```

```
    N = ro*m + (ro - p1)/(1-ro);
```

```
    Nq = p1*ro/(1-ro);
```

```
    W = ro*p1/(lambda*(1-ro));
```

```
    T = 1/mu + ro*p1/(lambda*(1-ro));
```

```
}
```

```
else
```

```
{
```

```
    if (m == 3) {
```

```
        sum1 = (1 + (m*ro) + ((m*ro)*(m*ro)/2));
```

```
        sum2 = (m*ro)*(m*ro)*(m*ro)/(6*(1-ro));
```

```
        p = 1/(sum1 + sum2);
```

```
        p1 = p*sum2;
```

```
        N = ro*m + (ro - p1)/(1-ro);
```

```
        Nq = p1*ro/(1-ro);
```

```
        W = ro*p1/(lambda*(1-ro));
```

```
        T = 1/mu + ro*p1/(lambda*(1-ro));
```

```
    }
```

```
else
```

```
{
```

```
    if (m == 4) {
```

```
        sum1 = (1 + (m*ro) + ((m*ro)*(m*ro)/2) + ((m*ro)*(m*ro)*(m*ro)/6));
```

```
        sum2 = (m*ro)*(m*ro)*(m*ro)*(m*ro)/(24*(1-ro));
```

```
        p = 1/(sum1 + sum2);
```

```
        p1 = p*sum2;
```

```
        N = ro*m + (ro - p1)/(1-ro);
```

```
        Nq = p1*ro/(1-ro);
```

```
        W = ro*p1/(lambda*(1-ro));
```



```

    T = 1/mu + ro*p1/ (lambda*(1-ro));
}
else
{
if (m == 5) {
    sum1 = (1 + (m*ro) + ((m*ro)*(m*ro)/2) + ((m*ro)*(m*ro)*(m*ro)/6)
    + ((m*ro)*(m*ro)*(m*ro)*(m*ro)/24));
    sum2 = (m*ro)*(m*ro)*(m*ro)*(m*ro)*(m*ro)/ (120*(1-ro));
    p = 1/ (sum1 + sum2);
    p1= p*sum2;
    N = ro*m + (ro - p1)/ (1-ro);
    Nq = p1*ro/ (1-ro);
    W = ro*p1/ (lambda*(1-ro));
    T = 1/mu + ro*p1/ (lambda*(1-ro));
}
else
{
if (m == 6) {
    sum1 = (1 + (m*ro) + ((m*ro)*(m*ro)/2) + ((m*ro)*(m*ro)*(m*ro)/6) +
    ((m*ro)*(m*ro)*(m*ro)*(m*ro)/24) + ((m*ro)*(m*ro)*(m*ro)*(m*ro)*(m*ro)/120));
    sum2 = (m*ro)*(m*ro)*(m*ro)*(m*ro)*(m*ro)*(m*ro)/(720*(1-ro));
    p = 1/ (sum1 + sum2);
    p1= p*sum2;
    N = ro*m + (ro - p1)/ (1-ro);
    Nq = p1*ro/ (1-ro);
    W = ro*p1/ (lambda*(1-ro));
    T = 1/mu + ro*p1/ (lambda*(1-ro));
}
}
}
}
}

```

```

document.myForm.roComp.value = ro;
document.myForm.p0Comp.value = p;
document.myForm.LsComp.value = N;
document.myForm.LqComp.value = Nq;
document.myForm.WsComp.value = W;
document.myForm.WqComp.value = T;

}

}

//-----End -----

function Initialisation () // Model initialisation
{
    lambda = document.myForm.arrivalRate.value;
    mu    = document.myForm.serviceRate.value;
    time  = 0;           //{ simulated time }
    runlength = document.myForm.expduration.value; //{ length of the experiment }

    interval = 0;           //{ next interval }
    intmean = 1/lambda;     //{ mean interval between arrivals }
    intmin = runlength;     //{ minimum arrival interval }
    intmax = 0;             //{ maximum arrival interval }
    nextarr = 0;            //{ time of next arrival }
    numoarr = 0;            //{ number of arrivals }
    nowait = 0;             //{ number of not waiting arrivals }

    service = 0;           //{ next service duration }
    sermean = 1/mu;        //{ mean service duration }
    sermin = runlength;    //{ minimum service duration }
    sermax = 0;            //{ maximum service duration }
    numoser = 0;           //{ number of completed services }

```

