NATIONAL OPEN UNIVERSITY OF NIGERIA

SCHOOL OF SCIENCE AND TECHNOLOGY

COURSE CODE: CIT305

COURSE TITLE:
NETWORKING AND COMMUNICATION TECHNOLOGY

COURSE GUIDE

CIT305
NETWORKING AND COMMUNICATION TECHNOLOGY

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Introduction

CIT305 – Networking and Communication Technology is a three [3] credit unit course of fifteen units. It teaches the various forms of networking, network design, communication technology used by people to accomplish different organisational or individual task.

It also gives an insight into various forms of computer networking ranging from *LAN* to *WAN* and even go as far as looking into the wireless networks that are in use in today's technology. The course also explains the enterprise network, which forms a branch computer networking technology.

What You Will Learn in This Course

The main purpose of this course is to provide the necessary tools for designing and managing *information systems*. It makes available the steps and tools that will enable you to make proper and accurate decision on database designs and operations whenever the need arises.

Course Aims

This course sets out to achieve the following aims:

- introduce the concepts associated with information systems development;
- provide necessary tools for analysing, designing, developing a database of any size;
- provide you with the necessary foundation in Database programming
- introduction of web services and their architectural frameworks; and

Course Objectives

Certain objectives have been set out to ensure that the course achieves its aims. Apart from the course objectives, every unit of this course has set objectives. In the course of the study, you will need to confirm, at the end of each unit, if you have met the objectives set at the beginning of each unit. By the end of this course you should be able to:

- Explain the fundamental of Networking
- define the term "data links"
- explain network protocols

- discuss what is involved in the building of internetworks using *TCP/IP* and routers
- explain the network standards (IEEE 802 Standards)
- state the fundamentals of enterprise network
- discuss what is involved in signal transmission and impairment
- explain digital technology
- describe the concept of packet switching

Working through This Course

In order to have a thorough understanding of the study units, you will need to read and understand the contents, practise the steps by designing an information system of your own, and be committed to learning and implementing your knowledge.

This course is designed to cover approximately seventeen weeks, and it will require your devoted attention. You should do the exercises in the Tutor-Marked Assignments (TMAs) and submit to your tutors.

Course Materials

These include the list below:

1. Course guide
2. Study units
3. Recommended texts
4. A file for your assignments and for records to monitor your progress.

Study Units

There are Twenty three study units in this course- as listed below.

Module 1 Networking

Unit 1	Introduction to Networking
Unit 2	Data Links
Unit 3	Deploying Physical Media
Unit 4	Network Protocols

Module 2 Network Design

Unit 1	Harnessing Wifi for User Mobility
Unit 2	Building Internetworks Using Tcp/Ip and Routers
Unit 3	Network Standards (Ieee 802 Standards)
Unit 4	Implementing Security Best Practices

Module 3	**Enterprise Network**
Unit 1	Creating Enterprise Network
Unit 2	Planning and Selection of Enterprise Network
Unit 3	*Lan* and *Wan*

Module 4	**Communication Technology**
Unit 1	Modem and Modulation Concept
Unit 2	Multiplexers
Unit 3	Digital Technologies
Unit 4	Signal Transmission and Impairment

Module 5	**Network Technologies**
Unit 1	*ISDN*
Unit 2	*DSL*
Unit 3	*SONET*
Unit 4	Packet Switching
Unit 5	Internet and *TCP/IP*

Make use of the course materials, do the exercises to enhance your learning.

Assignments File

These are of two types- the self-assessment exercises and the Tutor-Marked Assignments (TMAs). The self-assessment exercises will enable you monitor your performance by yourself, while the Tutor-Marked Assignment (TMA) is a supervised assignment. The assignments take a certain percentage of your total score in this course. The tutor-marked assignments will be assessed by your tutor within a specified period. The examination at the end of this course will aim at determining the level of mastery of the subject matter. This course includes twelve tutor-marked assignments and each must be done and submitted accordingly. Your best scores however, will be recorded for you. Be sure to send these assignments to your tutor before the deadline to avoid loss of marks.

Tutor-Marked Assignments (TMAS)

There are twelve tutor-marked assignments in this course. You need to submit all the assignments. The total marks for the best four (4) assignments will be 30% of your total course mark.

Assignment questions for the units in this course are contained in the Assignment File. You should be able to complete your assignments from the information and materials contained in your set textbooks, reading and study units. However, you may wish to use other references to broaden your viewpoint and provide a deeper understanding of the subject.

When you have completed each assignment, send it together with form to your tutor. Make sure that each assignment reaches your tutor on or before the deadline given. If, however, you cannot complete your work on time, contact your tutor before the assignment is done to discuss the possibility of an extension.

Examination and Grading

The final examination for the course will carry 70% percentage of the total marks available for this course. The examination will cover every aspect of the course, so you are advised to revise all your corrected assignments before the examination.

This course endows you with the status of a teacher and that of a learner. This means that you teach yourself and that you learn, as your learning capabilities would allow. It also means that you are in a better position to determine and to ascertain the what, the how, and the when of your language learning. No teacher imposes any method of learning on you.

The course units are similarly designed with the introduction following the table of contents, then a set of objectives and then the dialogue and so on.

The objectives guide you as you go through the units to ascertain your knowledge of the required terms and expressions.

Presentation Schedule

This gives you the important dates for the completion of tutor-marked assignments and tutorials. Remember, you are required to submit all your assignments by the due date. You should guard against lagging behind in your work.

Course Marking Scheme

This table shows the breakdown of the actual marking scheme for the course.

Assessment	Marks
Assignment 1- 4	Four assignments, best three marks of the four count at 30% of course marks
Final Examination	70% of overall course marks
Total	100% of course marks

Course Overview

Unit	Title of Work	Weeks Activity	Assessment (End of Unit)
	Course Guide	Week 1	
	Module 1		
1	Introduction to Networking	Week 1	Assignment 1
2	Data Links	Week 1	Assignment 1
3	Deploying Physical Media	Week 2	Assignment 2
4	Network Protocols	Week 3	Assignment 3
	Module 2		
1	Harnessing Wi-Fi for user mobility	Week 4	Assignment 4
2	Building Internetworks using TCP/IP and Routers	Week 5	Assignment 5
3	Network Standards (IEEE 802 Standards)	Week 6	Assignment 6
4	Implementing Security best practices	Week 7	Assignment 7
	Module 3		
1	Creating Enterprise Network	Week 8	Assignment 8
2	Planning and Selection of Enterprise Network		
3	Advanced *WAN* and *LAN* Classes	Week 9	Assignment 9
	Module 4		
1	Modem and Modulation Concepts	Week 10	Assignment 10
2	Multiplexers	Week 11	Assignment 11
3	Digital Technologies	Week 12	Assignment 12
	Module 5		
1	Integrated Services Digital Network (ISDN)	Week 13	Assignment 12
2	Digital Subscriber Line (DSL)	Week 14	Assignment 13
3	Synchronous Optical Network (SONET)	Week 15	Assignment 14
4	Packet Switching	Week 16	Assignment 15
5	Internet and *TCP/IP*	Week 16	Assignment 16

	Revision	Week 17	
	Examination		
	Total	**17 Weeks**	

How to Get the Most from This Course

In distance learning the study units replace the university lecture room situation. This is one of the great advantages of distance learning; you can read and work through specially designed study materials at your own pace, and at a time and place that suit you best. Think of it as reading the lecture instead of listening to a lecturer. In the same way that a lecturer might set you some reading to do, the study units tell you when to read your set books or other material. Just as a lecturer might give you an in-class exercise, your study units provide exercises for you to do at appropriate points.

Each of the study units follows a common format. The first item is an introduction to the subject matter of the unit and how a particular unit is integrated with the other units and the course as a whole. Next is a set of learning objectives. These objectives enable you know what you should be able to do by the time you have completed the unit. You should use these objectives to guide your study. When you have finished the units you must go back and check whether you have achieved the objectives. If you make a habit of doing this you will significantly improve your chances of passing the course.

Remember that your tutor's job is to assist you. When you need help, don't hesitate to call and ask your tutor to provide it.

1. Read this *Course Guide* thoroughly.

2. Organise a study schedule. Refer to the 'course overview' for more details. Note the time you are expected to spend on each unit and how the assignments relate to the units. Whatever method you chose to use, you should decide on it and write in your own dates for working on each unit.

3. Once you have created your own study schedule, do everything you can to stick to it. The major reason that students fail is that they lag behind in their course work.

4. Assemble the study materials. Information about what you need for a unit is given in the 'overview' at the beginning of each unit. You will almost always need both the study unit you are working on and one of your set of books on your desk at the same time.

5. Work through the unit. The content of the unit itself has been arranged to provide a sequence for you to follow. As you work through the unit you will be instructed to read sections from your set books or other articles. Use the unit to guide your reading.

6. Review the objectives for each study unit to confirm that you have achieved them. If you feel unsure about any of the objectives, review the study material or consult your tutor.

7. When you are confident that you have achieved a unit's objectives, you can then start on the next unit. Proceed unit by unit through the course and try to pace your study so that you keep yourself on schedule.

8. When you have submitted an assignment to your tutor for marking, do not wait for its return before starting on the next unit. Keep to your schedule. When the assignment is returned, pay particular attention to your tutor's comments, both on the tutor-marked assignment form and also written on the assignment. Consult your tutor as soon as possible if you have any questions or problems.

9. After completing the last unit, review the course and prepare yourself for the final examination. Check that you have achieved the unit objectives (listed at the beginning of each unit) and the course objectives (listed in this *Course Guide*).

Facilitators/Tutors and Tutorials

There are 12 hours of tutorials provided in support of this course. You will be notified of the dates, times and location of these tutorials, together with the name and phone number of your tutor, as soon as you are allocated a tutorial group.

Your tutor will mark and comment on your assignments, keep a close watch on your progress and on any difficulties you might encounter and provide assistance to you during the course. You must mail or submit your tutor-marked assignments to your tutor well before the due date (at least two working days are required). They will be marked by your tutor and returned to you as soon as possible.

Do not hesitate to contact your tutor by telephone, or e-mail if you need help. The following may be circumstances in which you would find help necessary. Contact your tutor if:

- you do not understand any part of the study units or the assigned readings,
- you have difficulty with the self-tests or exercises,
- you have a question or problem with an assignment, with your tutor’s comments on an assignment or with the grading of an assignment.

You should try your best to attend the tutorials. This is the only chance to have face to face contact with your tutor and to ask questions which are answered instantly. You can raise any problem encountered in the course of your study. To gain the maximum benefit from course tutorials, prepare a question list before attending them. You will learn a lot from participating in discussions actively.

Summary

Networking and Communication Technology introduces you to the concepts associated with computer network development which is critical in understanding the various computer technology and data communications technology. The content of the course material has been planned and written to ensure that you acquire the proper knowledge and skills for the appropriate situations. The essence is to help you in acquiring the necessary knowledge and competence by equipping you with the necessary tools to accomplish this.

We hope that by the end of this course you would have acquired the required knowledge to view Computer Network in a new way.

I wish you success with the course and hope that you will find it both interesting and useful.

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MODULE 1 NETWORKING

UNIT 1 INTRODUCTION TO NETWORKING

CONTENTS

1.0 INTRODUCTION

Having read through the course guide, you will have a general understanding of what this unit is about and how it fits into the course as a whole. This unit will describe the general concept of networking, types and its application areas.

2.0 OBJECTIVES

At the end this unit, you should be able to:

- explain the term networking system
- identify the various types of networking
- highlight the history of networking
- describe the areas of work of networking.

3.0 MAIN CONTENT

3.1 Overview

The concept of a network is pretty simple. A couple of computers have some cable strung between them, and send data back and forth using electrical signaling over the cable. More or less the same as telephones do or, in a very rough sense, like two kids speaking into tin cans connected by a string; but how does the data actually move from computer A to computer B? How does computer A find the physical location of computer B on the network? If they communicate with electrical signaling, so the data is traveling "at the speed of light"; why does it take so long to send a big file across the network?

The internet is not one single entity; it is a massive interconnection of hosts such as your computer, and another- halfway across the world. We understand that our Internet Service Providers (ISPs) provide internet access to our host, but it may not necessarily provide service to the server that you are communicating with. In this case, how does information make its way from your host to the other host and vice versa?

The answer lies in the interconnection of many *ISPs* themselves. *ISPs* can be categorised into Tier 1, 2 and 3 *ISPs*. Tier 1 *ISPs* are major internet service providers that usually sell access to smaller Tier 2 *ISPs*. Tier 2 *ISPs* may service entire countries or cities, but not the rest of the world. Tier 3 *ISPs* are also customers of Tier 2 *ISPs*, and usually service end users such as you. *ISPs* peer with each other to allow data from your host to reach the other host. From this, we understand that data passes through multiple ISPs in order to be delivered from one location to another.

3.2 Definition of Networking

A computer network, often simply referred to as a network, is a collection of computers and devices interconnected by communications channels that facilitate communications among users and allows users to share resources.

A network can as well be define as a collection of computers or other hardware devices that are connected together, either physically or logically, using special hardware and software, to allow them to exchange information and cooperate. Networking is the term that describes the processes involved in designing, implementing, upgrading, managing and otherwise working with networks and network technologies.

3.3 History of Computer Networks

Before the advent of computer networks that were based upon some type of telecommunications system, communication between calculation machines and early computers was performed by human users by carrying instructions between them. Many of the social behaviours seen in today's Internet were demonstrably present in the nineteenth century and arguably in even earlier networks using visual signals.

In September 1940, George Stibitz used a teletype machine to send instructions for a problem set from his Model at Dartmouth College to his Complex Number Calculator in New York and received results back by the same means. Linking output systems like teletypes to computers was an interest at the Advanced Research Projects Agency (ARPA) when, in 1962, J.C.R. Licklider was hired and developed a working group he called the "Intergalactic Network", a precursor to the ARPANET.

In 1964, researchers at Dartmouth developed the Dartmouth Time Sharing System for distributed users of large computer systems. The same year, at Massachusetts Institute of Technology, a research group supported by General Electric and Bell Labs used a computer to route and manage telephone connections.

Throughout the 1960s, Leonard Kleinrock, Paul Baran and Donald Davies independently conceptualised and developed network systems which used packets that could be used in a network between computer systems. In 1965, Thomas Merrill and Lawrence G. Roberts created the first wide area network (WAN).

The first widely used telephone switch that used true computer control was introduced by Western Electric in 1965. In 1969, the University of California at Los Angeles, the Stanford Research Institute, University of California at Santa Barbara, and the University of Utah were connected as the beginning of the ARPANET network using 50 kbit/s circuits. Commercial services using X.25 were deployed in 1972, and later used as an underlying infrastructure for expanding TCP/IP networks.

Today, computer networks are the core of modern communication. All modern aspects of the Public Switched Telephone Network (PSTN) are computer-controlled, and telephony increasingly runs over the Internet Protocol, although not necessarily the public Internet. The scope of communication has increased significantly in the past decade, and this boom in communications would not have been possible without the progressively advancing computer network. Computer networks and the technologies needed to connect and communicate through and between them, continue to drive computer hardware, software, and peripherals industries. This expansion is mirrored by growth in the numbers and types of users of networks from the researcher to the home user.

3.4 Purpose of Computer Networks

Computer networks can be used for a variety of purposes; these are listed below.

i. Facilitating communications

Using a network, people can communicate efficiently and easily via email, instant messaging, chat rooms, telephone, video telephone calls, and video conferencing.

ii. Sharing hardware

In a networked environment, each computer on a network may access and use hardware resources on the network, such as printing a document on a shared network printer.

iii. Sharing files, data, and information

In a network environment, authorized user may access data and information stored on other computers on the network. The capability of providing access to data and information on shared storage devices is an important feature of many networks.

iv. Sharing software

Users connected to a network may run application programs on remote computers.

3.5 Network Classification

The following list presents categories used for classifying networks.

a. Connection method

Computer networks can be classified according to the hardware and software technology that is used to interconnect the individual devices in the network, such as optical fiber, Ethernet, wireless *LAN*, *HomePNA*, power line communication or *G.hn*.

Ethernet, as it is defined by *IEEE* 802, utilises various standards and mediums that enable communication between devices. Frequently deployed devices include hubs, switches, bridges, or routers. Wireless *LAN* technology is designed to connect devices without wiring. These devices use radio waves or infrared signals as a transmission medium. *ITU-T G.hn* technology uses existing home wiring (coaxial cable, phone lines and power lines) to create a high-speed (up to 1 *Gigabit/s*) local area network.

b. Wired technologies

Twisted pair wire is the most widely used medium for telecommunication. Twisted-pair cabling consist of copper wires that are twisted into pairs. Ordinary telephone wires consist of two insulated copper wires twisted into pairs. Computer networking cabling consist of 4 pairs of copper cabling that can be utilised for both voice and data transmission. The use of two wires twisted together helps to reduce crosstalk and electromagnetic induction. The transmission speed ranges from 2 million bits per second to 100 million bits per second. Twisted pair cabling comes in two forms which are Unshielded Twisted Pair (UTP) and Shielded twisted-pair (STP) which are rated in categories which are manufactured in different increments for various scenarios.

Coaxial cable is widely used for cable television systems, office buildings, and other work-sites for local area networks. The cables consist of copper or aluminum wire wrapped with insulating layer typically of a flexible material with a high dielectric constant, all of which are surrounded by a conductive layer. The layers of insulation help minimise interference and distortion. Transmission speed range from 200 million to more than 500 million bits per second.

Optical fiber cable consists of one or more filaments of glass fiber wrapped in protective layers. It transmits light which can travel over extended distances. Fiber-optic cables are not affected by electromagnetic radiation. Transmission speed may reach trillions of bits per second. The transmission speed of fiber optics is hundreds of times faster than for coaxial cables and thousands of times faster than a twisted-pair wire.

SELF-ASSESSMENT EXERCISE 1

Identify the key problems with wired media.

c. Wireless technologies

Take note of the following.

Terrestrial microwave – terrestrial microwaves use Earth-based transmitter and receiver. The equipment looks similar to satellite dishes. Terrestrial microwaves use low-gigahertz range, which limits all communications to line-of-sight. Path between relay stations spaced approx, 30 miles apart. Microwave antennas are usually placed on top of buildings, towers, hills, and mountain peaks.

Communications satellites – the satellites use microwave radio as their telecommunications medium which are not deflected by the Earth's atmosphere. The satellites are stationed in space, typically 22,000 miles (for geosynchronous satellites) above the equator. These Earth-orbiting systems are capable of receiving and relaying voice, data, and TV signals.

Cellular and PCS systems – use several radio communications technologies. The systems are divided to different geographic areas. Each area has a low-power transmitter or radio relay antenna device to relay calls from one area to the next area.

Wireless LANs – wireless local area network use a high-frequency radio technology similar to digital cellular and a low-frequency radio technology. Wireless LANs use spread spectrum technology to enable communication between multiple devices in a limited area. An example of open-standards wireless radio-wave technology is *IEEE*.

Infrared communication- this can transmit signals between devices within small distances not more than 10 meters peer to peer or (face to face) without anybody in the line of transmitting.

3.6 Types of Networks Based on Physical Scope

Common types of computer networks may be identified by their scale.

a. Local area network

A local area network (LAN) is a network that connects computers and devices in a limited geographical area such as home, school, computer laboratory, office building, or closely positioned group of buildings. Each computer or device on the network is a node. Current wired *LANs* are most likely to be based on Ethernet technology, although new standards like *ITU-T G.hn* also provide a way to create a wired *LAN* using existing home wires (coaxial cables, phone lines and power lines).

Typical library network, in a branching tree topology and controlled access to resources All interconnected devices must understand the network layer (layer 3), because they are handling multiple subnets (the different colors).

Those inside the library, which have only 10/100 Mbit/s Ethernet connections to the user device and a Gigabit Ethernet connection to the central router, could be called "layer 3 switches" because they only have Ethernet interfaces and must understand *IP*. It would be more correct to call them access routers, where the router at the top is a distribution router that connects to the Internet and academic networks' customer access routers. The defining characteristics of *LANs*, in contrast to *WANs* (Wide Area Networks), include their higher data transfer rates, smaller geographic range, and no need for leased telecommunication lines. Current Ethernet or other IEEE 802.3 LAN technologies operate at speeds up to 10 Gbit/s. This is the data transfer rate. IEEE has projects investigating the standardization of 40 and 100 Gbit/s.

b. Personal area network

A Personal Area Network (PAN) is a computer network used for communication among computer and different information technological devices close to one person. Some examples of devices that are used in a *PAN* are personal computers, printers, fax machines, telephones, *PDAs*, scanners, and even video game consoles. A *PAN* may include wired and wireless devices. The reach of a *PAN* typically extends to 10 meters. A wired *PAN* is usually constructed with *USB* and Firewire connections while technologies such as Bluetooth and infrared communication typically form a wireless *PAN*.

c. Home area network

A Home Area Network (HAN) is a residential *LAN* which is used for communication between digital devices typically deployed in the home, usually a small number of personal computers and accessories, such as printers and mobile computing devices. An important function is the sharing of Internet access, often a broadband service through a *CATV* or Digital Subscriber Line (DSL) provider. It can also be referred to as an office area network (OAN).

d. Wide area network

A Wide Area Network (WAN) is a computer network that covers a large geographic area such as a city, country, or spans even intercontinental distances, using a communications channel that combines many types of media such as telephone lines, cables, and air waves. A *WAN* often uses transmission facilities provided by common carriers, such as telephone companies. *WAN* technologies generally function at the lower three layers of the *OSI* reference model: the physical layer, the data link layer, and the network layer.

e. Campus network

A campus network is a computer network made up of an interconnection of local area networks (LAN's) within a limited geographical area. The networking equipments (switches, routers) and transmission media (optical fiber, copper plant, Cat5 cabling etc.) are almost entirely owned (by the campus tenant / owner: an enterprise, university, government etc.).