```

nexteos = 0;          //{ time of next end of service }
arrserved = 0;        //{ arrival time of the served customer }
qsize = document.myForm.Qsize.value;    //{ queue array size }
queue.length = 0;     //{ truncating the array }
queue[qsize-1] = 0;   //{ fixing the array size }
qhead = 0;            //{ points to the first item }
qtail = 0;            //{ points to first free slot behind the last item }
qlength = 0;          //{ queue length }
maxqlength = 0;       //{ maximum queue length }
qlsum = 0;            //{ queue length integral }
qlsumt = 0;           //{ queue length integral last update }
sbusy = 0;            //{ service busy: 0=no,1=yes }
susum = 0;            //{ service busy integral }
susumt = 0;           //{ service busy integral last update }
wtime = 0;            //{ waiting time }
maxwtime = 0;         //{ maximum waiting time }
swtime = 0;           //{ sum of waiting times }
stime = 0;            //{ time in system }
maxstime = 0;         //{ maximum time in system }
sstime = 0;           //{ sum of times in system }
}

```

//-----End-----

```

function Accumulate()  //{ Updating integrals }
{
    qlsum = qlsum + qlength*(time-qlsumt);
    qlsumt = time;
    susum = susum + sbusy*(time-susumt);
    susumt = time;
}

```

//----- End -----

```

function Evaluation()    //{ Displays results }
{
    document.myForm.WsSim.value = sstime/numoser;
    n = numoser;
    if (sbusy == 1) n--;
    document.myForm.WqSim.value = swtime/n;
    document.myForm.LqSim.value = qlsum/time;
    document.myForm.LsSim.value = (qlsum + susum)/time;
    document.myForm.p0Sim.value = nowait/numoarr;
    document.myForm.roSim.value = susum/time;

}

```

//-----

```

function Simulation()    //{ Simulation run }
{
    if (TestStability())
    {
        Initialisation();

        // Scheduling first arrival:
        nextarr = -intmean * Math.log(Random());
        nexteos = nextarr + 1; //{ this means no service scheduled }

        while (time < runlength)
        {
            if (nextarr < nexteos)
            {
                //{ arrival to occur }
                time = nextarr;    //{ advance time }
                Accumulate();
                numoarr++;
                interval = -intmean * Math.log(Random());
                nextarr += interval; //{ scheduling the next arrival }
            }
        }
    }
}

```

```

if (interval > intmax) intmax = interval;
if (interval < intmin) intmin = interval;
if (sbusy==0)
{
    //{ service not busy }
    nowait++;
    sbusy = 1;
    service = -sermean * Math.log(Random());
    nexteos = time + service; //{ scheduling the end of service }
    if (service > sermax) sermax = service;
    if (service < sermin) sermin = service;
    arrserved = time;
}
else
{
    //{ queueing necessary }
    qlength++;
    queue[qtail++] = time; //{ record arrival time in queue }
    if (qtail == qsize) qtail = 0; //{ wrap around }
    if (qlength > maxqlength) maxqlength = qlength;
}
}
else
{
    //{ end of service to occur }
    time = nexteos; //{ advance time }
    Accumulate();
    numoser++;
    stime = time - arrserved;
    sstime += stime;
    if (stime > maxstime) maxstime = stime;
    sbusy = 0;
    nexteos = nextarr + 1; //{ this means no service scheduled }
    if (qlength > 0)
    {
        //{ start the next service }
        sbusy = 1;
        service = -sermean * Math.log(Random());
    }
}

```

```

nexteos = time + service; //{ scheduling the end of service }
if (service > sermax) sermax = service;
if (service < sermin) sermin = service;
wtime = time - queue[qhead]; //{ waiting time }
swtime += wtime;
if (wtime > maxwtime) maxwtime = wtime;
arrserved = queue[qhead++];
if (qhead == qsize) qhead = 0; //{ wrap around }
qlength--;
}
}
status = "Running - Time: " + Math.round(time) + " of " + runlength + " Queue: " + qlength +
" Arrivals: " + numoarr;
}
Evaluation();
}
}

```

//----- End-----

// global variables:

```

var queue = new Array(); //{ creating the queue array }
var lambda = 0;
var mu = 0;
var time = 0; //{ simulated time }
var runlength = 0; //{ length of the experiment }

var interval = 0; //{ next interval }
var intmean = 0; //{ mean interval between arrivals }
var intmin = 0; //{ minimum arrival interval }
var intmax = 0; //{ maximum arrival interval }
var nextarr = 0; //{ time of next arrival }
var numoarr = 0; //{ number of arrivals }

```



```

var nowait = 0;    //{ number of not waiting arrivals }

var service = 0;   //{ next service duration }
var sermean = 0;   //{ mean service duration }
var sermin = 0;    //{ minimum service duration }
var sermax = 0;    //{ maximum service duration }
var numoser = 0;   //{ number of completed services }
var nexteos = 0;   //{ time of next end of service }
var arrserved = 0; //{ arrival time of the served customer }

var qsize = 0;     //{ queue array size }
var qhead = 0;     //{ points to the first item }
var qtail = 0;     //{ points to first free slot behind the last item }
var qlength = 0;   //{ queue length }
var maxqlength = 0; //{ maximum queue length }
var qlsum = 0;     //{ queue length integral }
var qlsumt = 0;    //{ queue length integral last update }

var sbusy = 0;     //{ service busy: 0=no,1=yes }
var susum = 0;     //{ service busy integral }
var susumt = 0;    //{ service busy integral last update }

var wtime = 0;     //{ waiting time }
var maxwtime = 0;  //{ maximum waiting time }
var swtime = 0;    //{ sum of waiting times }

var stime = 0;     //{ time in system }
var maxstime = 0;  //{ maximum time in system }
var sstime = 0;    //{ sum of times in system }

// -->
</SCRIPT>
</HEAD>

```

```

<BODY bgcolor="#bC0bC0bC0">
<CENTER><h1>M/M/m Simulator</h1>
</CENTER>

```

```

<FORM NAME="myForm">
<CENTER><h2>Data Entry Form</h2></CENTER>

```

```

<CENTER>
<TABLE BRCOLOR=yellow BORDER CELLPADDING="8" CELLSPACING="4" >
<TR BGCOLOR=cyan><TH> Parameter </TH><TH> Value </TH><TH> Description
</TH></TR>
<TR>
<TD> Arrival Rate (<FONT FACE="Symbol">&#108;</FONT>) </TD>
<TD> <INPUT TYPE="text" NAME="arrivalRate" SIZE=10 VALUE="1"
onBlur="testPositiveValue(arrivalRate.value)"> </TD>
<TD ALIGN="CENTER"> <INPUT TYPE="button" VALUE="Help" NAME="help1Button"
onClick="alert('Arrival rate is the average number of arrivals per time unit. It is the inverted
value of the average interval between two adjacent arrivals.')"></TD>
</TR>
<TR>
<TD> Service Rate (<FONT FACE="Symbol">&#109;</FONT>) </TD>
<TD> <INPUT TYPE="text" NAME="serviceRate" SIZE=10 VALUE="2"
onBlur="testPositiveValue(serviceRate.value)"></TD>
<TD ALIGN="CENTER"> <INPUT TYPE="button" VALUE="Help" NAME="help2Button"
onClick="alert('Service rate is the average number of completed services per time unit,
provided there are always customers in the queue. It is the inverted value of the average service
duration.')"></TD>
</TR>
<TR>
<TD> Total Number of the Server (<FONT FACE="N";</FONT>) </TD>
<TD> <INPUT TYPE="text" NAME="totalNumber" SIZE=10 VALUE="2"

```

```

onBlur="testPositiveValue(totalNumber.value)"></TD>
<TD ALIGN="CENTER"> <INPUT TYPE="button" VALUE="Help" NAME="help2Button"
onClick="alert ('this parameter is used to determine the total number of the server')"></TD>
</TR>
<TR>
<TD> Experiment duration </TD>
<TD> <INPUT TYPE="text" NAME="expduration" SIZE=10 VALUE="1000"
onBlur="testExper()"></TD>
<TD ALIGN="CENTER"> <INPUT TYPE="button" VALUE="Help" NAME="help3Button"
onClick="alert('Experiment duration should be considerably higher than the maximum service
duration and the maximum interval between arrivals. In the M/M/2 theoretical model both can be
very large, so make sure that the experiment duration is at least much longer than both average
values.')"></TD>
</TR>
<TR>
<TD> Maximum queue length </TD>
<TD> <INPUT TYPE="text" NAME="Qsize" SIZE=10 VALUE="1000"
onBlur="testArray()"></TD>
<TD ALIGN="CENTER"> <INPUT TYPE="button" VALUE="Help" NAME="help4Button"
onClick="alert('The queue is implemented as a circular list in an array. This number is the size
of the array. After simulation check, that the array size (this number) is bigger than the maximum
queue length reached. If not, repeat the experiment with a longer array.')"></TD>
</TR>
</TABLE>
</CENTER>
<P><CENTER>
<TABLE BORDER CELLPADDING="4" CELLSPACING="2" >
<TR BRCOLOR=magenta>
<TD ALIGN=Right WIDTH=120>
<INPUT TYPE="button" VALUE="Compute Results" NAME="resultsButton"
onClick="Results()">
</TD>
<TD ALIGN=Right WIDTH=120>
<INPUT TYPE="button" VALUE="Run Simulation" NAME="simulateButton"

```

```

onClick="Simulation()"><br>
</TD>
</TR>
</TABLE>
</CENTER>
<p><hr>
<CENTER><h2>Basic Results of the simulation</h2></CENTER>
<CENTER>
<TABLE BGCOLOR= ORANGE BORDER CELLPADDING="8" CELLSPACING="4" >
<TR BGCOLOR=cyan><TH> Result </TH><TH> Computed value </TH><TH> Simulated
value </TH><TH> Explanation </TH></TR>
<TR>
<TD> Packets in system (L<sub>s</sub>) </TD>
<TD> <INPUT TYPE="text" NAME="LsComp" SIZE=10> </TD>
<TD> <INPUT TYPE="text" NAME="LsSim" SIZE=10> </TD>

<TD ALIGN="CENTER"> <INPUT TYPE="button" VALUE="Help" NAME="help5Button"
onClick="alert('Average number of customers in the system (including the one being
served).')"></TD>
</TR>
<TR>
<TD> Packets in queue (L<sub>q</sub>) </TD>
<TD> <INPUT TYPE="text" NAME="LqComp" SIZE=10> </TD>
<TD> <INPUT TYPE="text" NAME="LqSim" SIZE=10> </TD>

<TD ALIGN="CENTER"> <INPUT TYPE="button" VALUE="Help" NAME="help6Button"
onClick="alert('Average number of customers in the queue (average queue length including
empty queue intervals).')"></TD>
</TR>
<TR>
<TD> Time in system (W<sub>s</sub>) </TD>
<TD> <INPUT TYPE="text" NAME="WsComp" SIZE=10> </TD>
<TD> <INPUT TYPE="text" NAME="WsSim" SIZE=10> </TD>

```

```

<TD ALIGN="CENTER"> <INPUT TYPE="button" VALUE="Help" NAME="help7Button"
  onClick="alert('Average time spent in the system (average of waiting + service
time).')"></TD>
</TR>
<TR>
<TD> Time in queue ( $W_q$ ) </TD>
<TD> <INPUT TYPE="text" NAME="WqComp" SIZE=10> </TD>
<TD> <INPUT TYPE="text" NAME="WqSim" SIZE=10> </TD>

<TD ALIGN="CENTER"> <INPUT TYPE="button" VALUE="Help" NAME="help7Button"
  onClick="alert('Average time spent in the queue (average waiting time including zero - no
waiting - values).')"></TD>
</TR>
<TR>
<TD> Idle probability ( $p_0$ ) </TD>
<TD> <INPUT TYPE="text" NAME="p0Comp" SIZE=10> </TD>
<TD> <INPUT TYPE="text" NAME="p0Sim" SIZE=10> </TD>
<TD ALIGN="CENTER"> <INPUT TYPE="button" VALUE="Help" NAME="help8Button"
  onClick="alert('Probability, that the arriving customer does not wait = Probability, that the
system is empty = 1 - Probability, that the server is busy.')"></TD>
</TR>
<TR>
<TD> Server utilisation ( $\rho$ ) </TD>
<TD> <INPUT TYPE="text" NAME="roComp" SIZE=10> </TD>
<TD> <INPUT TYPE="text" NAME="roSim" SIZE=10> </TD>
<TD ALIGN="CENTER"> <INPUT TYPE="button" VALUE="Help" NAME="help9Button"

onClick="alert('Server utilisation = Probability, that the server is busy = (1 - Probability that the
server is idle) = Traffic rate. Traffic rate is computed as a ratio Arrival rate / Service
rate.')"></TD>
</FORM>
</BODY>
</HT

```

Appendix D

E-Model Script program