In the case of a university campus-based campus network, the network is likely to link a variety of campus buildings including; academic departments, the university library and student residence halls.

f. Metropolitan area network

A Metropolitan area network is a large computer network that usually spans a city or a large campus.

g. Enterprise private network

An enterprise private network is a network build by an enterprise to interconnect various company sites, e.g., production sites, head offices, remote offices, shops, in order to share computer resources.

h. Virtual private network

A Virtual Private Network (VPN) is a computer network in which some of the links between nodes are carried by open connections or virtual circuits in some larger network (e.g., the Internet) instead of by physical wires. The data link layer protocols of the virtual network are said to be tunneled through the larger network when this is the case. One common application is secured communications through the public internet; but a *VPN* need not have explicit security features, such as authentication or content encryption. *VPNs*, for example, can be used to separate the traffic of different user communities over an underlying network with strong security features.

VPN may have best-effort performance, or may have a defined service level agreement (SLA) between the *VPN* customer and the *VPN* service provider. Generally, a *VPN* has a topology more complex than point-to-point.

i. Internetwork

An internetwork is the connection of two or more private computer networks via a common routing technology (OSI Layer 3) using routers. The internet is an aggregation of many internetworks; hence its name has been shortened to internet.

j. Backbone network

A Back-Bone Network (BBN)-or network backbone, is part of a computer network infrastructure that interconnects various pieces of network, providing a path for the exchange of information between different *LANs* or sub networks. A backbone can tie together diverse networks in the same building, in different buildings in a campus environment, or over wide areas. Normally, the backbone's capacity is greater than the networks connected to it.

A large corporation that has many locations may have a backbone network that ties all of the locations together, for example, if a server cluster needs to be accessed by different departments of a company that are located at different geographical locations. The pieces of the network connections (for example: Ethernet, wireless) that bring these departments together is often mentioned as network backbone. Network congestion is often taken into consideration while designing backbones.
Backbone networks should not be confused with the internet backbone.

k. Global area network

A Global Area Network (GAN) is a network used for supporting mobile communications across an arbitrary number of wireless *LANs*, satellite coverage areas, etc. The key challenge in mobile communications is handing off the user communications from one local coverage area to the next. In *IEEE* Project 802, this involves a succession of terrestrial wireless *LANs*.

l. Internet

The internet is a global system of interconnected governmental, academic, corporate, public, and private computer networks. It is based on the networking technologies of the Internet Protocol Suite. It is the successor of the Advanced Research Projects Agency Network (ARPANET) developed by *DARPA* of the United States Department of Defense. The Internet is also the communications backbone underlying the World Wide Web (WWW).

Participants in the internet use a diverse array of methods of several hundred documented, and often standardised, protocols compatible with the Internet Protocol Suite and an addressing system (IP addresses) administered by the Internet Assigned Numbers Authority and address registries. Service providers and large enterprises exchange information about the reachability of their address spaces through the Border Gateway Protocol (BGP), forming a redundant worldwide mesh of transmission paths.

m. Intranets and extranets

Intranets and extranets are parts or extensions of a computer network, usually a local area network.

An intranet is a set of networks, using the Internet Protocol and IP-based tools, such as web browsers and file transfer applications, which are under the control of a single administrative entity. That administrative entity closes the intranet to all but specific, authorised users. Most commonly, an intranet is the internal network of an organisation. A large intranet will typically have at least one web server to provide users with organisational information.

An extranet is a network that is limited in scope to a single organisation or entity and also has limited connections to the networks of one or more other usually, but not necessarily, trusted organisations or entities—a company's customers may be given access to some part of its intranet—while at the same time the customers may not be considered trusted from a security standpoint. Technically, an extranet may also be categorised as a *CAN*, *MAN*, *WAN*, or other type of network, although an extranet cannot consist of a single *LAN*; it must have at least one connection with an external network.

n. Overlay network

An overlay network is a virtual computer network that is built on top of another network. Nodes in the overlay are connected by virtual or logical links, each of which corresponds to a path, perhaps through many physical links, in the underlying network.

A sample overlay network: *IP* over *SONET* over *Optical*

For example, many peer-to-peer networks are overlay networks because they are organised as nodes of a virtual system of links run on top of the Internet. The Internet was initially built as an overlay on the telephone network. Overlay networks have been around since the invention of networking when computer systems were connected over telephone lines using modem, before any data network existed.

Nowadays the internet is the basis for many overlaid networks that can be constructed to permit routing of messages to destinations specified by an *IP* address. For example, distributed hash tables can be used to route messages to a node having a specific logical address, whose *IP* address is known in advance.

Overlay networks have also been proposed as a way to improve internet routing, such as through quality of service guarantees to achieve higher-quality streaming media. Previous proposals such as *IntServ*, *DiffServ*, and *IP Multicast* have not seen wide acceptance largely because they require modification of all routers in the network. On the other hand, an overlay network can be incrementally deployed on end-hosts running the overlay protocol software, without cooperation from Internet service providers. The overlay has no control over how packets are routed in the underlying network between two overlay nodes, but it can control, for example, the sequence of overlay nodes a message traverses before reaching its destination.

SELF-ASSESSMENT EXERCISE 2

What is the relevance of *ISP* in internet connections?

3.7 Introduction to Telecommunication Circuit

A **telecommunication circuit** is any line, conductor, or other conduit by which information is transmitted. A **dedicated circuit**, **private circuit**, or **leased line** is a line that is dedicated to only one use. Originally, this was analog, and was often used by radio stations as a studio/transmitter link (STL) or remote pickup unit (RPU) for their audio, sometimes as a backup to other means. Later lines were digital, and used for private corporate data networks.

A **telecommunication circuit** may be defined as follows.

- The complete path between two terminals over which one-way or two-way communications may be provided. See communications protocol.
- An electronic path between two or more points, capable of providing a number of channels.
- A number of conductors connected together for the purpose of carrying an electric current.
- An electronic closed-loop path among two or more points used for signal transfer.
- A number of electrical components, such as resistors, inductances, capacitors, transistors, and power sources connected together in one or more closed loops.

4.0 CONCLUSION

In this unit, you have been introduced to the fundamentals of Networking. You have also learnt the different types of Networking and its classification. You are advised to go over this unit again since it is the basis for understanding the course.

5.0 SUMMARY

In this unit, you have learnt that:

- computer network, often simply referred to as a network, is a collection of computers and devices interconnected by communications channels that facilitate communications among users and allows users to share resources.
- a telecommunication circuit is any line, conductor, or other conduit by which information is transmitted.
- types of telecommunication circuit are dedicated circuit, private circuit, or leased line

6.0 TUTOR-MARKED ASSIGNMENT

1. Briefly describe the following:-

i. Internetwork
ii. Backbone Network
iii. Intranet

2. State and explain all the types of networks based on their physical scope.

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UNIT 2 DATA LINKS

CONTENTS

1.0 INTRODUCTION

In the previous unit, you have been introduced to the basic concept of networking. In this unit, you will learn about another concept of networking- which is data link. In telecommunication, a **data link** is the means of connecting one location to another for the purpose of transmitting and receiving digital information. It can also refer to a set of electronics assembly, consisting of a transmitter and a receiver [two Data Terminal Equipments (DTEs)] and the interconnecting data telecommunication circuit. These are governed by a link protocol enabling digital data to be transferred from a data source to a data sink.

2.0 OBJECTIVES

At the end of this unit, you should be able to:

- discuss *OSI* model
- explain the term “data link”
- identify the various type of data link
- state the relevance of data link to networking.

3.0 MAIN CONTENT

3.1 Introduction to Open System Interconnection (OSI) Model

OSI is not a physical model, though, it is a set of guidelines that application developers can use to create and implement application that run on a network. It also provides a framework for creating and implementing networking standards, devices and internetworking schemes.

OSI has seven different layers, divided into two groups. The top three layers define how the applications within the end stations will communicate with each other and with other users. The bottom four layers define how data is transmitted end to end.

Table 2.1 below shows the layers of the *OSI* reference model

Table 2.1: OSI Reference Model

	Layers	**Summary of Functions**
7	Application layer	Interacts with the operating systems and applications
6	Presentation layer	Transformation of data. Responsible for functions like data compression, encryption, etc.
5	Session layer	Defines a connection from a user to a network server or from a peer on a network to another peer
4	Transport layer	Manages the flow of information from one network node to another. Ensures that packets are decoded in the proper sequence and received logically
3	Network layer	Defines how data move from one point to another on a network. It determines what goes into each packet and defines different packet protocols.
2	Data link layer	Defines standards that assign meaning to bits carried by the physical layer
1	Physical layer	Defines the properties of the physical medium used to make a network connection

The *OSI* model is fully discussed in other courses in computer network. However, data link layer is expected to be fully discussed in this course.

3.2 Data Link Layer

The data link layer is the second layer in the Open System Interconnection (OSI) reference model. The data link layer defines the rules for accessing and using the physical layer. The data link layer provides the physical transmission of the data and handles error notification, network topology and flow control. This means that the data link layer will ensure that messages are delivered to the proper device on a *LAN* using hardware addresses and will translate messages from the Network layer into bits for the physical layer to transmit.

The data link layer in the model combine packets and bytes into frames, it as well provides access to media using *MAC* address. The data link layer does perform the error detection but it should be noted that it does not perform the error correction process.

The data layer formats the message into pieces, each called a data frame, and adds a customised header containing the hardware destination and source address.

3.3 Categories of Data Link

There are at least three categories of basic data-link configurations that can be conceived of and used:

- simplex communications- referring to all communications in one direction only e.g. radio transmission.
- half-duplex communications - referring to communications in both directions, but not both ways simultaneously, e.g. walkie-talkie.
- duplex communications, communications in both directions simultaneously, e.g. telephone conversation.

3.4 Types of Data Links

Let us consider these one after the other

a. Industrial Ethernet

Industrial ethernet is used to provide data link solutions for the communications and automation industry. Traditional office grade ethernet cannot meet the reliability demanded by industrial applications. A brief loss of service in an office environment may not be such a big issue, but in an industrial environment it may represent significant loss on your capital investment.

Industrial ethernet is specifically designed to operate in harsh environments such as factory floor automation, process control, *HVAC*, medical, manufacturing. Typically, industrial automation devices include, rugged case, din rail attachment, wide temperature specification, broad power source input, these features give us a reliable ethernet connection in demanding environments.

b. Radio modems

Radio modems are radio frequency transceivers for serial data communications. They connect to serial ports *RS232*, *RS422/485* and transmit to and receive signals from other matching radio (point to point) or radios (multidrop) network.

Wireless radio modems are designed to be transparent to the systems within which they operate. All communication appears to your system as if communicating across directly connected cables; no special preparation of your data is needed. MaxStream units provide you true plug-and-communicate wireless capability operating in the internationally recognised *2.4 GHz* license free band.

c. Ethernet to *RS232, RS485* serial device servers

An ethernet to *RS232* or *RS485* device server allows your network to enable virtually any serial *RS232/422/485* port device. They provide the ability to remotely monitor, control or diagnose your equipment over your *LAN* or even *WAN* (internet/web) link. Allowing you to maintain the existing investment you have made in serial interface plant and machine equipment.

d. Wireless *RS232* link

When creating an *RS-232* wireless link, you can replace conventional expensive RS232 serial cable runs, allowing for an easy to use, invisible connection. Handy wave bluetooth is the cable replacement solution for *RS-232*. Simply plug one unit into your *RS-232* device and the other into your *PC* for an instant wireless link with minimal setup and also gives the added flexibility and mobility not available with traditional wired *RS232* links.

e. *GSM* and *GPRS*

A *GSM* modem is a wireless modem that works with a *GSM* wireless network. A wireless modem behaves like a dial-up modem. The main difference between them is that a dial-up modem sends and receives data through a fixed telephone line while a wireless modem sends and receives data through radio waves. *GPRS* modem is a *GSM* modem which, additionally, supports the *GPRS* technology for data transmission. *GPRS* stands for General Packet Radio Service. It is a packet-switched technology that is an extension of *GSM*. (*GSM* is a circuit-switched technology.) A key advantage of *GPRS* over *GSM* is that *GPRS* has a higher data transmission speed. This Technology is ideal for *M2M*(machine to machine communications) applications such as meter reading, remote maintenance, traffic control systems, vending machines and building management systems *HVAC*

f. Power Over Ethernet (POE)

Power Over Ethernet or POE technology describes any system to transmit electrical power, along with data, to remote devices over standard twisted-pair cable in an Ethernet network. The standard is *IEEE 802.3af* which calls for *48 Volts DC* over two pairs of a four-pair cable at a maximum current of *350 AMP* for a maximum load power of *16Watts*.

g. Outdoor Ethernet switches

An outdoor ethernet switch is specifically designed for the toughest industrial environments. An outdoor switch is constructed from a rugged weather tight aluminum case and the design usually carries an *IP* rating which provide a waterproof and dust-tight connection. An outdoor Ethernet switch can be easily adopted in almost all kinds of industrial applications and provides the most reliable solutions for your network in outdoor environments, typical applications includes: railway, moving vehicles, factory automation, and marine (DNV Approval).

4.0 CONCLUSION

In this unit, you have been introduced to what is referred to as data link. You have also learnt the different types of data link in use and as well as the relevance of data link to networking.

5.0 SUMMARY

In this unit, you have learnt about:

- *OSI* model- which has 7 layers: application, presentation, session, transport, network, data link and physical.
- data link layer which defines rules for accessing and using physical layer
- 7 types of data link which includes industrial Ethernet, radio modems, Ethernet to RS232, wireless RS232 link, *GSM* and *GPRS*, Power Over Ethernet (POE) and outdoor Ethernet switches.

6.0 TUTOR-MARKED ASSIGNMENT

1. What do you understand by the term “data link”?
2. How relevant is data link to networking of computers?
3. State some protocols that are defined by the data link layer.

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UNIT 3 DEPLOYING PHYSICAL MEDIA

CONTENTS

1.0 INTRODUCTION

In the previous units, you have been introduced to network and data link layer of the *OSI* reference model. This unit will introduce you to physical media used in the networking of computer and it will, as well, teach you how to deploy the physical media.

2.0 OBJECTIVES

At the end of this unit, you should be able to:

- explain the term physical media
- identify the various type of physical media
- identify which physical media is good for each of the network types.

3.0 MAIN CONTENT

3.1 Introduction to Physical Media

In the *OSI* reference model, any physical means for transmitting data is referred to as the physical media. The bottom of the *OSI* model's physical layer provides an interface to such media. Specifications for the physical media themselves are not part of the *OSI* model.

3.2 Types of Physical Media

Let us look at these one after another.

a. **Twisted pair -** wire twisted to avoid crosstalk interference. It may be shielded or unshielded.

- UTP-Unshielded Twisted Pair. Normally *UTP* contains 8 wires or 4 pair; 100 meter maximum length; *4-100 Mbps* speed.
- STP-Shielded Twisted Pair. 100 meter maximum length; *16-155 Mbps speed*; lower electrical interference than *UTP*.

b. **Coaxial** - two conductors separated by insulation such as *TV 75 ohm cable*; maximum length of 185 to 500 meters; It is of two types, namely:

thinnet - *thinnet* uses a British Naval Connector (BNC) on each end. *Thinnet* is part of the *RG-58* family of cable. Maximum cable length is 185 meters. Transmission speed is 10Mbps. *Thinnet* cable should have *50 ohms* impedance; and its terminator has *50 ohms* impedance; barrel connector will have no impedance. Maximum *thinnet* nodes are *30* on a segment. One end of each cable is grounded.

thicknet - half inch rigid cable; maximum cable length is 500 meters. Transmission speed is *10Mbps*; it is expensive and is not commonly used- *(RG-11 or RG-8)*. A vampire tap or piercing tap is used with a transceiver attached to connect computers to the cable. 100 connections may be made. The computer has an Attachment Unit Interface (AUI) on its network card which is a *15 pin DB-15* connector. The computer is connected to the transceiver at the cable from its *AUI* on its network card using a drop cable. Maximum *thicknet* nodes are 100 on a segment. One end of each cable is grounded.

The *RG* value for cable types refers to its size. Coax cable types are listed below.

- RG-58 /U - 50 ohm, with a solid copper wire core for thin ethernet.
- RG-58 A/U - 50 ohm, with a stranded wire core.
- RG-58 C/U - Military version of RG-58 A/U.
- RG-59 - 75 ohm, for broadband transmission such as cable TV.
- RG-62 - 93 ohm, primarily used for ArcNet.
- RG-6 - used for satellite cable (if you want to run a cable to a satellite).
- RG-8 - 50 ohm thick ethernet.
- RG-11 - 75 ohm thick ethernet.

c. **Fiber-optic** - data is transmitted using light rather than electrons. Usually there are two fibers, one for each direction. Cable length of 2 Kilometers; speed from *100Mbps* to *2Gbps*. This is the most expensive and most difficult to install, but is not subject to interference. There are two types of cables:

- single mode cables for use with lasers have greater bandwidth and cost more. Injection laser diodes (ILD) work with single mode cable.
- multimode cables for use with Light Emitting Diode (LED) drivers; all signals appear to arrive at the same time. P intrinsic N diodes or photodiodes are used to convert light to electric signals when using multimode.

Table 3.1: Types of Fiber Cable

Fiber thickness (microns)	**Cladding thickness (microns)**	**Mode**
8.3	125	single
62.5	125	multi
50	125	multi
100	140	multi

SELF-ASSESSMENT EXERCISE 1

How is the fiber optic medium better than coaxial cable?

3.2 Physical Media Comparisons

Let us look at the features of different types of physical media in the table below.

Table 3.2: Comparing Physical Media

Media	**Distance(meters)**	**Speed**	**Approx. cost/station**
UTP	100	4-100Mbps	$90
STP	100	16-155Mbps	$125
Thinnet	185	10Mbps	$25
Thicknet	500	10Mbps	$50
Fiber	2000	100Mbps-2Gbps	$250 (multimode)

4.0 CONCLUSION

This unit has taken you through understanding the physical media and its various types. In this unit, we also compare the media- using the distance they can cover; their effective speed of transferring data as our basis of comparison. The next unit will introduce you to network protocols. Protocol can be viewed as a language that has to be understood by all the computers on a network.

5.0 SUMMARY

In this unit, you have learnt that:

- a physical medium is any physical means for transmitting data
- the 3 types of physical media are twisted pair, coaxial cable and fiber optic.

6.0 TUTOR-MARKED ASSIGNMENT

1. Discuss the various types of physical media as regard the distance they can cover.
2. State the features of 8 types of coaxial cable.
3. Explain the two types of fiber optic cable.

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UNIT 4 NETWORK PROTOCOLS

CONTENTS

1.0 INTRODUCTION

In this unit, you will be taken through Network Protocol and its general concepts.

2.0 OBJECTIVES

At the end of this unit, you should be able to:

- explain the meaning of network protocols
- state the types of network protocols.

3.0 MAIN CONTENT

3.1 Introduction to Network Protocols

A protocol is a set of rules that governs the communications between computers on a network. These rules include guidelines that regulate the following characteristics of a network: access method, allowed physical topologies, types of cabling, and speed of data transfer.

Packets can be transmitted across networks or over telephone lines. In fact, network protocols and several communications protocols use packet switching to establish a connection and route information. The format of a packet depends on the protocol that creates the packet the network computer must have a network protocol driver loaded. This program may be referred to as the transport protocol, or just as protocol. It operates between the adapter and the initial layer of network software to package and unpack data for the *LAN*.

3.2 Network Protocol Overview

The *OSI* model, and any other network communication model, provides only a conceptual framework for communication between computers, but the model itself does not provide specific methods of communication. Actual communication is defined by various communication protocols. In the context of data communication, a protocol is a formal set of rules, conventions and data structure that governs how computers and other network devices exchange information over a network. In other words, a protocol is a standard procedure and format that two data communication devices must understand, accept and use to be able to talk to each other.

In modern protocol design, protocols are "layered" according to the *OSI* 7 layer model or a similar layered model. Layering is a design principle which divides the protocol design into a number of smaller parts, each part accomplishing a particular sub-task and interacting with the other parts of the protocol only in a small number of well-defined ways. Layering allows the parts of a protocol to be designed and tested without a combinatorial explosion of cases, keeping each design relatively simple. Layering also permits familiar protocols to be adapted to unusual circumstances.

The header and/or trailer at each layer reflect the structure of the protocol. Detailed rules and procedures of a protocol or protocol group are often defined by a lengthy document. For example, *IETF* uses *RFCs* (Request For Comments) to define protocols and updates to the protocols.

A wide variety of communication protocols exists. These protocols were defined by many different standard organisations throughout the world and by technology vendors over years of technology evolution and development. One of the most popular protocol suites is *TCP/IP*, which is the heart of Internetworking communications. The IP- Internet Protocol is responsible for exchanging information between routers, so that the routers can select the proper path for network traffic; while *TCP* is responsible for ensuring that data packets are transmitted across the network, reliably, and error free. *LAN* and *WAN* protocols are also critical protocols in network communications. The *LAN* protocols suite is for the physical and data link layers of communications over various *LAN* media, such as ethernet wires and wireless radio waves. The *WAN* protocol suite is for the lowest three layers and defines communication over various wide-area media, such as fiber optic and copper cables.

Network communication has slowly evolved. Today's new technologies are based on the accumulation of technologies over the years- which may be either still existing or obsolete. As a result of this, the protocols which define network communication are highly inter-related. Many protocols rely on others for operation. For example, many routing protocols use other network protocols to exchange information between routers.

In addition to standards for individual protocols in transmission, there are now also interface standards for different layers to talk to the ones above or below (usually operating system specific). For example, Winsock and Berkeley socket between layers 4 and 5; *NDIS* and *ODI* between layers 2 and 3.