```
*/ Declare Function
```

```
Private Sub cmdReset_Click()
```

```
If MsgBox("Do you wish to proceed with reset of calculator to default values?",
```

```
vbYesNo + vbQuestion) = vbYes Then
```

```
Loading = True
```

```
'Sender Loudness Rating (dB)
```

```
SLR = 8
```

```
'Receiver Loudness Rating (dB)
```

```
RLR = 2
```

```
'Overall Loudness Rating (dB)
```

```
OLR = 10
```

```
'Sidetone Masking Rating (dB)
```

```
STMR = 15
```

```
'Listener Sidetone Rating (dB)
```

```
LSTR = 18
```

```
'D Value of Telephone, Send Side
```

```
Ds = 3
```

```
'D Value of Telephone, Receiver Side
```

```
Dr = 3
```

```
'Talker Echo Loudness Rating (dB)
```

```
TELR = 65
```

```
'Weighted Echo Path Loss (dB)
```

```
WEPL = 110
```

```
'Mean One-Way Delay of the Echo Path (ms)
```

```
T = 0
```

```
'Round-trip Delay in a 4-wire Loop (ms)
```

```
Tr = 0
```

'Absolute Delay in Echo-Free Connections (ms)

Ta = 0

'Number of Quantisation Distortion Units

qdu = 1

'Equipment Impairment Factor

Ie = 0

'Packet-Loss Robustness Factor

Bpl = 1

'Random Loss Probability (%)

Ppl = 0

'Circuit Noise referred to 0-dBr point (dBm0p)

Nc = -70

'Noise Floor at the Receive Side (dBmp)

Nfor = -64

'Room Noise at the Send Side (dB(A))

Ps = 35

'Room Noise at the Receiver Side (dB(A))

Pr = 35

'Advantage Factor

A = 0

Loading = False

Compute

End If

End Sub

*** // This function used to call the compute function if the form loading is false**

Private Sub Form_Load ()

Loading = False

Compute
End Sub

***/ this is the main function of the simulation, it contain all the analytical formula**

Private Sub Compute()

On Error GoTo CalcErr

If Loading = True Then

Exit Sub

End If

Dim Nr1 As Double, Pro As Double, Pr1 As Double, Nr2 As Double, Nfo As Double,

No As Double, Nt As Double

Nr1 = Ps - Val(SLR.Text) - Val(Ds.Text) - 100 + 0.004 * (Ps - Val(SLR.Text) -

Val(RLR.Text) - Val(Ds.Text) - 14) ^ 2

Pro = Pr + 10 * Log(1 + 10 ^ (10 - Val(LSTR.Text)) / 10) / Log(10#)

Pr1 = Pro + 0.008 * (Pro - 35) ^ 2

Nr2 = Pr1 - 121 + Val(RLR.Text)

Nfo = Val(Nfor.Text) + Val(RLR.Text)

No = 10 * Log(10 ^ (Nr1 / 10) + 10 ^ (Nr2 / 10) + 10 ^ (Nc / 10) + 10 ^ (Nfo / 10)) /
Log(10)

Nt = No - Val(RLR.Text)

Dim Ro As Double

Ro = 15 - 1.5 * (Val(SLR.Text) + No)

Dim Xolr As Double, Iolr As Double

Xolr = Val(SLR.Text) + Val(RLR.Text) + 0.2 * (64 + Nt)

Iolr = 20 * ((1 + (Xolr / 8) ^ 8) ^ (1 / 8) - Xolr / 8)

Dim STMRo As Double, Ist As Double, Q As Double, G As Double, Iq As Double