The protocols for data communication cover all areas as defined in the *OSI* model. However, the *OSI* model is only loosely defined. A protocol may perform the functions of one or more of the *OSI* layers, which introduces complexity to understanding protocols relevant to the *OSI* 7 layer model. In real-world protocols, there is some argument as to where the distinctions between layers are drawn; there is no one black and white answer.

To develop a complete technology that is useful for the industry, very often a group of protocols is required in the same layer or across many different layers. Different protocols often describe different aspects of a single communication; taken together, these form a protocol suite. For example, Voice over IP (VOIP), a group of protocols developed by many vendors and standard organisations, has many protocols across the 4 top layers in the *OSI* model.

Protocols can be implemented either in hardware or software or a mixture of both. Typically, the lower layers are implemented in hardware, with the higher layers being implemented in software.

Protocols could be grouped into suites (or families, or stacks) by their technical functions, or origin of the protocol introduction, or both. A protocol may belong to one or multiple protocol suites, depending on how you categorise it. For example, the Gigabit ethernet protocol *IEEE 802.3z* is a LAN (Local Area Network) protocol and it can also be used in MAN (Metropolitan Area Network) communications.

Most recent protocols are designed by the *IETF* for internetworking communications and by the *IEEE* for local area networking and metropolitan area networking. The *ITU-T* contributes mostly to Wide Area Networking (WAN) and telecommunications protocols. *ISO* has its own suite of protocols for internetworking communications, which is mainly deployed in European countries.

3.4 Types of Network Protocols

The most common network protocols are as listed below.

- Ethernet
- Local talk
- Token ring
- FDDI
- ATM

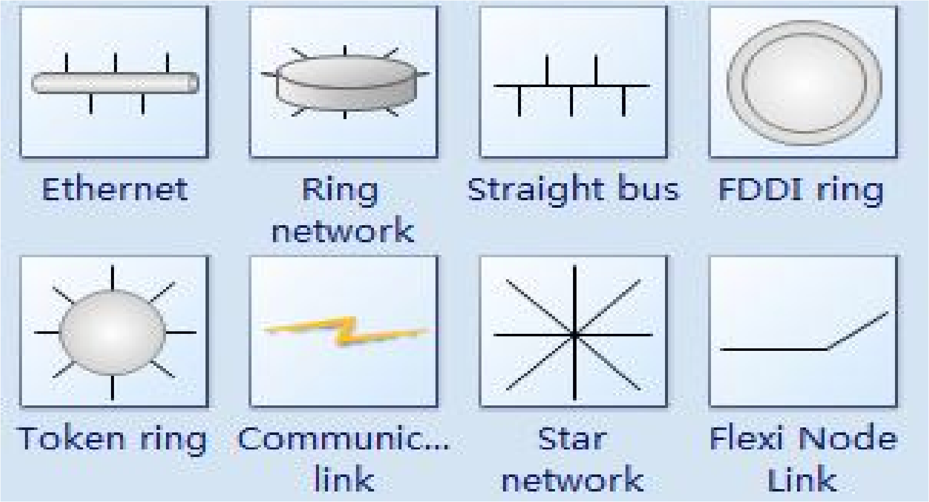

Figure 4. 1: Commonly Used Network Symbols for Different Kinds of Network Protocols

Ethernet

The ethernet protocol is, by far, the most widely used. Ethernet uses an access method called CSMA/CD (Carrier Sense Multiple Access/Collision Detection). This is a system where each computer listens to the cable before sending anything through the network. If the network is clear, the computer will transmit. If some other node is already transmitting on the cable, the computer will wait and try again when the line is clear. Sometimes, two computers attempt to transmit at the same instant. When this happens a collision occurs; each computer then backs off and waits a random amount of time before attempting to retransmit. With this access method, it is normal to have collisions. However, the delay caused by collisions and retransmitting is very small and does not normally affect the speed of transmission on the network.
The ethernet protocol allows for linear bus, star, or tree topologies. Data can be transmitted over wireless access points, twisted pair, coaxial, or fiber optic cable at a speed of 10 Mbps up to 1000 Mbps.

Fast ethernet

To allow for an increased speed of transmission, the ethernet protocol has developed a new standard that supports *100 Mbps*. This is commonly called fast ethernet. Fast ethernet requires the use of different, more expensive network concentrators/hubs and network interface cards. In addition, category 5 twisted pair or fiber optic cable is necessary. Fast ethernet is becoming common in schools that have been recently wired.

Local talk

Local talk is a network protocol that was developed by Apple Computer, Inc. for Macintosh computers. The method used by local talk is called CSMA/CA (Carrier Sense Multiple Access with Collision Avoidance). It is similar to *CSMA/CD* except that a computer signals its intent to transmit before it actually does so. Local talk adapters and special twisted pair cable can be used to connect a series of computers through the serial port. The Macintosh operating system allows the establishment of a peer-to-peer network without the need for additional software. With the addition of the server version of AppleShare software, a client/server network can be established.

The local talk protocol allows for linear bus, star, or tree topologies using twisted pair cable. A primary disadvantage of local talk is speed. Its speed of transmission is only 230 Kbps.

Token ring

The token ring protocol was developed by *IBM* in the mid-1980s. The access method used involves token-passing. In token ring, the computers are connected so that the signal travels around the network from one computer to another in a logical ring. A single electronic token moves around the ring from one computer to the next. If a computer does not have information to transmit, it simply passes the token on to the next workstation. If a computer wishes to transmit and receives an empty token, it attaches data to the token. The token then proceeds around the ring until it comes to the computer for which the data is meant. At this point, the data is captured by the receiving computer. The token ring protocol requires a star-wired ring using twisted pair or fiber optic cable. It can operate at transmission speeds of *4 Mbps* or *16 Mbps*. Due to the increasing popularity of ethernet, the use of token ring in school environments has decreased.

FDDI

Fiber Distributed Data Interface (FDDI) is a network protocol that is used primarily to interconnect two or more local area networks, often over large distances. The access method used by *FDDI* involves token-passing. *FDDI* uses a dual ring physical topology. Transmission normally occurs on one of the rings; however, if a break occurs, the system keeps information moving by automatically using portions of the second ring to create a new complete ring. A major advantage of *FDDI* is speed. It operates over fiber optic cable at *100 Mbps*.

ATM

Asynchronous Transfer Mode (ATM) is a network protocol that transmits data at a speed of *155 Mbps* and higher. *ATM* works by transmitting all data in small packets of a fixed size; whereas, other protocols transfer variable length packets. *ATM* supports a variety of media such as video, CD-quality audio, and imaging. *ATM* employs a star topology, which can work with fiber optic as well as twisted pair cable.

ATM is most often used to interconnect two or more local area networks. It is also frequently used by Internet Service Providers (ISP) to utilise high-speed access to the internet for their clients. As *ATM* technology becomes more cost-effective, it will provide another solution for constructing faster local area networks.

Gigabit ethernet

The most recent development in the ethernet standard is a protocol that has a transmission speed of *1 Gbps*. Gigabit ethernet is primarily used for backbones on a network at this time. In the future, it will probably be used for workstation and server connections also. It can be used with both fiber optic cabling and copper. The *1000BaseTX*, the copper cable used for Gigabit ethernet, became the formal standard in 1999.

Table 4. 1: Comparison of Network Protocols

Protocol	**Cable**	**Speed**	**Topology**
Ethernet	Twisted Pair, Coaxial, Fiber	10 Mbps	Linear Bus, Star, Tree
Fast Ethernet	Twisted Pair, Fiber	100 Mbps	Star
LocalTalk	Twisted Pair	.23 Mbps	Linear Bus or Star

Token Ring	Twisted Pair	4 Mbps - 16 Mbps	Star-Wired Ring
FDDI	Fiber	100 Mbps	Dual ring
ATM	Twisted Pair, Fiber	155-2488 Mbps	Linear Bus, Star, Tree

4.0 CONCLUSION

This unit has introduced basic rudiments of network protocols, under which you have learnt different types of network protocols which include *ATM*, *FDDI* etc.

5.0 SUMMARY

In this unit, you have learnt about:

- network protocol, which is a set of rules that governs communication between computers on a network
- types of network protocols, namely- ethernet, local talk, token ring, *FDDI* and *ATM*
- Comparison between different network protocols.

6.0 TUTOR-MARKED ASSIGNMENT

1. What is the significance of network protocol?
2. Differentiate between 5 types of network protocols.
3. State the main features of *FDDI* and *ATM*

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MODULE 2 NETWORK DESIGN

UNIT 1 HARNESSING WI-FI FOR USER MOBILITY

CONTENTS

1.0 INTRODUCTION

This unit will expose you to the concept of wi-fi technology and its features.

2.0 OBJECTIVES

At the end of this unit, you should be able to:

- define the concept of wi-fi technology
- explain the details of what brought about wi-fi technology.

3.0 MAIN CONTENT

3.1 Overview of Wi-Fi Technology

Testing techniques developed for wired devices and networks fall short when applied to the *WLAN* market. The inherent instability of the unwired medium — air — in which the wireless world operates and the constant mobility of the *WLAN* user make the *802.11* protocol an order of magnitude more complex than equivalent wired protocols. As a result, the metrics used to benchmark wired protocols are only a starting point for the *WLAN* industry.

Differences between wired and wireless networks require metrics and methodology for performance benchmarking that address the intricacies of the *802.11* protocol. This unit addresses wireless-specific functions central to business-critical applications on a *wi-fi* infrastructure and includes typical results and interpretations.

3.2 Introduction to Wi-Fi Technology

The *IEEE 802.11b* standard is popularly known as “Wireless Fidelity” (or Wi-Fi- for short). It has become widely popular for wireless *LANs* in office environments. Proponents of this technology consider it great competition to third generation wireless networks, which also provide high data rate, mobile internet access. Wi-fi can be used to provide broadband wireless internet access.

Access Points (APs) can be installed at various locations in the city. The *APs* are also called “hotspot”. All the *APs* in a city can be interconnected through an *ATM*-based backbone network. As the wireless device moves from one location to another, the mobile device is connected to the nearest *AP*.

3.3 *WLAN* Performance Metrics

Wi-fi protocols address differences between wired and wireless networks, and the implementation of the more advanced wireless protocol demands performance validation. Algorithms used in a network's clients and *APs* (and the capacity of these devices to process the algorithms) limit the network's performance. The objective of the validation process and test metrics is to identify critical test parameters and find the correct method of testing them.

Testing Ethernet network performance is essentially a measure of packet forwarding rate. In addition to packet forwarding measurements, *WLAN*s must undergo tests related to the unstable physical layer and end-user mobility; including automatic data rate adaptation, roaming, verification of security, *QOS* and overlapping *BSSs*, as well as behavioral tests that measure performance under abnormal network conditions. The primary focus of the testing effort should be parameters that eventually affect network efficiency and operation.

Data rate adaptation- wired *LANs* support fixed data rates: *10/100/1000 Mb/s*. Wireless networks support multiple data rates: *11/5.5/2/1 Mb/s for 802.11b; 54/48/36/24/18/12/9/6 Mb/s for 802.11a and 802.11g*. The critical difference is that *WLANs* support dynamic rate adaptation and can operate at multiple data rates automatically determined by the end point (AP or client).

In addition, because the *802.11* standard does not specify exact criteria for data rate adaptation, the algorithms can vary from device to device. The rate adaptation algorithm should be based on optimising throughput; that is, when the number of errors at a specific data rate increases to the point where throughput is severely affected, the device should drop to a lower data rate to recover the best throughput at that distance from the *AP*.

The challenge is how to measure this repeatedly, while creating a metric that can be set as the *golden standard*. To measure throughput at fixed points, many vendors often use an interference-free environment with a long, direct line-of-site area where they can simulate data rate adaptation by wheeling a client up and down on a cart. This method, however, lacks accurate rate adaptation data and is less efficient than newer devices that offer controlled *RF* environments and accurate signal attenuation through test setup automation. While characterising *range vs. data* rate, the test should simultaneously characterise *range vs. throughput* and *range vs. packet* error rate.

Roaming- as a client moves out of range of one *AP*, it dissociates from the *AP* and must associate and authenticate with another. If the client predicts this roam will occur by noticing the drop in signal and searching for an alternate *AP* before it is actually disassociated from the first, it can optimise the roam time and network disruption caused by the roam.

The client device makes the decision to roam based on its position relative to different *APs* and their signal strengths. The client might periodically analyse signal strength of the *APs* that surround it and decide which one to associate with if it needs to roam. Load-sharing protocols used by some *WLAN* network vendors depart from the traditional client-based decision process, orchestrating client devices to associate with specific *APs* and spreading the load evenly among APs and optimising the entire network throughput. In addition, the *IEEE* is advancing its work on roaming through better *RF* measurement (802.11k) and fast roaming processes (802.11r). As the roaming process increases in complexity, it is critical to have standard roaming metrics for testing *WLAN* networks and equipment.

Roaming is critical because it takes time, which can cause data loss that can ultimately disrupt a communication session. Data loss is particularly important for time-sensitive enterprise applications, such as *VoWLAN*, that are especially susceptible to packet delay caused by roaming. Roaming metrics include *roaming time, packet loss and session continuity*. Roaming time can be broken down into the following stages- scanning, associating to the new *AP*, authentication with the new *AP* and data flow. Analysing the time of each phase of this process will help ensure the most efficient roam time.

Packet forwarding- forwarding rate is a function of a device: in wired networks, the ethernet switch; in *WLANs*, the *AP*. Packet forwarding rate testing is always done at the highest signal strength and at the highest data rate because this puts the most demand on the device and measures its packet processing power in the most extreme case.

Like wired throughput tests, a wireless packet forwarding test varies the packet size to ensure the ability of the device to work with diverse traffic; but unlike wired devices, there are other factors to consider. The most critical is *security*, because wireless network devices must encrypt each packet. This additional overhead must be added to evaluate its effect on the packet forwarding rate. Another important factor is client capacity. Running the test with a large number of users stresses the *AP's* ability to handle a large number of users, each sending a portion of the bandwidth. This also affects how the *AP* functions under such conditions.

Security- this is a critical consideration for enterprise networks; because they are susceptible to intruders, wireless networks have more stringent security requirements than their wired counterparts. Wireless security protocols (802.11i) rely heavily on authentication and encryption, which depend on the processing power of the *AP* and client, and cryptography accelerators for data encryption. The efficiency with which the devices handle key management and encryption will have an effect on performance measurements, such as forwarding rate and roaming.

When a client initially accesses the network or roams between *APs*, authentication occurs using protocols such as *EAP-TLS, EAP-TTLS* and *LEAP*. Complex key derivation algorithms can overload *APs* if multiple simultaneous authentication requests are made. Authentication of wireless networks is tested by measuring how efficiently and quickly an *AP* manages simultaneous authentication requests.

Encryption protocols used in wi-fi, such as *WEP*, *TKIP* and *AES/CCMP*, can also impact throughput performance. The security metric is performed by making a series of comparative throughput measurements using different encryption methods.

***Quality of Service (QOS)* -** because *802.11* is a shared media protocol without *QOS*, *WLANs* cannot prioritise real-time applications such as voice and video, over data applications. QOS protocols for *WLANs* must account for jitter, delay and packet loss, which have required minimums for real-time applications including *VOIP* and multimedia streaming. Jitter, or inter-packet delay, is particularly critical in packetised voice.

3.4 Testing Methodology for Wireless Networks

Traditionally, wireless system designers have had a variety of testing options. Most are home-grown or custom-built, and include isolated screen rooms for *RF* control, large open spaces for testing mobility and expensive off-the-shelf meters that focus on point-to-point tests of the physical layer. Another approach is emerging for integrated chassis-based wi-fi testing. The older methods are costly, and in many cases do not provide the systems level configuration needed to accurately and usefully provide relevant information about the metrics discussed above. *Isolated screen rooms for controlling RF interference* — *WLAN* system developers reduce the effects of *RF* interference by conducting tests in a large screen room that isolates devices under test from extraneous *RF* interference. It is the wireless equivalent of a "clean room." Although screen rooms can eliminate interference effects, they cannot test real-world network conditions such as mobility and roaming. Screen rooms can be expensive to erect and maintain; and are not portable, which limits their use and effectiveness.

SELF-ASSESSMENT EXERCISE

Describe the performance of Wi-fi technology.

3.5 Test Setup and Sample Results

The roaming test is an example that shows the efficiency that can be achieved using a chassis-based test platform. Using two test modules — a *WLA* and an *RFM* — a phone or a *PC* client is connected between the two *APs* through two *80 dB* programmable attenuators. The attenuators are programmed to force the client to roam in a controlled way from one *AP* to another. Data collection is performed on the source and destination channel simultaneously by the integrated dual-channel *WLA* module. The roaming test is fully automated and can be configured to repeat the measurements for a set period.

One attenuator is initially set to minimum and the other to maximum so that the client receives a strong signal from *AP1* and associates with it; while *AP2* is out of the client's range. The attenuator between *AP1* and the client is then gradually increased, eventually making *AP1* invisible to the client while the attenuator between the client and *AP2* is gradually decreased, "moving" *AP2* within range and forcing a roam. The ranges and the rate of change of attenuators are configurable within the test script. At the end of the roaming test, the test script tabulates the details of each roam.

This automated roaming test implementation provides accurate time measurements and identifies specific time intervals in a way that emulates real-life roaming to the clients and access points: gradual signal strength decrease and increase. The entire test can be repeated as many times as is needed, and multiple roams can be performed in a short period without human involvement.

4.0 CONCLUSION

This unit introduces you to wi-fi technology and also teaches the advantage of the technology over wired technology. This unit has also exposed you to the testing methodology for wireless networks**.**

5.0 SUMMARY

In this unit, you have learnt that:

- The *IEEE 802.11b* standard is popularly known as ‘Wireless Fidelity’ (or Wi-Fi).
- Wi-fi protocols address the differences between wired and wireless networks
- Wi-fi can be used to provide broadband wireless internet access.
- *WLAN* must be tested continually; the primary focus of the testing effort should be parameters that eventually affect network efficiency and operation.
- Wi-fi technology has become popular for wireless *LANs* in office environment

6.0 TUTOR-MARKED ASSIGNMENT

1. What do you understand by the term “wi-fi?”
2. In a tabular form, compare and contrast wi-fi and the *WLAN.*
3. State and explain the metrics for testing *WLAN*

7.0 REFERENCES/ FURTHER READING

"NASA Successfully Tests First Deep Space Internet". NASA media release 08-298, November 18, 2008 Archived.

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http://www.azimuthsystems.com/

UNIT 2 BUILDING INTERNETWORKS USING TRANSPORT CONTROL PROTOCOL (TCP)/INTERNET PROTOCOL (IP) AND ROUTERS

CONTENTS

1.0 INTRODUCTION

The language of the internet is Transport Control Protocol/Internet Protocol (TCP/IP). No matter what type of computer platform or software is being used, the information must move across the internet in this format. This protocol calls for data to be grouped together, in bundles, called network packets.

2.0 OBJECTIVES

At the end of this unit, you should be able to:

- explain the fundamentals of the *TCP/IP*
- state the relevance of *TCP/IP* in internetworks.

3.0 MAIN CONTENT

3.1 TCP Technology

TCP is a connection−oriented transport protocol that sends data as an unstructured stream of bytes. By using sequence numbers and acknowledgment messages, *TCP* can provide a sending node with delivery information about packets transmitted to a destination node. Where data has been lost in transit from source to destination, *TCP* can retransmit the data until either a timeout condition is reached or until successful delivery has been achieved. *TCP* can also recognise duplicate messages and will discard them appropriately. If the sending computer is transmitting too fast for the receiving computer, *TCP* can employ flow control mechanisms to slow data transfer. *TCP* can also communicate delivery information to the upper–layer protocols and applications it supports. All these characteristics make *TCP* an end–to–end reliable transport protocol. TCP is specified in *RFC 793*.

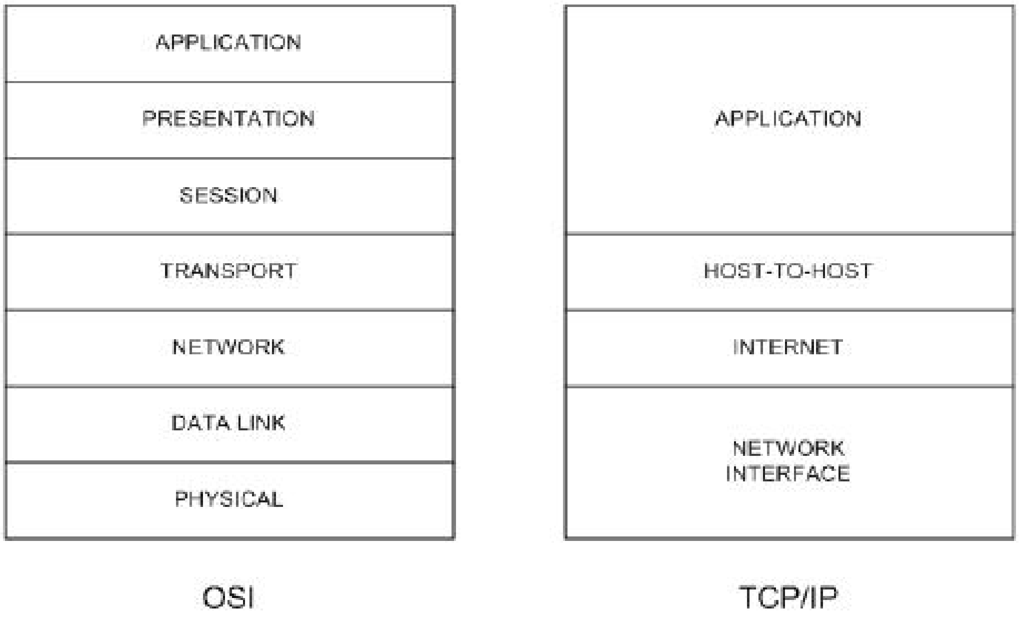

Figure2. 1: TCP/IP Protocol Suite in Relation to the OSI Reference Model

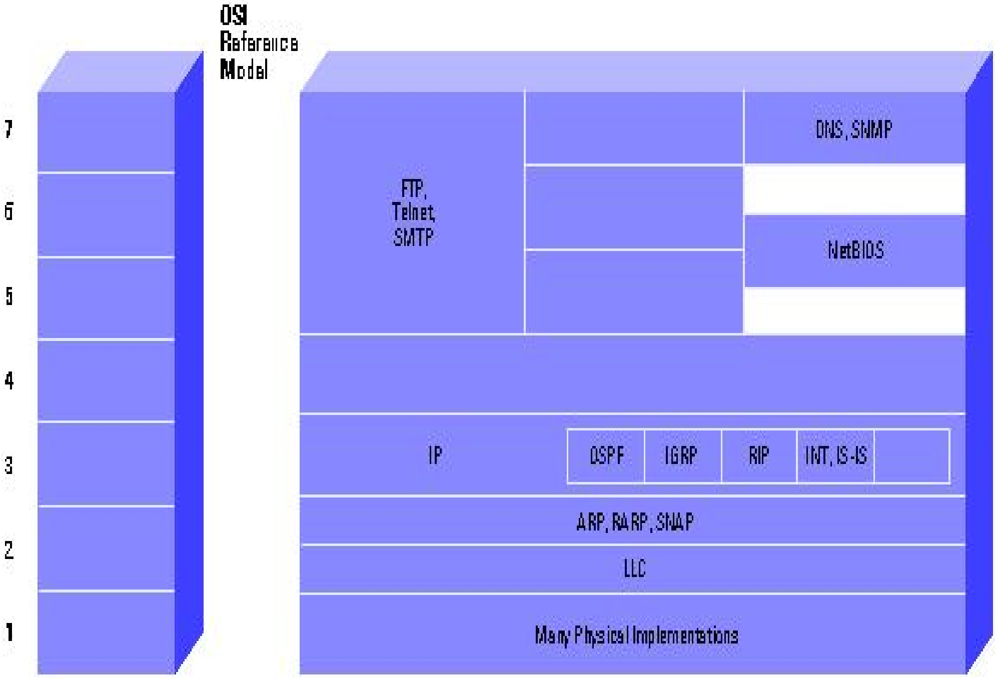

Figure 2.2: Important Internet Protocols in Relation to the OSI Reference Model

3.1.1 Internet Protocol (IP)

IP is the primary layer 3 protocol in the internet suite. In addition to internetwork routing, *IP* provides error reporting and fragmentation and re-assembly of information units called datagrams for transmission over networks with different maximum data unit sizes. *IP* represents the heart of the internet protocol suite**(note:** the term *IP* in the section refers to *IPv4* unless otherwise stated explicitly).

IP addresses are globally unique, 32−bit numbers assigned by the network information center. Globally unique addresses permit *IP* networks anywhere in the world to communicate with each other. An *IP* address is divided into two parts. The first part designates the network address while the second part designates the host address. The *IP* address space is divided into different network classes. Class *A* networks are intended mainly for use with a few very large networks, because they provide only 8 bits for the network address field. Class *B* networks allocate 16 bits, and Class *C* networks allocate 24 bits for the network address field. Class *C* networks only provide 8 bits for the host field however, so the number of hosts per network may be a limiting factor. In all three cases, the left most bit(s) indicate the network class. *IP* addresses are written in dotted decimal format; for example, *34.0.0.1*. Figure 2.3 below shows the address formats for Class *A, B*, and *C IP* networks.

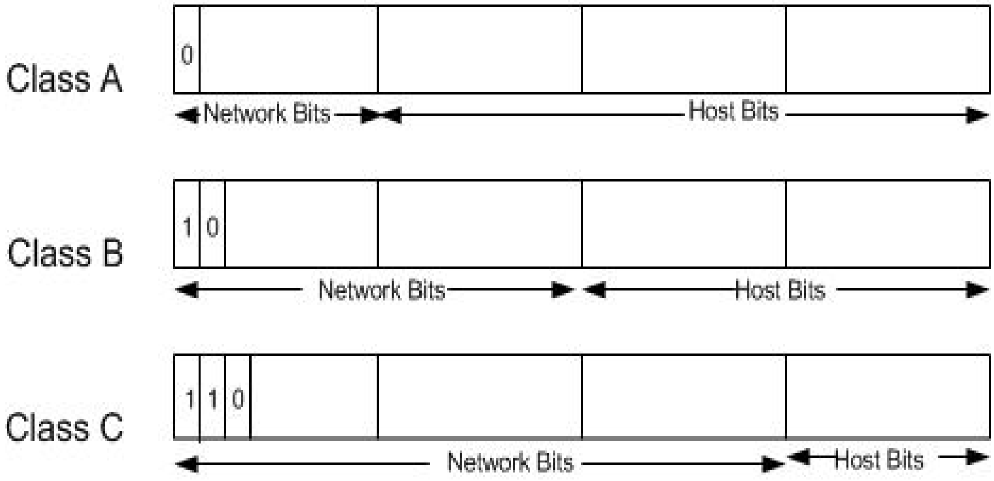

Figure2. 3: Address Formats for Class A, B, and C IP Networks

IP networks also can be divided into smaller units called subnetworks or ‘subnets’. Subnets provide extra flexibility for the network administrator. For example, assume that a network has been assigned a Class *A* address and all the nodes on the network use a Class *A* address. Further assume that the dotted decimal representation of this network's address is 34.0.0.0. (All zeros in the host field of an address specify the entire network). The administrator can subdivide the network using subnetting. This is done by "borrowing" bits from the host portion of the address and using them as a subnet field, as depicted in Figure 2. 4 below.

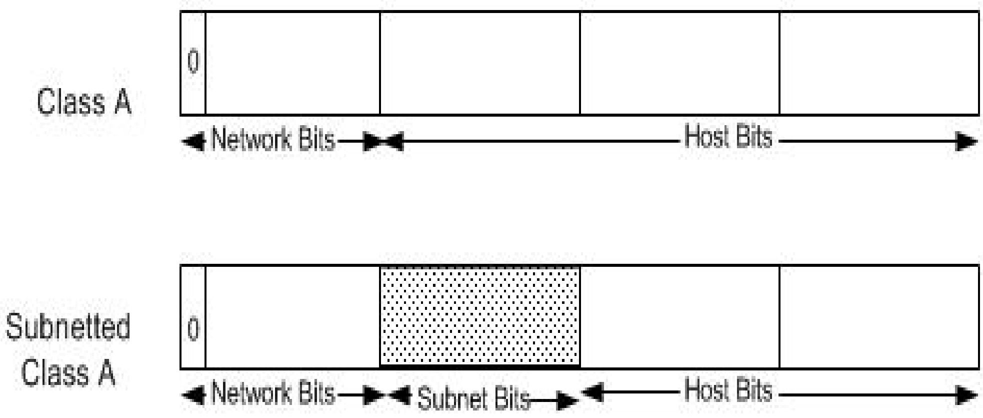

Figure2.4: "Borrowing" Bits

If the network administrator has chosen to use 8 bits of subnetting, the second octet of a Class *A IP* address provides the subnet number. In our example, address *34.1.0.0* refers to network 34, subnet 1; address *34.2.0.0* refers to network 34, subnet 2, and so on.

The number of bits that can be borrowed for the subnet address varies. To specify how many bits are used to represent the network and the subnet portion of the address, *IP* provides subnet masks. Subnet masks use the same format and representation technique as *IP* addresses. Subnet masks have ones in all bits except those that specify the host field. For example, the subnet mask that specifies 8 bits of subnetting for Class *A* address *34.0.0.0* is *255.255.0.0*. The subnet mask that specifies 16 bits of subnetting for Class *A* address *34.0.0.0* is *255.255.255.0*. Both of these subnet masks are pictured in Figure 2.5 below. Subnet masks can be passed through a network on demand so that new nodes can learn how many bits of subnetting are being used on their network.

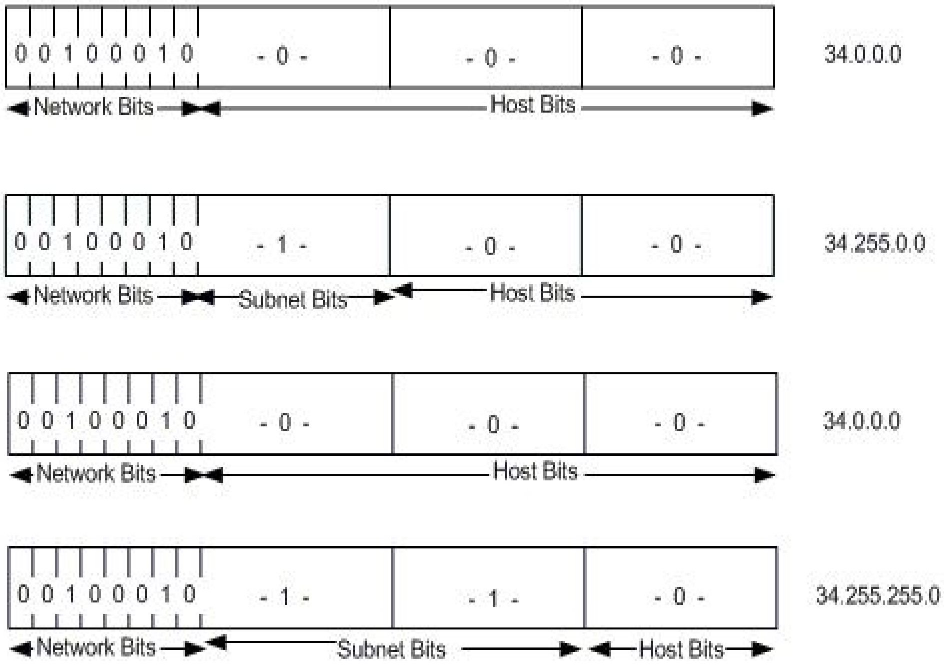

Figure 2.5: Subnet Masks

Traditionally, all subnets of the same network number used the same subnet mask. In other words, a network manager would choose an eight−bit mask for all subnets in the network. This strategy is easy to manage for both network administrators and routing protocols. However, this practice wastes address space in some networks. Some subnets have many hosts and some have only a few, but each consumes an entire subnet number. Serial lines are the most extreme example, because each has only two hosts that can be connected via a serial line subnet. As *IP* subnets have grown, administrators have looked for ways to use their address space more efficiently.

One of the techniques of subnetting is called Variable Length Subnet Masks (VLSM). With *VLSM*, a network administrator can use a long mask on networks with few hosts and a short mask on subnets with many hosts. However, this technique is more complex than making them all one size; and addresses must be assigned carefully.

Of course in order to use *VLSM*, a network administrator must use a routing protocol that supports it. Cisco routers support *VLSM* with Open Shortest Path First (OSPF), Integrated Intermediate System to Intermediate System (Integrated IS−IS), Enhanced Interior Gateway Routing Protocol (Enhanced IGRP), and static routing. On some media, such as *IEEE 802 LANs*, *IP* addresses are dynamically discovered through the use of two other members of the internet protocol suite: Address Resolution Protocol (ARP) and Reverse Address Resolution Protocol (RARP). *ARP* uses broadcast messages to determine the hardware (MAC layer) address corresponding to a particular network−layer address. *ARP* is sufficiently generic to allow use of *IP* with virtually any type of underlying media access mechanism. *RARP* uses broadcast messages to determine the network−layer address associated with a particular hardware address. *RARP* is especially important to diskless nodes, for which network−layer addresses usually are unknown at boot time.

3.2 Routing in *IP* Environments

An internet is a group of interconnected networks. The internet, on the other hand, is the collection of networks that permits communication between most research institutions, universities, and many other organisations around the world. Routers within the internet are organised hierarchically. Some routers are used to move information through one particular group of networks under the same administrative authority and control. (Such an entity is called an autonomous system.) Routers used for information exchange within autonomous systems are called interior routers, and they use a variety of Interior Gateway Protocols (IGPs) to accomplish this end. Routers that move information between autonomous systems are called exterior routers; they use the Exterior Gateway Protocol (EGP) or Border Gateway Protocol (BGP).

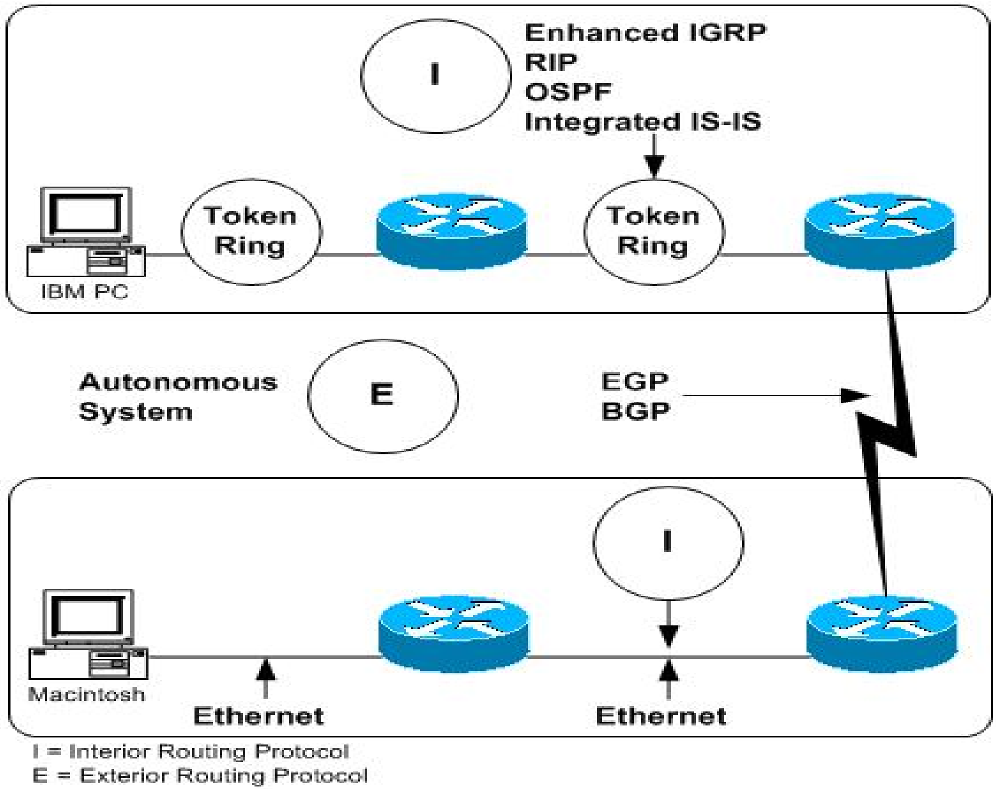

Figure 2.6: Representation of the Internet Architecture

Routing protocols used with *IP* are dynamic in nature. Dynamic routing requires the software in the routing devices to calculate routes. Dynamic routing algorithms adapt to changes in the network and automatically select the best routes. In contrast with dynamic routing, static routing calls for routes to be established by the network administrator. Static routes do not change until the network administrator changes them. *IP* routing tables consist of destination address/next hop pairs.

4.0 CONCLUSION

This unit has taken you through the *TCP/IP* networks, its model and implementation procedure.

5.0 SUMMARY

In this unit, you have learnt that:

- *TCP* is a connection−oriented transport protocol that sends data as an unstructured stream of bytes
- *IP* is the primary layer 3 protocol in the internet protocol suite and *IP* represents the heart of the protocol
- one of the techniques of subnetting is called Variable Length Subnet Masks (VLSM); with VLSM, a network administrator can use a long mask on networks with few hosts and a short mask on subnets with many hosts
- *IP* networks also can be divided into smaller units called subnetworks or subnets. Subnets provide extra flexibility for the network administrator
- routing protocols used with *IP* are dynamic in nature.

6.0 TUTOR-MARKED ASSIGNMENT

1. Explain the *TCP/IP* model of networking.
2. With the aid of diagram, explain internet architecture.
3. What are the differences between the Class *A*, Class *B* and Class *C IP* networks?

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UNIT 3 NETWORK STANDARDS (*IEEE* 802 STANDARDS)

CONTENTS

1.0 INTRODUCTION

IEEE 802 refers to a family of *IEEE* standards dealing with local area networks and metropolitan area networks.

2.0 OBJECTIVES

At the end of this unit, you should be able to:

- state what the *IEEE* 802 standard stands for
- explain the various *IEEE* 802 standards and their descriptions.

3.0 MAIN CONTENT

3.1 Overview of *IEEE* 802

Specifically, the *IEEE* 802 standards are restricted to networks carrying variable-size packets. (By contrast, in cell relay networks, data are transmitted in short, uniformly sized units called cells. Isochronous networks, where data is transmitted as a steady stream of octets, or groups of octets, at regular time intervals, are also out of the scope of this standard). The number 802 was simply the next free number *IEEE* could assign, though “802” is sometimes associated with the date the first meeting was held — February 1980.

The services and protocols specified in *IEEE* 802 map to the lower two layers (data link and physical) of the seven-layer *OSI* networking reference model. In fact, *IEEE* 802 splits the *OSI* data link layer into two sub-layers named Logical Link Control (LLC) and Media Access Control (MAC), so that the layers can be listed like this:

- data link layer
- physical layer

The *IEEE* 802 family of standards is maintained by the *IEEE* 802 LAN/MAN Standards Committee (LMSC). The most widely used standards are for the ethernet family, token ring; wireless *LAN*, bridging and virtual bridged *LANs*. An individual working group provides the focus for each area. Table 3.1 below presents different *IEEE* network standards and their descriptions.

Table 3.1: *IEEE* Network Standards and their Descriptions

Name	**Description**	**Note**
IEEE 802.1	Bridging (networking) and Network Management	
IEEE 802.2	LLC	Inactive
IEEE 802.3	Ethernet	
IEEE 802.4	Token bus	Disbanded
IEEE 802.5	Defines the MAC layer for a Token Ring	Inactive
IEEE 802.6	MANs	Disbanded
IEEE 802.7	Broadband LAN using Coaxial Cable	Disbanded
IEEE 802.8	Fiber Optic TAG	Disbanded
IEEE 802.9	Integrated Services LAN	Disbanded
IEEE 802.10	Interoperable LAN Security	Disbanded
IEEE 802.11 a/b/g/n	Wireless LAN (WLAN) & Mesh (Wi-Fi certification)	
IEEE 802.12	100BaseVG	Disbanded
IEEE 802.13	Unused	
IEEE 802.14	Cable modems	Disbanded
IEEE 802.15	Wireless PAN	
IEEE 802.15.1	Bluetooth certification	
IEEE 802.15.2	IEEE 802.15 and IEEE 802.11 coexistence	
IEEE 802.15.3	High-Rate wireless PAN	
IEEE 802.15.4	Low-Rate wireless PAN (e.g. ZigBee)	
IEEE 802.15.5	Mesh networking for WPAN	
IEEE 802.16	Broadband Wireless Access (WiMAX certification)	

IEEE 802.16.1	Local Multipoint Distribution Service	
IEEE 802.17	Resilient packet ring	
IEEE 802.18	Radio Regulatory TAG	
IEEE 802.19	Coexistence TAG	
IEEE 802.20	Mobile Broadband Wireless Access	
IEEE 802.21	Media Independent Handoff	
IEEE 802.22	Wireless Regional Area Network	
IEEE 802.23	Emergency Services Working Group	New(March, 2010

4.0 CONCLUSION

In this unit, you learnt that the *IEEE 802* standards are restricted to networks carrying variable-size packets. You also learnt about the services and protocols specified in *IEEE* 802.

5.0 SUMMARY

In this unit, you have learnt that:

- the services and protocols specified in *IEEE* 802 map to the lower two layers (data link and physical) of the seven-layer *OSI* networking reference model
- the *IEEE* 802 family of standards is maintained by the *IEEE* 802 LAN/MAN Standards Committee (LMSC).
- *IEEE* 802 standards are widely used for the ethernet family, token ring; wireless *LAN*, bridging and virtual bridged *LANs*.
- there are variations of *IEEE* 802 standards.

6.0 TUTOR-MARKED ASSIGNMENT

1. Describe the importance of *IEEE*802 standards.
2. State all the working group of the *IEEE* 802 standard and their description.

7.0 REFERENCES/FURTHER READING

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UNIT 4 IMPLEMENTING SECURITY BEST PRACTICES

CONTENTS

1.0 INTRODUCTION

Network security involves maintaining Confidentiality, Integrity and Availability (CIA) of network. Confidentiality involves preventing unauthorised disclosure of resources or information on the network. Integrity means prevention of unauthorised modification of resources or information on the network. Integrity involves prevention of unauthorised denial of resources or information on the computer system.

2.0 OBJECTIVES

At the end of this unit, you should be able to:

- explain what network security is all about
- state the best network security
- highlight the features of best security practices.

3.0 MAIN CONTENT

3.1 Overview of Best Network Security

Efficient network security is needed for corporations, universities, schools, public libraries, internet cafes and other applications where administrator has to secure and maintain a lot of network workstations located in different places. Administrator does not need to physically visit workstations to change security settings or install patches. An efficient network security programme is needed for securing, protecting, and maintaining *PC* workstations within a corporate network. There exist network-based password-protected security software that allows you to completely secure workstations over your network as well as maintain them by uploading and installing any executable patches remotely. They support tons of security restrictions, options and tweaks to control access to every bit of windows.

You can deny access to each individual component of several control panel applets, including display, network, passwords, printers, system and internet options; disable the boot keys, context menus, Disk Operating System (DOS) windows, registry editing, internet and network access. You can also hide desktop icons, individual drives, start menu items, and taskbar; apply password protection to windows and restrict users to running specific applications only, control internet usage and much more. In total, best network security supports over 600 different security restrictions, options and tweaks that allow you to restrict access to almost every corner of windows.