```

STMRo = -10 * Log(10 ^ (-Val(STMR.Text) / 10) + 10 ^ (-Val(TELR.Text) / 10) *
Exp(-Val(T.Text) / 4)) / Log(10)
Ist = 12 * (1 + ((STMRo - 13) / 6) ^ 8) ^ (1 / 8) - 28 * (1 + ((STMRo + 1) / 19.4) ^ 35) ^
(1 / 35) - 13 * (1 + ((STMRo - 3) / 33) ^ 13) ^ (1 / 13) + 29
Q = 37 - 15 * Log(Val(qdu.Text)) / Log(10)
G = 1.07 + 0.258 * Q + 0.0602 * Q ^ 2
Iq = 15 * Log(1 + 10 ^ ((Ro - 100) / 15) * 10 ^ (46 / 8.4 - G / 9) + 10 ^ (46 / 30 - G / 40))
/ Log(10)

```

```

Dim Isyn As Double

```

```

Isyn = Iolr + Ist + Iq

```

```

Dim TERVist As Double, TERVsec As Double, TERV As Double

```

```

TERVist = Val(TELR.Text) + 6 * Exp(-0.3 * Val(T.Text) ^ 2) - 40 * Log((1 +
Val(T.Text) / 10) / (1 + Val(T.Text) / 150)) / Log(10)

```

```

TERVsec = TERVist + 0.5 * Ist

```

```

If (Val(STMR.Text) >= 9) Then

```

```

    TERV = TERVist

```

```

Else

```

```

    TERV = TERVsec

```

```

End If

```

```

Dim Re As Double, Roe As Double, Xdt As Double

```

```

Re = 80 + 2.5 * (TERV - 14)

```

```

Roe = -1.5 * (No - Val(RLR.Text))

```

```

Xdt = (Roe - Re) / 2

```

```

Dim IdteIst As Double, Idte As Double

```

```

IdteIst = (1 - Exp(-Val(T.Text))) * ((Xdt + (Xdt ^ 2 + 100) ^ 0.5) - 1)

```

```

If Val(STMR.Text) > 20 Then

```

```

    Idte = IdteIst

```

Else

$Idte = (IdteIst^2 + Ist^2)^{0.5}$

End If

Dim Rle As Double, Xdl As Double, Idle As Double

$Rle = 10.5 * (Val(WEPL.Text) + 7) * (Val(Tr.Text) + 1)^{-1/4}$

$Xdl = (Ro - Rle) / 2$

$Idle = Xdl + (Xdl^2 + 169)^{0.5}$

Dim X As Double, Idd As Double

If Val(Ta.Text) = 0 Then

X = 0

Else

$X = \text{Log}(Val(Ta.Text) / 100)$

End If

If (Val(Ta.Text) <= 100) Then

Idd = 0

Else

$Idd = 25 * ((1 + X^6)^{1/6} - 3 * (1 + (X/3)^6)^{1/6} + 2)$

End If

Dim Id As Double

$Id = Idte + Idle + Idd$

Dim Ieef As Double

$Ieef = Val(Ie.Text) + (95 - Val(Ie.Text)) * (Val(Ppl.Text) / (Val(Ppl.Text) + Val(Bpl.Text)))$

Dim RValue As Double

```
RValue = CInt((Ro - Isyn - Id - Ieef + Val(A.Text)) * 10 + 0.5) / 10
R.Caption = Trim(Str(RValue))
```

```
Exit Sub
```

```
CalcErr:
```

```
MsgBox "Error detected during calculation. Please verify parameter values in  
Compute.", vbExclamation
```

```
End Sub
```

***// All the following functions used for controlling the parameters**

```
Private Sub Dr_LostFocus()
```

```
Dr.Text = Trim(Dr.Text)
```

```
If (Dr.Text = "") Or (Dr.Text = "-") Then
```

```
Exit Sub
```

```
End If
```

```
If (Val(Dr.Text) < -3) Or (Val(Dr.Text) > 3) Or (IsNumeric(Dr.Text) = False) Then
```

```
Dr.SetFocus
```

```
MsgBox "Invalid value for Dr", vbExclamation
```

```
End If
```

```
LSTR.Text = Trim(Str(Round(Val(STMR.Text) + Val(Dr.Text))))
```

```
Compute
```

```
Exit Sub
```

```
End Sub
```

```
Private Sub Ds_LostFocus()
```

```
Ds.Text = Trim(Ds.Text)
```

```
If (Ds.Text = "") Or (Ds.Text = "-") Then
```

```
Exit Sub
```

```

End If
If (Val(Ds.Text) < -3) Or (Val(Ds.Text) > 3) Or (IsNumeric(Ds.Text) = False) Then
    Ds.SetFocus
    MsgBox "Invalid value for Ds", vbExclamation
End If
Compute
Exit Sub
End Sub

```