After installing the remote client service application on your workstations, the maintenance becomes absolutely hassle-free. You just connect your administrator's computer to the net from any place and remotely change security settings, upload and execute patches as well as schedule reboots, shutdowns, and windows explorer restarts just with a click of the mouse. The best solution for corporations, universities, schools, public libraries, internet cafes etc., is to upload and install any executable patches remotely, apply security restrictions, options and tweaks. It supports over 600 different security restrictions.

3.2 Strategies for Best Network Security

The ten (10) key strategies for ensuring efficient network security are as listed below.

i. Create security training and awareness programme within the organisation. Let the personnel in your organisation be aware of:

- emerging threats
- possible consequences of a single security breach.
- security requirement

There must be regular security training. Collaborations with other cyber security organisations are also necessary.

ii. Have an efficient access control system

In order to control insider threats, there must be clear definition of roles and responsibilities in organisations. Efficient authorisation system should be implemented in order to limit the activities of personnel.

iii. Implement trusted and efficient intrusion prevention and detection systems

To effectively present cyber attacks, a trusted and adaptive intrusion prevention and detection system must be implemented. Nowadays, it is also necessary to implement intrusion forecasting system.

iv. Identify and document all resources on the network

All categories of resources on the network must be well documented and their status monitored and reported.

v. All connections to the network must be known, documented and monitored

Identify and monitor all connections (cable, wireless, local or internet) and ensure that these connections are well protected. All unwanted or unnecessary connections should be disconnected.

vi. Have an effective programme for the risk analysis and incident building

A periodic and thorough analysis of potential risks to network resources and vulnerability assessment is an essential requirement for effective cyber security management.

vii. Prevent unknown applications

Implement strategies that disallow any unknown application to run on your system.

viii. Use security applications from reliable vendors

Trusted security tools from known vendors should be used. The vendors must implement security features in form of patches and upgrades.

ix. Establish effective configuration management processes

Since changes in hardware and software configurations can introduce vulnerabilities, an assessment of the security implications of all changes should be carried out.

x. Ensure periodic evaluation of all security apparatus

Organisations should ensure periodic testing and evaluation of all security apparatus.

SELF-ASSESSMENT EXERCISE

What do you understand by cyber security?

4.0 CONCLUSION

This unit discusses the security issues in a network; and you have also learnt about the best security practices.

5.0 SUMMARY

In this unit, you have learnt that:

- network security involves maintaining confidentiality, integrity and availability (CIA) of network
- efficient network security is needed for corporations, universities, schools, public libraries, internet cafes and other applications where administrator has to secure and maintain a lot of network workstations located in different places.
- there are certain key strategies required for ensuring best network security practices

6.0 TUTOR-MARKED ASSIGNMENT

1. Why do we need to protect a network?
2. Describe how the network within your organisation is protected.

7.0 REFERENCES/FURTHER READING

http://www.azimuthsystems.com/

NASA Successfully Tests First Deep Space Internet. NASA media release 08-298, November 18, 2008 Archived.

MODULE 3 ENTERPRISE NETWORK

UNIT 1 CREATING ENTERPRISE NETWORK

CONTENTS

1.0 INTRODUCTION

This unit introduces you to enterprise network. An enterprise network is used to connect computers and other digital equipment of an organisation together.

2.0 OBJECTIVES

At the end of this unit, you should be able to:

- explain enterprise network
- describe predefined enterprise network.

3.0 MAIN CONTENT

3.1 Overview of Enterprise Network

An enterprise private network is a network built by an enterprise to interconnect various company sites, e.g., production sites, head offices, remote offices, shops, in order to share computer resources. Enterprise networks are used for configuring access rules in an enterprise policy that can be applied to any array in the enterprise and for configuring enterprise network rules that apply to all arrays in the enterprise. Only enterprise networks can be used to create enterprise-level rules. Enterprise networks can also be used for defining array-level access and publishing rules and for defining array-level network rules. However, you cannot use array-level networks when creating enterprise-level rules.

Any number of user-defined enterprise networks can also be included in a network defined in an array by including references to them. An enterprise network which its set of *IP* address range corresponds exactly to the *IP* addresses included in a protected array-level network, such as the Internal network, defined in one array can be used to reference that network in all the other arrays of the enterprise.

When you configure enterprise networks, you specify only the *IP* address ranges and do not specify any of the other properties that you would define for array-level networks. In particular, you cannot configure Network Load Balancing or Cache Array Routing Protocol (CARP) for an enterprise network.

The *IP* addresses that are included in an enterprise network are excluded from the default external network in each array in the enterprise, even if the enterprise network is not included in any network defined in the array.

3.2 Predefined Enterprise Networks

The following predefined enterprise networks are created upon installation:

- External
- Local Host
- Quarantined VPN Clients
- VPN Clients

These predefined enterprise networks implicitly define the same *IP* address sets as their array-level counterparts. They can be used for defining rules in an enterprise policy and for defining enterprise network rules. When an enterprise policy is assigned to an array, each predefined enterprise network in a rule will be interpreted as the array-level network of the same name. For example, you can create an enterprise access rule that applies to requests sent to the local host enterprise network. When a request is handled in an array to which the enterprise policy containing this access rule is assigned, the rule will apply to the

IP addresses in the local host network on the array member handling the request.

3.2.1 Residual Networks

IP addresses that belong to a configurable enterprise network, but do not belong to any configurable array-level network are considered to be part of a residual network.

3.4 Threat to Enterprise Network

Today, there is an ever-growing dependency on computer networks for business transactions. With the free flow of information and the high availability of many resources, managers of enterprise networks have to understand all the possible threats to their networks. These threats take many forms, but all result in loss of privacy to some degree and possibly malicious destruction of information or resources that can lead to large monetary losses.

3.4.1 Types of Threat

Listed below are some common types of threats- especially when the enterprise network is connected to the internet.

a. Insider threats

An insider attack involves someone from the inside, such as a disgruntled employee, attacking the network. Insider attacks can be malicious or not malicious. Malicious insiders intentionally eavesdrop, steal, or damage information; use information in a fraudulent manner; or deny access to other authorised users. This can be controlled using efficient authentication and authorisation systems and *IDS.*

b. Malware

One of the biggest threats to computer users on the internet today is malware. Malware are malicious codes that can cause distortion on your browser and eventually cripple your system. They can hijack your browser, redirect your search attempts, serve up nasty pop-up ads, track what web sites you visit, and prevent you from performing certain functions. Examples are worm, viruses, Trojan horse, spyware, adware, dialers, hijackers, etc. They are controlled using frequently updated and highly rated antivirus, antispyware, etc. Efficient management of patches, *SPAM* filters and two-way authentication are also used for their control.

c. Socially engineered attacks

Most traditional social engineering attacks capitalise on this vulnerability. Examples are phishing, *pharming*, masquerading. General technical controls such as a firewall, internet content filtering, antivirus software, anti-spam software, security awareness and patch management can help reduce or eliminate many phishing attacks.

d. Authentication attacks

An attacker tries to crack the passwords stored in a network account database or a password-protected file. Examples include dictionary, a brute-force, shoulder surfing, key logging and hybrid attacks. They are controlled using robust and multifactor authentication and authorisation systems.

SELF-ASSESSMENT EXERCISE

Mention other types of threats that are not listed here

4.0 CONCLUSION

This unit has taken you through the rudiment of enterprise network. It has also introduced you to residual networks.

5.0 SUMMARY

In this unit, you have learnt that:

- an enterprise private network is a network built by an enterprise to interconnect various company sites, e.g., production sites, head offices, remote offices, shops, in order to share computer resources.
- common types of threats to enterprise network are insider attacks, socially engineered, malware and authentication attacks.
- predefined enterprise network are used for defining rules in an enterprise policy and for defining enterprise network rules.

6.0 TUTOR-MARKED ASSIGNMENT

1. Discuss the three types of threat to enterprise network.
2. What do you understand by enterprise network?

7.0 REFERENCES/FURTHER READING

Abbate, Janet (1999). *Inventing the Internet*. Cambridge: MIT.

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UNIT 2 PLANNING AND SELECTION OF ENTERPRISE NETWORK

CONTENTS

1.0 INTRODUCTION

This unit gives you an overview of network and the steps involved in planning and selecting enterprise network.

2.0 OBJECTIVES

At the end of this unit, you should be able to:

- highlight how to make proper planning for every phase of an enterprise network
- explain the design phases in enterprise networks.

3.0 MAIN CONTENT

3.1 Overview

Network planning and design methodology describes a process with 9 specific steps and a sequence for those activities. It is an engineering life cycle that supports technical initiatives such as windows migration, *IP* telephony and wireless design, to name a few examples. The methodology begins with examining company business requirements. It is absolutely essential that you understand the company business model, business drivers and how they are growing from a business perspective. That will build the foundation for a design proposal that serves the business, technical and operational requirements of the company.

3.2 Steps to Effective Network Planning and Design

Here, let us look at the following steps.

Step 1- Business requirements

Any design project starts with an understanding of what the company does and what they need to accomplish from a business perspective. This begins with an understanding of their business model, which really describes how their company works from an operational and business perspective to generate revenues and reduce costs. Many vendors today have conducted their own return on investment (ROI) studies for new implementations such as unified communications and telephony. It is an effective sales tool that illustrates the cost benefits compared with investment over a specified period of time.

Some typical business drivers are listed below.

- Reduce operating costs
- Generate revenue
- Client satisfaction
- Employee productivity

This is a list of some typical project business requirements.

- Budget constraints
- Office consolidations
- Company mergers and acquisitions
- Business partner connectivity
- Telecommuter remote access
- New offices and employees
- New data center applications
- Reduce network outage costs
- Cost effective network management
- Vendor contracts

Step 2- Design requirements

Now that you have learnt the basic business requirements of the company, you can determine the standard and specific design requirements. The design requirements process is focused on defining requirements from a technical perspective. Those requirements along with the business requirements will build the framework that is used to define infrastructure, security and management. Design requirements are defined as standard and miscellaneous. The standard design requirements are generic and represent those considered with many design projects. Miscellaneous requirements are those that are not defined with any of the standard requirements.

Standard design requirements are:

- performance
- availability
- scalability
- standards compatibility
- rapid deployment

Step 3- Network assessment

A network assessment is conducted after we have finished the business and design requirements of the company. A network assessment provides a quick snapshot of the current network with an examination of the infrastructure, performance, availability, management and security. That information is utilised for making effective strategy recommendations and design proposals to the client concerning specific information systems modifications. The network assessment model has three sequential activities, which are- assessment, analysis and recommendations. The current network is examined using five primary surveys, namely- infrastructure, performance, availability, management and security. When the surveys are completed, the information collected is then reviewed for trends, problems and issues that are negatively affecting the network.

Step 4- Infrastructure selection

After doing a network assessment we are ready to start selecting specific infrastructure components for the network design. This phase starts building the infrastructure with a specific sequence that promotes effective equipment selection and design. It is important that you consider business requirements, design requirements and the network assessment when building your infrastructure.

The following numbered list describes the specific infrastructure components and their particular sequence.

a. Enterprise *WAN* topology
b. Campus topology
c. Traffic model
d. Equipment selection
e. Circuits
f. Routing protocol design
g. Addressing

h. Naming conventions
i. *IOS* Services
j. Domain name services
k. *DHCP* services

Step 5: Security strategy

We must now define a security strategy for securing the infrastructure. The need for enterprise network security should not be ignored with the proliferation of the Internet. Companies are continuing to leverage the public infrastructure for connecting national and international offices, business partners and new company acquisitions. The security requirements and network assessment recommendations should drive the selection of security equipment, protocols and processes. It identifies what assets must be protected, what users are allowed access and how those assets will be secured.

Step 6- Network management strategy

This section will define a network management strategy for managing all equipment defined from infrastructure and security. It is necessary to define how the equipment is going to be monitored and determine if the current management strategy is adequate or if new applications, equipment, protocols and processes must be identified. Management components are then integrated with infrastructure and security to finish building the proposed design. These primary elements comprise any well-defined management strategy and should be considered when developing your strategy.

- Management groups
- *SNMP* applications
- Monitored devices and events

Step 7- Proof of concept

All infrastructure, security and management components must now be tested with a proof of concept plan. It is important to test the current design, configuration and *IOS* versions in a non-production environment or on the production network with limited disruption. Implementation of newer network modules at a router, for instance, could require that you change the current *IOS* version that is implemented. Making those changes can affect *WAN* or campus modules already installed at production routers. That is the real value of doing a proof of concept and certifying that the new equipment and *IOS* versions integrate with each device as well as the network. The following list describes the advantages of doing a proof of concept with your network design. The proof of concept test results should be examined and used to modify current infrastructure, security and management specifications before generating a design proposal. The proof of concept model suggested here involves prototype design, equipment provisioning, defining tests, building equipment scripts and examining test results.

1. Prototype design
2. Provision of equipment
3. Define tests
4. Build equipment scripts
5. Review test results

Step 8- Design proposal/review

With the proof of concept finished, you are now ready to build a design proposal for the design review meeting. Your intended audience could be the Director, *CIO*, *CTO*, Senior network engineer, Consultant or anyone that is approving a budget for the project. It is important to present your ideas with clarity and professionalism. If a presentation is required, Powerpoint slides work well and could be used to support concepts from the design proposal document. The focus is on what comprises a standard design proposal and the sequence for presenting that information.

The working design proposal is presented to the client after addressing any concerns from proof of concept assurance testing. The design review is an opportunity for you to present your design proposal to the client and discuss any issues. It is an opportunity for the client to identify concerns they have and for the design engineer to clarify issues. The focus is to agree on any modifications, if required, and make changes to the infrastructure, security and management before implementation starts. Business and design requirements can change from when the project started which sometimes will necessitate changes to infrastructure, security and management specifications. Any changes should then go through proof of concept testing again before final changes to the design proposal.

Step 9- Implementation

The final step entails defining an implementation process for the specified design. This describes a suggested implementation methodology of the proposed design, which should have minimal disruption to the production network. As well it should be efficient and as cost effective as possible. As with previous methodologies there is a sequence that should be utilised as well.

Once the implementation is finished, the network has to be monitored to checkmate problems which may arise. Design and configuration modifications are then made to address any problems or concerns.

SELF-ASSESSMENT EXERCISE

A manufacturing company is interested in setting up a network. Describe how you will carry out network planning and design for the organisation.

4.0 CONCLUSION

In this unit, you have learnt the necessary steps to consider in implementing an enterprise network.

5.0 SUMMARY

In this unit, you have learnt that:

- network planning and design methodology is an important activity towards having an efficient network
- network planning and design involve nine specific steps, namely-identification of business requirements, design requirements, network assessment, infrastructure selection, security strategy, network management strategy, proof of concept and implementation.

6.0 TUTOR-MARKED ASSIGNMENT

1. What are the possible consequences of lack of effective network planning?
2. State and explain the phases involved in creating an enterprise network

7.0 REFERENCES/FURTHER READING

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UNIT 3 LOCAL AREA NETWORK (LAN) AND WIDE AREA NETWORK (WAN)

CONTENTS

1.0 INTRODUCTION

Computers networked together in a self-contained group form a Local Area Network or LAN. A *LAN* typically is contained within a single building or a group of neighbouring buildings. Two computers linked together at home are the simplest form of a *LAN*. Several hundred computers cabled together across several buildings at school form a more complex *LAN*. *LANs* are usually connected with coaxial or *CAT5* cable.

On the other hand, a *WAN* is geographically large. It is often formed by the joining together of *LANs* in distant places. A national banking organisation, for example, may use a *WAN* to connect all of its branches across the country. The difference between *LANs* and *WANs* is getting blurry as fibre optic cables have allowed *LAN* technologies to connect devices many kilometers apart. *WANs* are usually connected using the internet, *ISDN* landlines or satellite.

2.0 OBJECTIVES

At the end of this unit, you should be able to:

- discuss the background of *LAN* and *WAN*
- explain the differences between *LAN* and *WAN*
- state the relationship between *LAN* and *WAN*.

3.0 MAIN CONTENT

3.1 Understanding *WAN* and *LAN*

A Local Area Network (LAN) exists in a house or a university campus, while a Wide Area Network (WAN) exists across many office buildings separated by a vast distance. The office buildings in a *WAN* may be in different countries or even continents. For example, the headquarters building may be in *USA*, the regional office building may be in *UK*, and the branch office building may be in India. The workers in the three buildings can use *WAN* to collaborate with one another. The internet can also be considered as a *WAN*.

3.2 Differences between *LAN* and *WAN*

One of the differences between *LAN* and *WAN*, is the speed of the network. The maximum speed of a *LAN* can be 10 megabits per second, while the speed of a *WAN* can go up to 150 megabits per second. This means the speed of a *WAN*, is one-tenth of the speed of a *LAN*. A *WAN* is usually slower because it has lower bandwidth.

Computers in a *LAN* can share a printer, if they are all in the same *LAN*. On the other hand, a *WAN* cannot share a printer, so a computer in one country cannot use a printer in another country. A *LAN* does not need a dedicated computer to direct traffic to and from the internet, unlike a *WAN* that needs a special-purpose computer, whose only purpose is to send and receive data from the internet.

Another basis for comparing *LAN* and *WAN* is the cost of the network. A *WAN* is more expensive than a *LAN*. It is easier to expand a *LAN* than a *WAN*. The equipment needed for a *LAN* is a Network Interface Card (NIC), a switch and a hub. On the other hand, the equipment needed to connect a *WAN* to the internet is a modem and a router. The modem may be a cable modem or a *DSL* modem that is connected to a wall jack, while the router should be configured so that it can handle the packets traveling between the *WAN* and the internet.

Also, between *LAN* and *WAN*, there is a difference in the networking standard used. A *LAN* uses the ethernet standard, while a *WAN* uses the t1 standard. Before ethernet, the protocols used for *LAN* were Attached Resource Computer Network (ARCNET) and token ring. The protocols used for *WAN* are frame relay and Asynchronous Transfer Mode (ATM). Another protocol for *WAN* is Packet Over SONET/SDH (POS), where SONET stands for Synchronous Optical Networking and SDH stands for Synchronous Digital Hierarchy. The first *WAN* protocol was *x.25*, while an advanced *WAN* protocol is multiprotocol label switching (mpls). The hardware in a *LAN* is connected with *10base-t* cable connectors, while a *WAN* is connected via leased lines or satellites.

Furthermore, a *LAN* is easy to set up, as you need to slip the *NIC* into the *PCI* slot (for desktop computers) or *PCMCIA* slot (for laptop computers). You also need to install the driver for the *NIC*. The *NIC* can be connected to the network using the *RJ45* port.

On the other hand, a *WAN* is very difficult to set up. There is often an appliance to optimise the *WAN*. There is also a device to cache *WAN* data, so workers in the branch office can quickly access documents. The router also has quality of service (QoS) built in, so that it gives priority to certain kinds of traffic.

There are various topologies available in *LAN* and *WAN* networking. The most common topologies in *LAN* and *WAN* networks are ring and star. The ring topology is a network in which every node (every computer) is connected to exactly two other nodes. The star topology is a network in which all the nodes (called leaf nodes or peripheral nodes) are connected to a central node.

SELF-ASSESSMENT EXERCISE

Get valid diagrams to represent the structure of *LAN* and *WAN*

4.0 CONCLUSION

This unit has taken you through the nitty-gritty of Local Area Network and Wide Area Network classes. It also examined the relationship between *LAN* and *WAN* and also the differences between the two types of network.

5.0 SUMMARY

In this unit, you have learnt that:

- a local area network (LAN) exists in a house or a university campus, while a wide area network (WAN) exists across many office buildings separated by a vast distance.
- one of the differences between *LAN* and *WAN*, is the speed of the network.
- between *LAN* and *WAN*, *as well*, there is a difference in the networking standard used.

6.0 TUTOR-MARKED ASSIGNMENT

1. Discuss the differences between the *LAN* and the *WAN* classes.
2. Discuss other types of networks apart from *LAN* and *WAN*

7.0 REFERENCES/FURTHER READING

"http://en.wikipedia.org/wiki/History_of_the_Internet

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MODULE 4 COMMUNICATION TECHNOLOGY

UNIT 1 MODEM AND MODULATION CONCEPTS

CONTENTS

1.0 INTRODUCTION

A **modem** (**mo**dulator-**dem**odulator) is a device that modulates an analog carrier signal to encode digital information, and also demodulates such a carrier signal to decode the transmitted information. The goal is to produce a signal that can be transmitted easily and decoded to reproduce the original digital data. Modems can be used over any means of transmitting analog signals, from driven diodes to radio. The most familiar example is a voice band modem that turns the digital data of a personal computer into modulated electrical signals in the voice frequency range of a telephone channel. These signals can be transmitted over telephone lines and demodulated by another modem at the receiver side to recover the digital data.

Modulation is the process of facilitating the transfer of information over a medium. Sound transmission in air has limited range for the amount of power your lungs can generate. To extend the range your voice can reach, we need to transmit it through a medium other than air, such as a phone line or radio. The process of converting information (voice in this case) so that it can be successfully sent through a medium (wire or radio waves) is called modulation.

2.0 OBJECTIVES

At the end of this unit, you should be able to:

- define a modem
- explain modulation
- state the types of modulation
- describe common digital modulation techniques.

3.0 MAIN CONTENT

3.1 History

News wire services in 1920s used multiplex equipment that met the definition, but the modem function was incidental to the multiplexing function, so they are not commonly included in the history of modems. Modems grew out of the need to connect teletype machines over ordinary phone lines instead of more expensive leased lines which had previously been used for current loop-based teleprinters and automated telegraphs. George Stibitz connected a New Hampshire teletype to a computer in New York City by a subscriber telephone line in 1940.

In 1943, *IBM* adapted this technology to their unit record equipment and was able to transmit punched cards at 25 bits/second. Mass-produced modems in the United States began as part of the *SAGE* air-defense system in 1958; connecting terminals at various airbases, radar sites, and command-and-control centers to the *SAGE* director centers scattered around the U.S. and Canada. *SAGE* modems were described by AT&T's Bell Labs as conforming to their newly published Bell 101 dataset standard. While they ran on dedicated telephone lines, the devices at each end were no different from commercial acoustically coupled Bell 101, 110 baud modems.

In the summer of 1960, the name Data-phone was introduced to replace the earlier term digital subset. The 202 data-phone was a half-duplex asynchronous service that was marketed extensively in late 1960. In 1962, the 201A and 201B data-phones were introduced. They were synchronous modems using two-bit-per-baud phase-shift keying (PSK). The 201A operated half-duplex at 2,000 bit/s over normal phone lines, while the 201B provided full duplex 2,400 bit/s service on four-wire leased lines, the send and receive channels running on their own set of two wires each.

The famous Bell 103A dataset standard was also introduced by Bell Labs in 1962. It provided full-duplex service at 300 baud over normal phone lines. Frequency-shift keying was used with the call originator transmitting at *1,070 or 1,270 Hz* and the answering modem transmitting at *2,025* or *2,225 Hz*. The readily available *103A2* gave an important boost to the use of remote low-speed terminals such as the *KSR33*, the *ASR33*, and the *IBM 2741*. *AT&T* reduced modem costs by introducing the originate-only *113D* and the answer-only *113B/C* modems.

3.2 Significance of Digital Modulation

The aim of **digital modulation** is to transfer a digital bit stream over an analog band pass channel; for example over the public switched telephone network (where a band pass filter limits the frequency range to between 300 and 3400 Hz), or over a limited radio frequency band.

The aim of **analog modulation** is to transfer an analog baseband (or low pass) signal, for example an audio signal or *TV* signal, over an analog band pass channel, for example a limited radio frequency band or a cable *TV* network channel.

Analog and digital modulation facilitate Frequency Division Multiplexing (FDM), where several low pass information signals are transferred simultaneously over the same shared physical medium, using separate pass band channels.

The aim of **digital baseband modulation** methods, also known as line coding, is to transfer a digital bit stream over a baseband channel, typically a non-filtered copper wire such as a serial bus or a wired local area network.

The aim of **pulse modulation** methods is to transfer a narrowband analog signal, for example a phone call over a wideband baseband channel or, in some of the schemes, as a bit stream over another digital transmission system.

In music synthesisers, modulation may be used to synthesise waveforms with a desired overtone spectrum. In this case the carrier frequency is typically in the same order or much lower than the modulating waveform. See for example frequency modulation synthesis or ring modulation.

3.3 Analog Modulation Methods

In analog modulation, the modulation is applied continuously in response to the analog information signal.

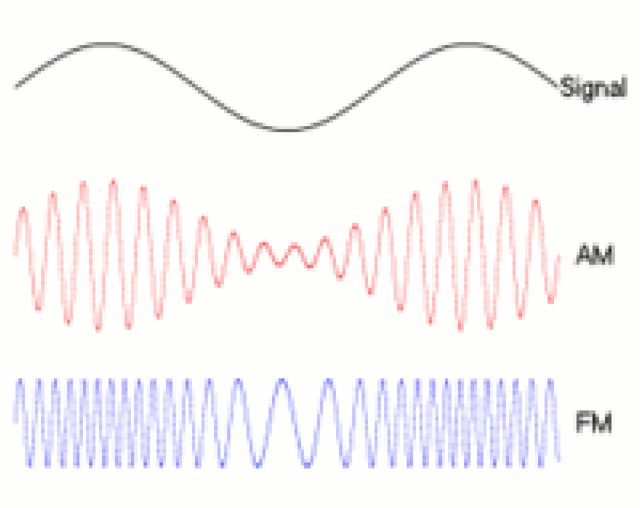

Figure 1.1: Analog Modulation Signal

A low-frequency message signal (top) may be carried by an *AM* or *FM* radio wave.

Common analog modulation techniques are as listed below.

- Amplitude Modulation (AM)- the amplitude of the carrier signal is varied in accordance to the instantaneous amplitude of the modulating signal
- Double-Sideband Modulation (DSB)

- Double-sideband modulation with carrier (DSB-WC) is used on the *AM* radio broadcasting band
- Double-sideband suppressed-carrier transmission (DSB-SC)
- Double-sideband reduced carrier transmission (DSB-RC)

- Single-sideband modulation (SSB, or SSB-AM),

- SSB with carrier (SSB-WC)
- SSB suppressed carrier modulation (SSB-SC)

- Vestigial sideband modulation (VSB, or VSB-AM)
- Quadrature Amplitude Modulation (QAM)
- Angle modulation
- Frequency Modulation (FM)- here the frequency of the carrier signal is varied in accordance to the instantaneous amplitude of the modulating signal.

- Phase Modulation (PM)- here the phase shift of the carrier signal is varied in accordance to the instantaneous amplitude of the modulating signal

The accompanying figure shows the results of (amplitude-) modulating a signal onto a carrier (both of which are sine waves). At any point along the *y-axis*, the amplitude of the modulated signal is equal to the sum of the carrier signal.

3.4 Digital Modulation Methods

In digital modulation, an analog carrier signal is modulated by a digital bit stream. Digital modulation methods can be considered as digital-to-analog conversion, and the corresponding demodulation or detection as analog-to-digital conversion. The changes in the carrier signal are chosen from a finite number of *M* alternative symbols (the modulation alphabet).

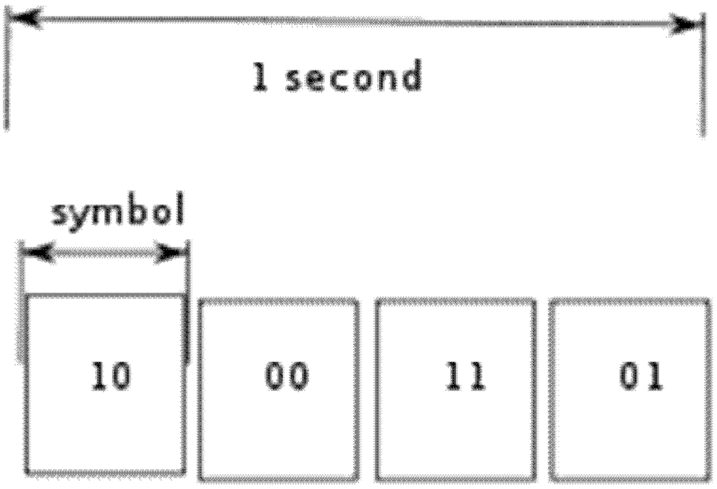

Figure 1.2: Schema of 4 Baud (8 bps) Data Link

A simple example- a telephone line is designed for transferring audible sounds- for example tones, and not digital bits (zeros and ones). Computers may however communicate over a telephone line by means of modems, which are representing the digital bits by tones, called symbols. If there are four alternative symbols (corresponding to a musical instrument that can generate four different tones, one at a time), the first symbol may represent the bit sequence 00, the second 01, the third 10 and the fourth 11. If the modem plays a melody consisting of 1000 tones per second, the symbol rate is 1000 symbols/second, or baud. Since each tone (i.e., symbol) represents a message consisting of two digital bits in this example, the bit rate is twice the symbol rate, i.e. 2000 bits per second. This is similar to the technique used by dialup modems as opposed to *DSL* modems.

3.5 Common Digital Modulation Techniques

These are listed below:

- Amplitude-Shift Keying (ASK)
- Frequency-Shift Keying (FSK)
- Phase-Shift Keying (PSK)

i. Amplitude-Shift Keying (ASK)

In *ASK*, the amplitude of the carrier is changed in response to information, and all else is kept fixed. *Bit 1* is transmitted by a carrier of one particular amplitude. To transmit *0*, we change the amplitude keeping the frequency constant. On-Off Keying (OOK) is a special form of *ASK*.

Advantages

Scheme is simple, so it is easy to implement transmitter and receiver with several components.
Low bandwidth requirements

Disadvantages

ASK is heavily effected by noise and interference and can be easily demodulated.

ii. Frequency-Shift Keying (FSK)

In *FSK*, we change the frequency in response to information, one particular frequency for *1* and another frequency for *0* as shown below for the same bit sequence. In the example below, frequency *f1* for bit 1 is higher than *f2* used for the 0 bit.

$$FSK(t) = \begin{cases} \sin(2\pi f_1 t) & \mathit{for\ bit}\ 1 \\ \sin(2\pi f_2 t) & \mathit{for\ bit}\ 0 \end{cases}$$

Advantages

FSK is insensitive to channel fluctuations and not easily effected by noise.

Resilient to signal strength variations

Does not require linear amplifiers in the transmitter

Disadvantages

FSK is a low performance type of digital modulation.

iii. Phase-Shift Keying (PSK)

In *PSK*, we change the phase of the sinusoidal carrier to indicate information. Phase, in this context, is the starting angle at which the sinusoid starts. To transmit *0*, we shift the phase of the sinusoid by *180*. Phase shift represents the change in the state of the information in this case.

$$PSK(t) = \begin{cases} \sin(2\pi f t) & for\, bit\, 1 \\ \sin(2\pi f t + \pi) & for\, bit\, 0 \end{cases}$$

Advantages

PSK, phase shift keying enables data to be carried on a radio communications signal in a more efficient manner than Frequency Shift Keying (FSK), and some other forms of modulation.

Disadvantages

Implementation is complex and expensive.

3.6 Fundamental Digital Modulation Methods

The most fundamental digital modulation techniques are based on keying:

- in the case of *PSK* (phase-shift keying), a finite number of phases are used.
- in the case of *FSK* (frequency-shift keying), a finite number of frequencies are used.
- in the case of *ASK* (amplitude-shift keying), a finite number of amplitudes are used.
- in the case of *QAM* (quadrature amplitude modulation), a finite number of at least two phases, and at least two amplitudes are used.

In *QAM*, an in-phase signal (the *I* signal, for example a cosine waveform) and a quadrature phase signal (the Q signal, for example a sine wave) are amplitude modulated with a finite number of amplitudes, and summed. It can be seen as a two-channel system, each channel using *ASK*. The resulting signal is equivalent to a combination of *PSK* and *ASK*.

In all of the above methods, each of these phases, frequencies or amplitudes are assigned a unique pattern of binary bits. Usually, each phase, frequency or amplitude encodes an equal number of bits. This number of bits comprises the symbol that is represented by the particular phase, frequency or amplitude.

If the alphabet consists of $M = 2^N$ alternative symbols, each symbol represents a message consisting of *N* bits. If the symbol rate (also known as the baud rate) is f_S symbols/second (or baud), the data rate is Nf_S *bit/second*.

For example, with an alphabet consisting of *16* alternative symbols, each symbol represents *4* bits. Thus, the data rate is four times the baud rate. In the case of *PSK*, *ASK* or *QAM*, where the carrier frequency of the modulated signal is constant, the modulation alphabet is often conveniently represented on a constellation diagram, showing the amplitude of the *I* signal at the x-axis, and the amplitude of the *Q* signal at the *y-axis*, for each symbol.

3.7 Modulator and Detector Principles of Operation

PSK and *ASK*, and sometimes also *FSK*, are often generated and detected using the principle of *QAM*. The *I and Q* signals can be combined into a complex-valued signal *I+jQ* (where j is the imaginary unit). The resulting so called equivalent low-pass signal or equivalent baseband signal is a complex-valued representation of the real-valued modulated physical signal (the so called pass band signal or *RF* signal). These are the general steps used by the modulator to transmit data.

1. Group the incoming data bits into code words, one for each symbol that will be transmitted.
2. Map the code words to attributes, for example amplitudes of the *I* and *Q* signals (the equivalent low pass signal), or frequency or phase values.
3. Adapt pulse shaping or some other filtering to limit the bandwidth and form the spectrum of the equivalent low pass signal, typically using digital signal processing.
4. Perform Digital-to-Analog Conversion (DAC) of the *I* and *Q* signals (since, today, all of the above is normally achieved using Digital Signal Processing DSP).
5. Generate a high-frequency sine wave carrier waveform, and perhaps also a cosine quadrature component. Carry out the modulation, for example by multiplying the sine and cosine wave form with the *I and Q* signals, resulting in that the equivalent low pass signal is frequency shifted into a modulated pass band signal or *RF* signal. Sometimes this is achieved using *DSP* technology, for example direct digital synthesis using a waveform table, instead of analog signal processing. In that case, the above *DAC* step should be done after this step.
6. Amplification and analog band pass filtering to avoid harmonic distortion and periodic spectrum

At the receiver side, the demodulator typically performs the following functions.

1. Band pass filtering.
2. Automatic Gain Control- AGC (to compensate for attenuation, for example fading).
3. Shifting the *RF* signal to the equivalent baseband *I and Q* signals, or to an Intermediate Frequency (IF) signal, by multiplying the *RF* signal with a local oscillator sine wave and cosine wave frequency (see the super heterodyne receiver principle).
4. Sampling an Analog-to-Digital Conversion (ADC) Sometimes before or instead of the above point, for example, by means of under sampling.
5. Equalisation filtering; for example a matched filter, compensation for multipath propagation, time spreading, phase distortion and frequency selective fading, to avoid symbol interference and symbol distortion.
6. Detection of the amplitudes of the *I and Q* signals, or the frequency or phase of the *IF* signals.
7. Quantisation of the amplitudes, frequencies or phases to the nearest allowed symbol values.
8. Mapping of the quantised amplitudes, frequencies or phases to code words (bit groups).

9. Parallel-to-serial conversion of the code words into a bit stream.
10. Pass the resultant bit stream on for further processing such as removal of any error-correcting codes.

As is common to all digital communication systems, the design of both the modulator and demodulator must be done simultaneously. Digital modulation schemes are possible because the transmitter-receiver pair have prior knowledge of how data is encoded and represented in the communications system. In all digital communication systems, both the modulator at the transmitter and the demodulator at the receiver are structured so that they perform inverse operations.

Non-coherent modulation methods do not require a receiver reference clock signal that is phase synchronized with the sender carrier wave. In this case, modulation symbols (rather than bits, characters, or data packets) are asynchronously transferred. The opposite is coherent modulation.

3.8 List of Known Digital Modulation Techniques

The most common digital modulation techniques are listed below.

- Phase-Shift Keying (PSK)-

- Binary *PSK* (BPSK), using M=2 symbols
- Quadrature *PSK* (QPSK), using M=4 symbols
- *8PSK*, using M=8 symbols
- *16PSK*, using M=16 symbols
- Differential *PSK* (DPSK)
- Differential *QPSK* (DQPSK)
- Offset *QPSK* (OQPSK)
- π/4–*QPSK*

- Frequency-Shift Keying (FSK)-

- Audio Frequency-Shift Keying (AFSK)
- Multi-Frequency Shift Keying (M-ary FSK or MFSK)
- Dual-Tone Multi-Frequency (DTMF)
- Continuous-Phase Frequency-Shift Keying (CPFSK)

- Amplitude-Shift Keying (ASK)
- On-Off Keying (OOK), the most common *ASK* form

- M-ary vestigial sideband modulation, for example *8VSB*

- Quadrature amplitude modulation (QAM) - a combination of *PSK* and *ASK*-

- Polar modulation like *QAM* a combination of *PSK* and *ASK*.

- Continuous Phase Modulation (CPM) methods-

- Minimum-Shift Keying (MSK)
- Gaussian Minimum-Shift Keying (GMSK)

- Orthogonal Frequency-Division Multiplexing (OFDM) modulation-
- Discrete Multi-Tone (DMT) - including adaptive modulation and bit-loading.

- Wavelet modulation
- Trellis Coded Modulation (TCM), also known as trellis modulation
- Spread-spectrum techniques

- Direct-Sequence Spread Spectrum (DSSS)
- Chirp Spread Spectrum (CSS) according to *IEEE 802.15.4a CSS* uses pseudo-stochastic coding
- Frequency-Hopping Spread Spectrum (FHSS) applies a special scheme for channel release

SELF-ASSESSMENT EXERCISE

Differentiate between Amplitude-Shift Keying (ASK) and Frequency-Shift Keying (FSK).

4.0 CONCLUSION

In this unit, you have been introduced to the fundamental concepts of modem and modulation. You also learnt about types of modulation; you were also introduced to various methods of modulation. This knowledge will boost your understanding of *ICT* concepts.

5.0 SUMMARY

In this unit, you have learnt that:

- a modem (modulator-demodulator) is a device that modulates an analog carrier signal to encode digital information, and also demodulates such a carrier signal to decode the transmitted information.

- the aim of digital modulation is to transfer a digital bit stream over an analog channel
- common digital modulation techniques are Amplitude-Shift Keying (ASK), Frequency-Shift Keying (FSK) and Phase-Shift Keying (PSK).
-

6.0 TUTOR-MARKED ASSIGNMENT

1. How does the modem perform its function?
2. Explain what you understand by the term “modulation”.
3. Describe the different types of modulation.

7.0 REFERENCES/FURTHER READING

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UNIT 2 MULTIPLEXERS

CONTENT

1.0 INTRODUCTION

In electronics, a **multiplexer** or ***mux*** is a device that selects one of several analog or digital input signals and forwards the selected input into a single line. A multiplexer of 2^n inputs has *n* select lines, which are used to select which input line to send to the output. An electronic multiplexer makes it possible for several signals to share one device or resource- for example, one *A/D* converter or one communication line, instead of having one device per input signal. On the other end, a demultiplexer (or ***demux***) is a device taking a single input signal and selecting one of many data-output-lines, which is connected to the single input. A multiplexer is often used with a complementary demultiplexer on the receiving end. An electronic multiplexer can be considered as a multiple-input, single-output switch, and a demultiplexer as a single-input, multiple-output switch.

2.0 OBJECTIVES

At the end of this unit, you should be able to:

- explain the meaning of a multiplexer
- describe the types of multiplexers
- state the areas of applications of multiplexers.

3.0 MAIN CONTENT

3.1 Types of Multiplexing

Multiplexing technologies may be divided into several types, all of which have significant variations, namely- Space-Division Multiplexing (SDM), Frequency-Division Multiplexing (FDM), Time-Division Multiplexing (TDM), and Code Division Multiplexing (CDM). Variable bit rate digital bit streams may be transferred efficiently over a fixed bandwidth channel by means of statistical multiplexing, for example packet mode communication. Packet mode communication is an asynchronous mode time-domain multiplexing which resembles time-division multiplexing.

Digital bit streams can be transferred over an analog channel by means of code-division multiplexing (CDM) techniques such as frequency-hopping spread spectrum (FHSS) and direct-sequence spread spectrum (DSSS).

In wireless communications, multiplexing can also be accomplished through alternating polarization (horizontal/vertical or clockwise/counterclockwise) on each adjacent channel and satellite, or through phased multi-antenna array combined with a Multiple-input multiple-output communications (MIMO) scheme.

3.1.1 Space-Division Multiplexing

In wired communication, space-division multiplexing simply implies different point-to-point wires for different channels. Examples include an analogue stereo audio cable, with one pair of wires for the left channel and another for the right channel, and a multi-pair telephone cable. Another example is a switched star network such as the analog telephone access network (although inside the telephone exchange or between the exchanges, other multiplexing techniques are typically employed) or a switched ethernet network. A third example is a mesh network. Wired space-division multiplexing is typically not considered as multiplexing.

In wireless communication, space-division multiplexing is achieved by multiple antenna elements forming a phased array antenna. Examples are Multiple-Input and Multiple-Output (MIMO), Single-Input and Multiple-Output (SIMO) and Multiple-Input and Single-Output (MISO) multiplexing. For example, a *IEEE 802.11n* wireless router with *N* antennas makes it possible to communicate with *N* multiplexed channels, each with a peak bit rate of 54 *Mbit/s*, thus increasing the total peak bit rate with a factor *N*. Different antennas will give different multi-path propagation (echo) signatures, making it possible for digital signal processing techniques to separate different signals from each other. These techniques may also be utilised for space diversity (improved robustness to fading) or beam-forming (improved selectivity) rather than multiplexing.

3.1.2 Frequency-Division Multiplexing

Frequency-division multiplexing (FDM): The spectrums of each input signal are swifted in several distinct frequency ranges.

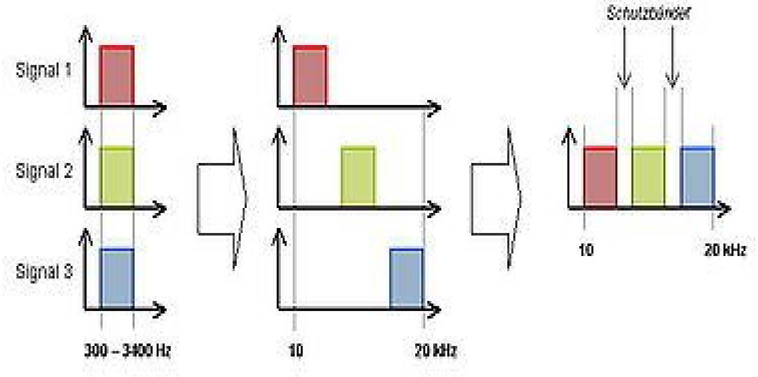

Figure 2. 1: Frequency-Division Multiplexing

Frequency-Division Multiplexing (FDM) is inherently an analog technology. *FDM* achieves the combining of several digital signals into one medium by sending signals in several distinct frequency ranges over that medium.

One of *FDM*'s most common applications is cable television. Only one cable reaches a customer's home but the service provider can send multiple television channels or signals, simultaneously, over that cable to all subscribers. Receivers must tune to the appropriate frequency (channel) to access the desired signal.