```

Private Sub Ie_LostFocus()
Ie.Text = Trim(Ie.Text)
If (Ie.Text = "") Or (Ie.Text = "-") Then
    Exit Sub
End If
If (Val(Ie.Text) < 0) Or (Val(Ie.Text) > 40) Or (IsNumeric(Ie.Text) = False) Then
    Ie.SetFocus
    MsgBox "Invalid value for Ie", vbExclamation
End If
Compute
Exit Sub
End Sub

```

```

Private Sub LSTR_LostFocus()
LSTR.Text = Trim(LSTR.Text)
If (LSTR.Text = "") Or (LSTR.Text = "-") Then
    Exit Sub
End If
If (Val(LSTR.Text) < 13) Or (Val(LSTR.Text) > 23) Or (IsNumeric(LSTR.Text) =
False) Then

```

```
LSTR.SetFocus
MsgBox "Invalid value for LSTR", vbExclamation
End If
Compute
Exit Sub
End Sub
```

```
Private Sub Nc_LostFocus()
Nc.Text = Trim(Nc.Text)
If (Nc.Text = "") Or (Nc.Text = "-") Then
    Exit Sub
End If
If (Val(Nc.Text) < -80) Or (Val(Nc.Text) > -40) Or (IsNumeric(Nc.Text) = False) Then
    Nc.SetFocus
    MsgBox "Invalid value for Nc", vbExclamation
End If
Compute
Exit Sub
End Sub
```

```
Private Sub Nfor_LostFocus()
Nfor.Text = Trim(Nfor.Text)
If (Nfor.Text = "") Or (Nfor.Text = "-") Then
    Exit Sub
End If
If (IsNumeric(Nfor.Text) = False) Then
    Nfor.SetFocus
    MsgBox "Invalid value for Nfor", vbExclamation
End If
Compute
```

Exit Sub

End Sub

Private Sub OLR_LostFocus()

OLR.Text = Trim(OLR.Text)

If (OLR.Text = "") Or (OLR.Text = "-") Then

Exit Sub

End If

If (IsNumeric(OLR.Text) = False) Then

OLR.SetFocus

MsgBox "Invalid value for OLR", vbExclamation

End If

Compute

Exit Sub

End Sub

Private Sub Ppl_LostFocus()

Ppl.Text = Trim(Ppl.Text)

If (Ppl.Text = "") Or (Ppl.Text = "-") Then

Exit Sub

End If

If (Val(Ppl.Text) < 0) Or (Val(Ppl.Text) > 20) Or (IsNumeric(Ppl.Text) = False) Then

Ppl.SetFocus

MsgBox "Invalid value for Ppl", vbExclamation

End If

Compute

Exit Sub

End Sub

```
End If
Compute
Exit Sub
End Sub
```

```
Private Sub Pr_LostFocus()
Pr.Text = Trim(Pr.Text)
If (Pr.Text = "") Or (Pr.Text = "-") Then
Exit Sub
End If
If (CInt(Val(Pr.Text)) <> Val(Pr.Text)) Or (Val(Pr.Text) < 35) Or (Val(Pr.Text) > 85)
Or (IsNumeric(Pr.Text) = False) Then
Pr.SetFocus
MsgBox "Invalid value for Pr", vbExclamation
End If
Compute
Exit Sub
End Sub
```