A variant technology, called Wavelength-Division Multiplexing (WDM) is used in optical communications.

3.1.3 Time-Division Multiplexing

Time-Division Multiplexing (TDM) is a digital technology. *TDM* involves sequencing groups of a few bits or bytes from each individual input stream, one after the other, and in such a way that they can be associated with the appropriate receiver. If done sufficiently and quickly, the receiving devices will not detect that some of the circuit time was used to serve another logical communication path.

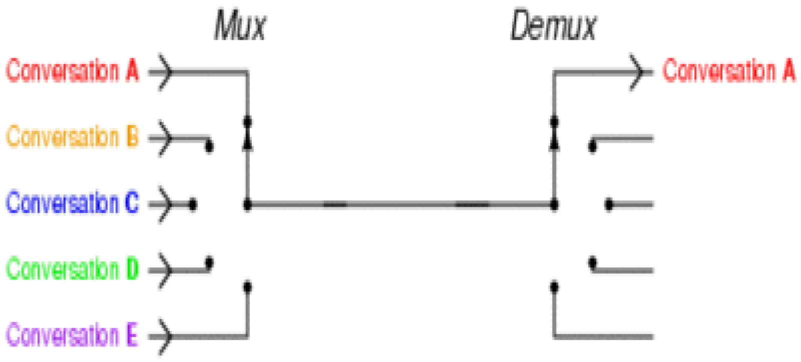

Figure2. 2: Time-Division Multiplexing (TDM)

Consider an application requiring four terminals at an airport to reach a central computer; each terminal communicates at *2400 bps*, so rather than acquire four individual circuits to carry such a low-speed transmission, the airline has installed a pair of multiplexers. A pair of *9600 bps* modems and one dedicated analog communications circuit from the airport ticket desk back to the airline data center are also installed.

3.1.4 Code-Division Multiplexing

Code division multiplexing is a technique in which each channel transmits its bits as a coded channel-specific sequence of pulses. This coded transmission typically is accomplished by transmitting a unique time-dependent series of short pulses, which are placed within chip times within the larger bit time. All channels, each with a different code, can be transmitted on the same fiber and asynchronously demultiplexed. Other widely used multiple access techniques are Time Division Multiple Access (TDMA) and Frequency Division Multiple Access (FDMA).

Code Division Multiplexing (CDM) techniques are used as an access technology, namely Code Division Multiple Access (CDMA), in Universal Mobile Telecommunications System (UMTS) standard for the third generation (3G) mobile communication identified by the *ITU*. Another important application of the CDMA is the Global Positioning System (GPS).

However, the term *Code Division Multiple Access* is also widely used to refer to a group of specific implementations of *CDMA* defined by Qualcomm for use in digital cellular telephony, which include *IS-95 and IS-2000*. The two different uses of this term can be confusing. Actually, *CDMA* (the Qualcomm standard) and *UMTS* have been competing for adoption in many markets.

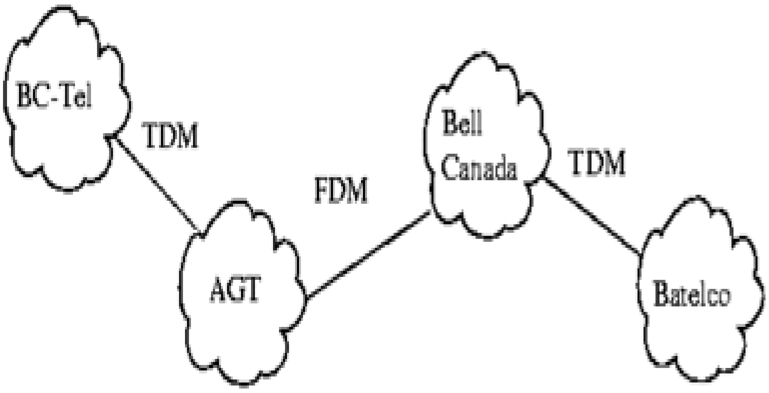

Figure 2.3: Telecommunication Multiplexing

SELF-ASSESSMENT EXERCISE

Find out about spread spectrum multiplexing.

3.2 Relation to multiple access

A multiplexing technique may be further extended into a multiple access method or channel access method, for example *TDM* into Time-Division Multiple Access (TDMA) and statistical multiplexing into Carrier Sense Multiple Access (CSMA). A multiple access method makes it possible for several transmitters connected to the same physical medium to share its capacity.

Multiplexing is provided by the physical layer of the *OSI* model, while multiple access also involves a media access control protocol, which is part of the data link layer. The transport layer in the *OSI* model as well as *TCP/IP* model provides statistical multiplexing of several application layer data flows to/from the same computer.

3.3 Application Areas

Let us consider the following areas.

a. Telegraphy

The earliest communication technology- using electrical wires, and therefore sharing an interest in the economies afforded by multiplexing, was the electric telegraph. Early experiments allowed two separate messages to travel in opposite directions simultaneously, first using an electric battery at both ends, then at only one end.

- Émile Baudot developed a time-multiplexing system of multiple Hughes machines in the 1870s.
- In 1874, the quadruplex telegraph developed by Thomas Edison transmitted two messages in each direction simultaneously, for a total of four messages transiting the same wire at the same time.
- Several workers were investigating acoustic telegraphy, a frequency-division multiplexing technique, which led to the invention of the telephone.

b. Telephony

In telephony, a customer's telephone line now typically ends at the remote concentrator box down the street, where it is multiplexed along with other telephone lines for that neighborhood or other similar area. The multiplexed signal is then carried to the central switching office on significantly fewer wires and for much further distances than a customer's line can practically go. This is likewise also true for Digital Subscriber Lines (DSL).

Fiber In The Loop (FITL) is a common method of multiplexing, which uses optical fiber as the backbone. It not only connects *POTS* phone lines with the rest of the *PSTN*, but also replaces *DSL* by connecting directly to ethernet wired into the home. Asynchronous Transfer Mode is often the communications protocol used.

Since all of the phone (and data) lines have been clumped together, none of them can be accessed except through a demultiplexer. This provides for more-secured communications, though they are not typically encrypted.

The concept is also now used in cable *TV*, which is increasingly offering the same services as telephone companies. IPTV also depends on multiplexing.

c. Video processing

In video editing and processing systems, multiplexing refers to the process of interleaving audio and video into one coherent *MPEG* transport stream (time-division multiplexing).

In digital video, such a transport stream is normally a feature of a container format which may include metadata and other information, such as subtitles. The audio and video streams may have variable bit rate. Software that produces such a transport stream and/or container is commonly called a statistical multiplexor or ***muxer***. A ***demuxer*** is a software that extracts or, otherwise, makes available for separate processing the components of such a stream or container.

d. Digital broadcasting

In digital television and digital radio systems, several variable bit-rate data streams are multiplexed together to a fixed bit rate transport stream by means of statistical multiplexing. This makes it possible to transfer several video and audio channels simultaneously over the same frequency channel, together with various services.

In the digital television systems, this may involve several Standard Definition Television (SDTV) programmes (particularly on *DVB-T, DVB-S2*, *ISDB* and *ATSC-C*), or one *HDTV*, possibly with a single *SDTV* companion channel over *6* to *8 MHz-wide TV* channel. The device that accomplishes this is called a statistical multiplexer. In several of these systems, the multiplexing results in an *MPEG* transport stream. The newer *DVB* standards *DVB-S2* and *DVB-T2* has the capacity to carry several *HDTV* channels in one multiplex. Even the original *DVB* standards can carry more *HDTV* channels in a multiplex if the most advanced *MPEG-4* compressions hardware is used.

On communications satellites which carry broadcast television networks and radio networks, this is known as **Multiple Channels Per Carrier** or **MCPC**. Where multiplexing is not practical (such as where there are different sources using a single transponder), single channel per carrier mode is used.

Signal multiplexing of satellite *TV* and radio channels is typically carried out in a central signal playout and uplink centre, such as *ASTRA* Platform Services in Germany, which provides play out, digital archiving, encryption, and satellite uplinks, as well as multiplexing, for hundreds of digital *TV* and radio channels.

In digital radio, both the *Eureka 147* system of digital audio broadcasting and the in-band on-channel *HD* Radio, *FMeXtra*, and digital radio *mondiale* systems can multiplex channels. This is essentially required with *DAB*-type transmissions (where a multiplex is called an **ensemble**), but is entirely optional with *IBOC* systems.

e. Analog broadcasting

In *FM* broadcasting and other analog radio media, multiplexing is a term commonly given to the process of adding subcarriers to the audio signal before it enters the transmitter, where modulation occurs. Multiplexing in this sense is sometimes known as ***MPX***, which in turn is also an old term for stereophonic *FM*, seen on stereo systems since the 1960s.

4.0 CONCLUSION

In this unit, you have been introduced to the fundamental concepts of Multiplexers. You also learnt about the types of multiplexers. You were also introduced to the various application areas of multiplexers. All these will go a long way to enrich your understanding of this course as a whole.

5.0 SUMMARY

In this unit, you have learnt that:

- a multiplexer is a device that selects one of several analog or digital input signals and forwards the selected input into a single line
- types of multiplexing are Space-Division Multiplexing (SDM), Frequency-Division Multiplexing (FDM), Time-Division Multiplexing (TDM), and Code Division Multiplexing (CDM).
- some application areas of multiplexing are telegraphy, telephony, video processing, digital broadcasting, analog broadcasting.

6.0 TUTOR-MARKED ASSIGNMENT

1. How is space-division multiplexing is achieved in wireless communication?
2. Describe the implementation of *FDM* in cable television
3. How is multiplexing related to *OSI* model?

7.0 REFERENCES/FURTHER READING

"Federal Standard 1037C: Glossary of Telecommunications Terms". Institute for Telecommunication Services. http://www.its.bldrdoc.gov/fs-1037/fs-1037c.htm. Retrieved 2009-10-19.

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UNIT 3 DIGITAL TECHNOLOGIES

1.0 INTRODUCTION

A **digital** system is a data technology that uses discrete (discontinuous) values. By contrast, non-digital (or analog) systems use a continuous range of values to represent information. Although digital representations are discrete, the information represented can be either discrete, such as numbers, letters or icons, or continuous, such as sounds, images, and other measurements of continuous systems. The word *digital* comes from the same source as the word digit and *digitus* (the Latin word for *finger*), as fingers are used for discrete counting. It is most commonly used in computing and electronics, especially where real-world information is converted to binary numeric form as in digital audio and digital photography.

2.0 OBJECTIVES

At the end of this unit, you should be able to:

- explain the meaning of a digital system
- state the properties of digital information.

3.0 MAIN CONTENT

3.1 History

Although, digital signals are generally associated with the binary electronic digital systems used in modern electronics and computing, digital systems are actually ancient, and need not be binary or electronic. Below are some examples of digital signals that are neither binary nor electronic.

- Written text in books (due to the limited character set and the use of discrete symbols - the alphabet in most cases)

- An *abacus* was created sometime between 1000 *BC* and 500 *BC*; it later became a form of calculation frequency. Nowadays it can be used as a very advanced, yet, basic digital calculator that uses beads on rows to represent numbers. Beads only have meaning in discrete up and down states, not in analog in-between states.
- A *beacon* is perhaps the simplest non-electronic digital signal, with just two states (on and off). In particular, *smoke signals* are one of the oldest examples of a digital signal, where an analog "carrier" (smoke) is modulated with a blanket to generate a digital signal (puffs) that conveys information.
- Morse code uses six digital states—dot, dash, intra-character gap (between each dot or dash), short gap (between each letter), medium gap (between words), and long gap (between sentences)—to send messages via a variety of potential carriers such as electricity or light, for example using an electrical telegraph or a flashing light.
- The Braille system was the first binary format for character encoding, using a six-bit code rendered as dot patterns.
- Flag semaphore uses rods or flags held in particular positions to send messages to the receiver watching them some distance away.
- International maritime signal flags have distinctive markings that represent letters of the alphabet to allow ships to send messages to each other.
- More recently invented, a modem modulates an analog "carrier" signal (such as sound) to encode binary electrical digital information, as a series of binary digital sound pulses. A slightly earlier, surprisingly reliable version of the same concept was to bundle a sequence of audio digital "signal" and "no signal" information (i.e. "sound" and "silence") on magnetic cassette tape for use with early home computers.

3.2 Properties of Digital Information

All digital information possesses common properties that distinguish it from analog communications methods. Let us consider these one by one.

- **Synchronisation-** since digital information is conveyed by the sequence in which symbols are ordered, all digital schemes have some method for determining the beginning of a sequence. In written or spoken human languages synchronisation is typically provided by pauses (spaces), capitalisation, and punctuation. Machine communications typically use special synchronisation sequences.
- **Language-** all digital communications require a *language,* which in this context consists of all the information that the sender and

receiver of the digital communication must both possess, in advance, in order for the communication to be successful. Languages are generally arbitrary and specify the meaning to be assigned to particular symbol sequences, the allowed range of values, methods to be used for synchronisation, etc.

- **Errors-** disturbances (noise) in analog communications invariably introduce some, generally small deviation or error between the intended and actual communication. Disturbances in a digital communication do not result in errors unless the disturbance is so large as to result in a symbol being misinterpreted as another symbol or disturb the sequence of symbols. It is therefore, generally, possible to have an entirely error-free digital communication. Furthermore, techniques such as check codes may be used to detect errors and guarantee error-free communications through redundancy or retransmission. Errors in digital communications can take the form of *substitution errors* in which a symbol is replaced by another symbol, or *insertion/deletion* errors in which an extra incorrect symbol is inserted into or deleted from a digital message. Uncorrected errors in digital communications have unpredictable and generally large impact on the information content of the communication.
- **Copying-** because of the inevitable presence of noise, making many successive copies of an analog communication is not feasible because each generation increases the noise. Since digital communications are generally error-free, copies of copies can be made indefinitely.
- **Granularity-** when a continuously variable analog value is represented in digital form there is always a decision as to the number of symbols to be assigned to that value. The number of symbols determines the precision or resolution of the resulting datum. The difference between the actual analog value and the digital representation is known as *quantisation error*.

SELF-ASSESSMENT EXERCISE

State other properties of digital information apart from those stated.

4.0 CONCLUSION

In this unit, you have been introduced to the fundamental concepts of digital technologies. You have also learnt the history of digital technologies and some of the properties of digital technologies.

5.0 SUMMARY

In this unit, you have learnt that:

- a digital system is a data technology that uses discrete (discontinuous) values; and that non-digital (or analog) systems use a continuous range of values to represent information.
- digital signals are generally associated with the binary electronic digital systems used in modern electronics and computing
- some properties of digital information are synchronisation, language requirement, low errors, high replication capability and granularity.

6.0 TUTOR-MARKED ASSIGNMENT

1. What are the differences between digital and analog signals?
2. What property of the digital signal makes it better than analog signal?

7.0 REFERENCES/FURTHER READING

Clark, David D. (1988). "The Design Philosophy of the DARPA Internet Protocols". *Computer Communications Review*, 18(4), pp. 106–114.

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UNIT 4 SIGNAL TRANSMISSION AND IMPAIRMENT

CONTENT

1.0 INTRODUCTION

Transmission impairment is a property of a transmission medium which causes the signal to be degraded, reduced in amplitude, distorted or contaminated. Impairment can introduce errors into digital signals. Examples of transmission impairments are attenuation, delay distortion, and several sources of noise including, thermal noise, impulse noise, and inter-modulation noise. It is important to understand transmission impairments for several reasons. Understanding the source of a transmission impairment like attenuation or dispersion will enable the user to partially correct for (equalize the signal) these effects. Understanding the source of transmission impairments (dispersion, attenuation, impulse noise, and thermal noise) can also help the user understand some of the constraints placed on the transmission of data as a result of these effects. Such constraints include the maximum length of network links, the choice of physical transmission media, the choice of encoding methods, and the data rate supported by the medium.

Attenuation is a property of the transmission medium. It measures how much energy is absorbed and/or radiated from the traveling signal due to it’s interaction with the transmission medium. Attenuation is measured as a function of the distance traveled through the transmission medium. The transmission medium absorbs energy because the signal is influenced by small impurities within it. Such impurities have different sizes and distributions depending on the type of medium. Impurities of different sizes effect different frequencies in the signal.

The effect of attenuation is, therefore, a function of frequency. The frequency variation of attenuation can be partially corrected, or equalised, by applying corrections based on a physical model. When a signal is attenuated, its amplitude is reduced. The interpretation of a received signal depends on being able to tell the difference between different signal levels. If the amplitude is reduced too much by attenuation it becomes impossible to accurately tell the difference between the different signal levels, and the information in the signal is lost.

2.0 OBJECTIVES

At the end of this unit, you should be able to:

- define transmission impairment
- describe the types of transmission impairment
- state the effects of noise
- explain the bit error rate
- highlight the differences between analog and digital transmission.

3.0 MAIN CONTENT

3.1 Types of Impairment

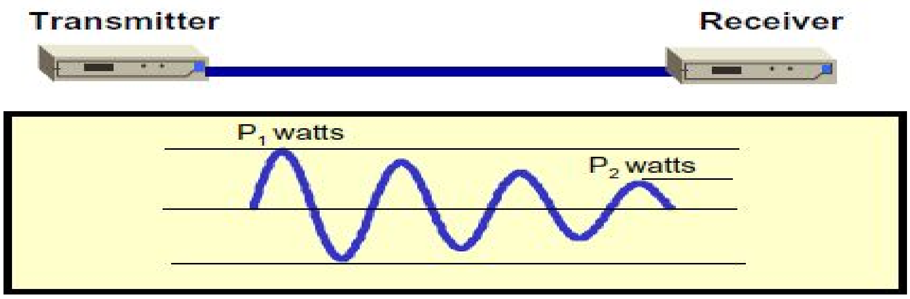
The different types of impairment which shall be treated in details in this unit are:

- attenuation
- delay distortion
- noise

Let us now consider this one after the other.

3.1.1 Attenuation

Signal amplitude decrease along a transmission medium. This is known as *signal attenuation*. Amplifiers (in case of analog signals) or repeaters (in case of digital signal) are inserted at intervals along the medium to improve the received signal as closed as to its original level. Attenuation and amplification are measured in *decibel* (db), which is expressed as a constant number of decibels per unit distance.



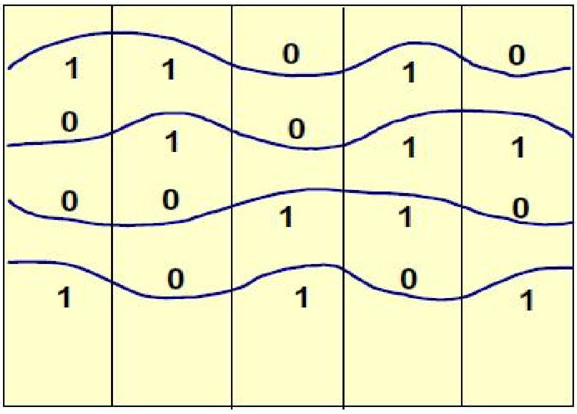
Attenuation $10 \log_{10} (P_1/P_2)$ dB

Amplification $10 \log_{10} (P_2/P_1)$ dB

Figure 4.1: Attenuated and Amplified Signal along the Transmission Media

3.1.2 Delay Distortion

The various frequency components in digital signal arrive at the receiver with varying delays, resulting in *delay distortion*. As bit rate increase, some of the frequency components associated with each bit transition are delayed and start to interfere with frequency components associated with a later bit, causing *inter-symbol interference*, which is a major limitation to maximum bit rate.



- Velocity of propagation of a signal through a guided medium varies with frequency
- Signal components of one bit position will spill over into other bit position
- Results : limit max, bit rate transmission
- Solving : equalizing

Figure 4.2: Velocity of Propagation of a Signal through Guided Media

3.1.3 Noise

Signal-to-noise ratio (S/N) is a parameter used to quantify how much noise there is in a signal. A high *SNR* means a high power signal relative to noise level, resulting in a good-quality signal.

S/N is given as
$S/N_{db} = 10 \log_{10} S/N$
Where S = average signal power
N = noise power

3.2 Noise Types

Here, we are going to consider some common examples.

Atmospheric noise

- Lightning - static discharge of clouds
- Solar noise - sun’s ionised gases
- Cosmic noise - distant stars radiate high frequency signal

Gaussian noise

Thermal noise- generated by random motion of free electrons

Impulse noise - sudden bursts of irregularly pulses

Crosstalk- this is interference generated when magnetic fields or current nearby wires interrupt electrical current in a wire. As electrical current travels through a wire, the current generates a magnetic field. Magnetic field from wires that are closed together can interfere with each other. Shielding the wire and twisting wire pairs around each other help decrease crosstalk

NEXT (near-end crosstalk)

- interference in a wire at the transmitting end of a signal sent on a different wire FEXT (far-end crosstalk)
- interference in a wire at the receiving end of a signal sent on a different wire

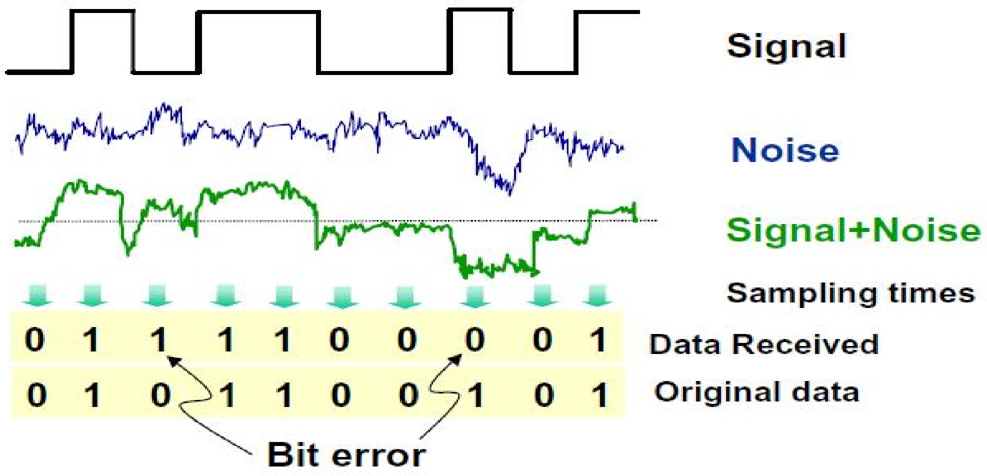

Signal
Noise
Signal+Noise
Sampling times
0 1 1 1 1 0 0 0 0 1 Data Received
0 1 0 1 1 0 0 1 0 1 Original data
Bit error

Figure 4.3: Effect of Noise

Impulse noise is the primary source of error for digital data. A sharp spike of energy of *0.01* seconds duration will not destroy any voice data, but will wash out many bits of digital data.