```
Private Sub Ps_LostFocus()
Ps.Text = Trim(Ps.Text)
If (Ps.Text = "") Or (Ps.Text = "-") Then
Exit Sub
End If
If (CInt(Val(Ps.Text)) <> Val(Ps.Text)) Or (Val(Ps.Text) < 35) Or (Val(Ps.Text) > 85)
Or (IsNumeric(Ps.Text) = False) Then
Ps.SetFocus
MsgBox "Invalid value for Ps", vbExclamation
End If
Compute
```

Exit Sub

End Sub

Private Sub qdu_LostFocus()

qdu.Text = Trim(qdu.Text)

If (qdu.Text = "") Or (qdu.Text = "-") Then

Exit Sub

End If

If (Val(qdu.Text) < 1) Or (Val(qdu.Text) > 14) Or (IsNumeric(qdu.Text) = False) Then

qdu.SetFocus

MsgBox "Invalid value for qdu", vbExclamation

End If

Compute

Exit Sub

End Sub

Private Sub RLR_LostFocus()

OLR.Text = Trim(OLR.Text)

If (OLR.Text = "") Or (OLR.Text = "-") Then

Exit Sub

End If

If (IsNumeric(OLR.Text) = False) Then

OLR.SetFocus

MsgBox "Invalid value for OLR", vbExclamation

End If

Compute

Exit Sub

End Sub

```

If (Val(SLR.Text) < 0) Or (Val(SLR.Text) > 18) Or (IsNumeric(SLR.Text) = False)
Then
    SLR.SetFocus
    MsgBox "Invalid value for SLR", vbExclamation
End If
OLR.Text = Trim(Str(Round(Val(RLR.Text) + Val(SLR.Text))))
Compute
Exit Sub
End Sub

```

```

Private Sub STMR_LostFocus()
STMR.Text = Trim(STMR.Text)
If (STMR.Text = "") Or (STMR.Text = "-") Then
    Exit Sub
End If
If (Val(STMR.Text) < 10) Or (Val(STMR.Text) > 20) Or (IsNumeric(STMR.Text) =
False) Then
    STMR.SetFocus
    MsgBox "Invalid value for STMR", vbExclamation
End If
LSTR.Text = Trim(Str(Round(Val(STMR.Text) + Val(Dr.Text))))
Compute
Exit Sub
End Sub

```

```

Private Sub T_LostFocus()
T.Text = Trim(T.Text)
If (T.Text = "") Or (T.Text = "-") Then

```

```

Exit Sub
End If
If (Val(T.Text) < 0) Or (Val(T.Text) > 500) Or (IsNumeric(T.Text) = False) Then
    T.SetFocus
    MsgBox "Invalid value for T", vbExclamation
End If
Tr.Text = Trim(Val(T.Text) * 2)
Ta.Text = T.Text
Compute
Exit Sub
End Sub

```

```

Private Sub Ta_LostFocus()
    Ta.Text = Trim(Ta.Text)
    If (Ta.Text = "") Or (Ta.Text = "-") Then
        Exit Sub
    End If
    If (Val(Ta.Text) < 0) Or (Val(Ta.Text) > 500) Or (IsNumeric(Ta.Text) = False) Then
        Ta.SetFocus
        MsgBox "Invalid value for Ta", vbExclamation
    End If
    Compute
    Exit Sub
End Sub

```

```

Private Sub TELR_LostFocus()
    TELR.Text = Trim(TELR.Text)
    If (TELR.Text = "") Or (TELR.Text = "-") Then
        Exit Sub
    End If

```

```

End If
If (Val(TEL.R.Text) < 5) Or (Val(TEL.R.Text) > 65) Or (IsNumeric(TEL.R.Text) = False)
Then
    TEL.R.SetFocus
    MsgBox "Invalid value for TELR", vbExclamation
End If
Compute
Exit Sub
End Sub

```

```

Private Sub Tr_LostFocus()
Tr.Text = Trim(Tr.Text)
If (Tr.Text = "") Or (Tr.Text = "-") Then
    Exit Sub
End If
If (Val(Tr.Text) < 0) Or (Val(Tr.Text) > 1000) Or (IsNumeric(Tr.Text) = False) Then
    Tr.SetFocus
    MsgBox "Invalid value for Tr", vbExclamation
End If
Compute
Exit Sub
End Sub

```

```

Private Sub WEPL_LostFocus()
WEPL.Text = Trim(WEPL.Text)
If (WEPL.Text = "") Or (WEPL.Text = "-") Then
    Exit Sub
End If
If (Val(WEPL.Text) < 5) Or (Val(WEPL.Text) > 110) Or (IsNumeric(WEPL.Text) =
False) Then

```

WEPL.SetFocus

MsgBox "Invalid value for WEPL", vbExclamation

End If

Compute

Exit Sub

End Sub

*// This Function used to close the program

Private Sub Command1_Click()

Unload Me

End Sub