3.3 Bit Error Rate

- BER (Bit Error Rate) is the probability of a single bit being corrupted in a define time interval.
- *BER* of 10-5 means on average 1 bit in 10-5 will be corrupted
- A *BER* of 10-5 over voice-graded line is typical.
- *BER* of less than 10-6 over digital communication is common.

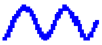
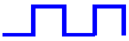
A Bit Error Rate (BER) is a significant measure of system performance in terms of noise. A *BER* of 10-6, for example, means that one bit of every million may be destroyed during transmission. Several factors that affect *BER* are as listed below.

- Bandwidth
- S/N
- Transmission medium
- Transmission distance
- Environment
- Performance of transmitter and receiver

3.4 Analog and Digital Transmission

Analog signal is characterised by being continuously variable along amplitude and frequency. Digital signal is characterised by series of discrete pulses, representing one and zero bits. Table4.1 presents the comparison between analog and digital transmission.

Table4.1: Analog and Digital Transmissions

	Analog	**Digital**
Data	continuous (e.g., voice)	discrete (e.g., text)
Signal	continuous electromagnetic waves  Used mainly for transmitting data across a network.	sequence of voltage pulses  Used mainly internally within computers.
transmission	Transmission of analog signals without regards to their content (the data may be analog or binary). The signals become weaker (attenuated) with the distance. Amplifiers may be used to strengthen the signals, but as side effect they also boost the noise. This might not be a problem for analog data, such as voice, but is a problem for digital data.	Transmission that is concerned with the content of the signal. Repeaters are used to overcome attenuation. A repeater recovers the digital pattern from the signal it gets, and resubmits a new signal.

3.4.1 Advantages of Digital Transmission over Analog

- **Technology-** there is a drop in cost due to *LSI* and *VLSI*
- **Data integrity-** repeaters allow longer distances over lines of lesser quality.
- **Capacity utilisation-** digital techniques can be more easily and cheaply utilised, through multiplexing, available transmission links of high bandwidth.
- **Security and privacy-** encryption techniques are more readily applied to digital data
- **Integration-** simplified, if digitised data is used everywhere.

SELF-ASSESSMENT EXERCISE

State the advantages of analog transmission over digital.

3.5 Signal Transmission Analysis

The analysis of electrical signals is a fundamental problem for many engineers and scientists. Even, if the immediate problem is not electrical, the basic parameters of interest are often changed into electrical signals by means of transducers. Common transducers include accelerometers and load cells in mechanical work, *EEG* electrodes and blood pressure probes in biology and medicine, and *pH* and conductivity probes in chemistry. The rewards for transforming physical parameters to electrical signals are great, as many instruments are available for the analysis of electrical signals in the time, frequency and modal domains. The powerful measurement and analysis capabilities of these instruments can lead to rapid understanding of the system under study.

4.0 CONCLUSION

In this unit, you have been introduced to the fundamental concepts of transmission impairment. You also learnt the types of transmission impairments. You were also introduced to bit error rate and some of the major differences between analog and digital transmissions.

5.0 SUMMARY

In this unit, you have learnt that:

- Transmission impairment is a property of a transmission medium which causes the signal to be degraded, reduced in amplitude, distorted or contaminated.
- Examples of transmission impairments are attenuation, delay distortion, and noise
- Types of noise include thermal noise, impulse noise, and inter-modulation noise
- Analog signal is characterised by being continuously variable along amplitude and frequency. Digital signal is characterised by series of discrete pulses, representing one and zero bits.

6.0 TUTOR-MARKED ASSIGNMENT

a. What do you understand by transmission impairment

b. What is the difference between *FEXT* and *NEXT*?

c. What do you understand by the term “noise”?

7.0 REFERENCES/FURTHER READING

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MODULE 5 NETWORK TECHNOLOGIES

UNIT 1 INTEGRATED SERVICE DIGITAL NETWORK (ISDN)

CONTENTS

1.0 INTRODUCTION

Having read through the course guide, you will have a general understanding of what this unit is about and how it fits into the course as a whole. This unit will describe the general concept of *ISDN*.

2.0 OBJECTIVES

At the end of this unit, you should be able to:

- explain the term *ISDN*
- describe *ISDN* interfaces
- discuss how *ISDN* works
- state the usefulness of *ISDN*

3.0 MAIN CONTENT

3.1 What is Integrated Services Digital Network (ISDN)?

Integrated Services Digital Network (*ISDN*) is a digital communications technology that enables a small business or an individual to connect directly to the internet and other sites/users (for videoconferencing, for instance). *ISDN* provides a standard interface for voice, fax, video, graphics, and data – all on a single telephone line.

"***Integrated services***" refers to *ISDN's* ability to deliver two simultaneous connections, in any combination of voice, fax, data, and video, over a single line. Multiple devices can be attached to the line, and used as needed.

"***Digital***" refers to the fact that it is a purely digital transmission, as opposed to the analog transmission method used by conventional telephone lines.

"***Network***" refers to the fact that *ISDN* is not simply a point-to-point connection like a leased telephone line – *ISDN* networks extend from the local telephone exchange to the remote user, and include all the switching equipment in between. If your *ISDN* equipment includes analog capabilities, you can also connect to telephones, fax machines, and analog modems – even though they may be connected to standard analog telephone lines.

ISDN service is provided by the same companies that provide telephone service – you get much faster, more dependable connections for voice, fax, data, and video – all through a single connection. While not new (*ISDN* has been in use for over 15 years), the advent of international standards has made *ISDN* viable as telephone companies around the world have upgraded their equipment to these *ISDN* standards. It is now commonly available in Europe, Japan, Australia, and from most major North American telephone companies – *AT&T*, *MCI*, and *Sprint* can provide long-distance *ISDN* lines for global connections. One of the reasons for its widespread use is that it works on the ordinary copper wire already in place in the telephone system.

One advantage of *ISDN* over other digital communications technologies is its ability to handle all types of information such as voice, computer data, studio-quality sound, and video.

3.2 History

The early phone network consisted of a pure analog system that connected telephone users directly by a mechanical interconnection of wires. This system was very inefficient, was very prone to breakdown and noise, and did not lend itself easily to long-distance connections. Beginning in the 1960s, the telephone system gradually began converting its internal connections to a packet-based, digital switching system. Today, nearly all voice switching in the United States (U.S.) is digital within the telephone network. Still, the final connection from the local central office to the customer equipment was, and still largely is, an analog Plain-Old Telephone Service (POTS) line.

A standards movement was started by the International Telephone and Telegraph Consultative Committee (CCITT), now known as the International Telecommunications Union (ITU). The ITU is a United Nations organisation that coordinates and standardises international telecommunications. Original recommendations of *ISDN* were in *CCITT* Recommendation I.120 (1984) which described some initial guidelines for implementing *ISDN*.

Local phone networks, especially the regional Bell operating companies, have long hailed the system, but they had been criticised for being slow to implement *ISDN*. One good reason for the delay is the fact that the two major switch manufacturers, Northern Telecom (now known as Nortel Networks), and AT&T (whose switch business is now owned by Lucent Technologies), selected different ways to implement the *CCITT* standards. These standards didn't always inter-operate. This situation has been likened to that of earlier 19th century railroading- "people had different gauges, different tracks... nothing worked well".

In the early 1990s, an ‘industry-wide’ effort was initiated to establish a specific implementation for *ISDN* in the U.S. Members of the industry agreed to create the National ISDN 1 (**NI-1**) standard so that end users would not have to know the brand of switch they are connected to in order to buy equipment and software compatible with it. However, there were problems agreeing on this standard. In fact, many western states would not implement *NI-1*. Both Southwestern Bell and U.S. West (now Qwest) said that they did not plan to deploy *NI-1* software in their central office switches due to incompatibilities with their existing *ISDN* networks.

Ultimately, all the Regional Bell Operating Companies (RBOCs) did support *NI-1*. A more comprehensive standardisation initiative, National ISDN 2 (**NI-2**), was later adopted. Some manufacturers of *ISDN* communication equipment- such as Motorola and U. S. Robotics (now owned by 3Com), worked with the *RBOCs* to develop configuration standards for their equipment. These kinds of actions, along with more competitive pricing, inexpensive *ISDN* connection equipment, and the desire for people to have relatively low-cost high-bandwidth internet access have made *ISDN* more popular in recent years.

Most recently, *ISDN* service has largely been displaced by broadband internet service, such as *xDSL* and cable modem service. These services are faster, less expensive, and easier to set up and maintain than *ISDN*. Still, *ISDN* has its place, as backup to dedicated lines, and in locations where broadband service is not yet available

3.3 Components of ISDN

While individual operating companies and ministries will define the specific services, within the *ISDN* architecture the *ITU* standards define a number of component parts and functions as listed below.

- ISDN Channels
- Access Types
- Devices
- Interfaces
- Protocols

ISDN Channels

A CHANNEL is the basic unit of *ISDN* service. The *ISDN* Standards define three basic types of channels:

- Bearer channels (B channels)
- Delta (or "Demand") channels (D channels)
- High-capacity channels (H channels)

B Channel

A *B channel* is a 64-Kbps unit of clear digital bandwidth. Based on the data rate required to carry one digital voice conversation, a *B channel* can carry any type of digital information (voice, data, or video) with no restrictions on format or protocol imposed by the *ISDN* carrier.

D Channel

A *D channel* is a signaling channel. It carries the information needed to connect or disconnect calls and to negotiate special calling parameters (i.e., automatic number *ID*, call waiting, data protocol). The *D channel* can also carry packet-switched data using the *X.25 protocol*.
The *D channel* is not a clear channel. It operates according to a well-defined pair of layered protocols:

- Q.921 (LAPD) at the data link layer (Layer 2)
- Q.931 at the upper layers (Layers 3 and above)

The data rate of a *D channel* varies according to the type of access it serves-a basic rate access *D channel* operates at 16 Kbps and a primary rate access *D channel* operates at 64 Kbps.

Signaling on the *D Channel*

The *ISDN D channel* carries all signaling between the customer's terminal device and the carrier's end switching office.
Signaling information with end-to-end significance (i.e., which must be received by the terminal device at a call's destination, such as Automatic Calling Number Identification information) travels between the carrier's switching offices on the carrier's common-channel signaling network and on to the destination terminal, through the receiving user's *D channel*.

H Channel

An *H channel* is a special, high-speed clear channel. *H channels*, designed primarily for full-motion color video, are not yet in common use. There are currently three kinds of H channel:

- H0 ("H-zero")
- H11 ("H-one-one")
- H12 ("H-one-two")

An *H0* channel operates at 384 Kbps (roughly one fourth of a North American primary rate access or one fifth of a European primary rate access). An *H1* channel operates at 1.536 Mbps and occupies one whole North American primary rate access. An *H12 channel* occupies an entire European primary rate access.

3.3.1 ISDN Access Types

ISDN offers two general types of access:

- BASIC RATE ACCESS (BRA)
- PRIMARY RATE ACCESS (PRA)

These differ from one another by the amount of information they can carry.

Basic rate access

Basic rate access is based on new technology conceived especially for *ISDN*. Designed to provide service to individual users or small businesses, basic rate access provides two 64-Kbps *B channels* and one 16-Kbps *D channel* (referred to as 2B+D). In other words, it provides transmission facilities for one voice conversation (one *B* channel), one medium-speed data session (the other *B* channel), and the signaling exchanges needed to make them work (the *D* channel).

Two *B channels* at 64 Kbps plus one *D channel* at 16 Kbps equals 144K bps. The *ISDN* basic rate transmission protocol uses an additional 48 Kbps of bandwidth for maintenance and synchronisation, so an *ISDN* basic rate access actually uses 192 Kbps.

Primary rate access

Primary rate access, which is based on pre-*ISDN* digital carrier technology, is designed to provide high-capacity service to large customers for applications such as PBX-to-PBX *trunking*. There are two kinds of primary rate access: 23B+D and 30B+D. Each depends on the kind of digital carrier available in a given country.

In North America and Japan, 23B+D primary rate access operates at 1.544 Mbps and offers 23 *B channels* plus 1 64-Kbps *D channel* (usually located in time-slot 23), or 4 *H0* channels, or 1 *H11* channel. In most of the rest of the world, 30B+D primary rate access operates at 2.048 Mbps and offers 30 B channels plus 1 64-Kbps *D channel* (located in time-slot 16), or 5 *H0* channels, or 1 *H12 channel*.

3.4 *ISDN* Devices

In the context of *ISDN* standards, standard devices refer not to actual hardware, but to standard collections of functions that can usually be performed by individual hardware units. The *ISDN* standard devices are listed below.

- Terminal Equipment (TE)
- Terminal Adapter (TA)
- Network Termination 1 (NT1)
- Network Termination 2 (NT2)
- Exchange Termination (ET)

Terminal Equipment (TE)

A *TE* is any piece of communicating equipment that complies with the *ISDN* standards. Examples include digital telephones, *ISDN* data terminals, Group IV Fax machines, and *ISDN*-equipped computers.

In most cases, a *TE* should be able to provide a full basic rate access (2B+D), although some *TEs* may use only 1B+D or even only a *D channel*.

Terminal Adapter (TA)

A *TA* is a special interface-conversion device that allows communicating devices that don't conform to *ISDN* standards to communicate over the *ISDN*.

The most common *TAs* provide basic rate access and have one *RJ*-type modular jack for voice and one *RS-232* or *V.35* connector for data (with each portable to connect to either of the available *B* channels). Some *TAs* have a separate data connector for the *D channel*.

Network Termination (NT1 and NT2)

The *NT* devices, *NT1* and *NT2*, form the physical and logical boundary between the customer's premises and the carrier's network. *NT1* performs the logical interface functions of switching and local-device control (local signaling). *NT2* performs the physical interface conversion between the dissimilar customer and network sides of the interface.

In most cases, a single device, such as a *PBX* or digital multiplexer, performs both physical and logical interface functions. In *ISDN* terms, such a device is called *NT12* ("NT-one-two") or simply *NT*.

Exchange Termination (ET)

The *ET* forms the physical and logical boundary between the digital local loop and the carrier's switching office. It performs the same functions at the end office that the *NT* performs at the customer's premises.

In addition, the *ET*:

- separates *B channels*, placing them on the proper interoffice trunks to their ultimate destinations
- terminates the signaling path of the customer's *D channel*, converting any necessary end-to-end signaling from the *ISDN D-channel* signaling protocol to the carrier's switch-to- switch trunk signaling protocol

3.5 *ISDN* Interfaces (Standard Reference Points)

The *ISDN* standards specify four distinct interfaces in the customer's connection to the network: R, S, T, and U, as discussed in subsection 3.5.1 through 3.5.4

From the standards viewpoint, these are not "real" physical interfaces, but simply *'standard reference points'* where physical interfaces may be necessary. However, in common practice, the names of reference points are used to refer to physical interfaces.

3.5.1 The *R* Interface

The interface at reference point *R* is the physical and logical interface between a non-*ISDN* terminal device and a Terminal Adapter (TA). The *R* interface is not really part of the *ISDN*; it can conform to any of the common telephone or data interface standards.

3.5.2 The *S* Interface

The interface at reference point *S* is the physical and logical interface between a *TE* (or TA) and an *NT*. The *S* interface uses four wires and employs a bipolar transmission technique known as Alternate Mark Inversion (AMI).

A special feature of the S interface is the "short passive bus" configuration, which allows up to eight *ISDN* devices (TE or TA) to contend for packet access to the *D* channel in a prioritised, round-robin fashion. Only one device at a time can use a given *B* channel.

3.5.3 The *T* Interface

The interface at reference point *T* is the physical and logical interface between *NT1* and *NT2*, whenever the two *NTs* are implemented as separate pieces of hardware. The specification for the *T* interface is identical to the specification for the S interface.

In most implementations, *NT1* and *NT2* exist in the same physical device, so there is no real *T* interface.

3.5.4 The *U* Interface

The interface at reference point *U* is the physical and logical interface between *NT* (or NT2) and the *ISDN* carrier's local transmission loop. It is also the legal demarcation between the carrier's loop and the customer's premises.

The *U* interface is implemented with two wires and uses a special quaternary signal format (i.e., four possible electrical states, with one pulse encoding a predefined combination of 2 bits) called *2B1Q*. Quaternary encoding allows the *U* interface to carry data with a logical bit rate of 192 Kbps over a signal with a physical pulse rate of only 96 Kbps. The slower pulse rate is better suited to the less-predictable environment of the outside-plant loop carrier system.

3.6 How Does *ISDN* Work?

The simplest *ISDN* connection (called Basic Rate or BRI) consists of two 64 Kbps (kilobits-per-second) data channels (called *B-channels*) plus a 16 Kbps control channel (called the *D-channel*). This is sometimes referred to as “2B+D.” On the other end of the spectrum is primary rate *ISDN* (called PRI) with 23 B-channels plus a *D-channel*(i.e.: “23B+D”).

To connect to the *ISDN* line, you need a black-box called an *NT1* Network Terminator – a power supply (which you also need) is often built-in. You will also need a Terminal Adapter (often called a “TA”) to connect non-*ISDN* equipment (such as your computer or fax machine) to the line – these are also available as plug-in cards for *PC’s*. Some *TA’s* work as ethernet bridges so that you can connect your *LAN* directly to the *ISDN* line.

3.7 The Usefulness of *ISDN*

One of the most common uses for this technology today is videoconferencing. By using from one to four *BRI* lines, a videoconference can be established between two or more sites – the more lines, the faster the connection. For a videoconference application, higher connection speed translates to higher resolution and video frame rates. The telephone company infrastructure allows these connections to be made in a similar fashion to dialing a telephone. While video conferencing has been around for a long time, in the past it has primarily been confined to large corporations. The ability to transmit quality voice and video over long distances used to require expensive equipment and costly leased lines – these could only be justified by the largest of companies. Due to this dependency on leased lines, videoconferences were point-to-point (e.g. headquarters may be, permanently, linked to a manufacturing plant). Videoconferencing on the scale of teleconferencing was simply impractical.

The advent of new low-cost videoconference hardware that can utilise *ISDN* is rapidly changing this. Both desktop conferencing (a participant uses a *PC* equipped with a microphone, a small video camera, and an *ISDN* interface) and true videoconferencing (where more sophisticated equipment and remote control cameras allow group participation) have become as easy to set up as voice conferencing. Due to *ISDN's* versatility, videoconferences can include the sharing of graphic images and presentations, computer applications, documents, and computer files. This capability is proving popular for telecommuting, long distance meetings, workgroup collaboration, security and surveillance, and dozens of other innovative applications.

4.0 CONCLUSION

In this unit, you have been introduced to the fundamental concepts of *ISDN*. You have also learnt the different components and interfaces of *ISDN*.

5.0 SUMMARY

In this unit, you have learnt about:

- the term *ISDN*
- *ISDN* interfaces
- how *ISDN* works
- the usefulness of *ISDN*.

6.0 TUTOR-MARKED ASSIGNMENT

a. According to *ITU*, describe *ISDN* architecture.
b. Describe *ISDN* access types.

7.0 REFERENCES/FURTHER READING

Friedhelm, Hillebrand (2002). *GSM and UMTS: The Creation of Global Mobile Communications* (Ed.). John Wiley & Sons.

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UNIT 2 DIGITAL SUBSCRIBER LINE (DSL)

CONTENTS

1.0 INTRODUCTION

DSL is a specialised and problem-oriented language. Contrary to a General Purpose Language (GPL) (e.g. *UML*, *Java* or *C#*), a *DSL* serves to accurately describe a domain of knowledge. The interest to combine a *DSL* and a transformation function is to raise the abstraction level of software. A *DSL* user concentrates her/his efforts on domain description while complexity, design and implementation decisions and details are hidden. The stake is to improve productivity and software quality.

However, what is the next consensus beyond this general definition? An experience consists in starting the development of a *DSL* editor coupled to a generator. Quickly, the issue of the variants of *DSL* editors and generators emerges. Regarding the language, is it a tree-based *DSL* or a set of data without real structure? Is it a graphical or textual notation? Is it a declarative or imperative style? Answers to most of these questions will be provided in this unit

2.0 OBJECTIVES

At the end of this unit, you should be able to:

- explain the term *DSL*
- state the features of *DSL*
- illustrate how *DSL* works
- highlight the advantages and disadvantages of *DSL*

3.0 MAIN CONTENT

3.1 What is *DSL*?

DSL is a telephone loop technology that uses existing copper phones lines, and provides a dedicated, high speed internet connection. The big advantage of some *DSLs* (notably *ADSL*), is that they can co-exist on the same line with a traditional voice service such as "POTS" (Plain Old Telephone Service), and even *ISDN*. This is accomplished by utilising different frequency ranges above the voice range (voice is up to 4 KHz). Essentially, this gives two lines in one- one for voice, and one for internet connectivity. When all is working normally, there should be no interference between the two "lines". This gives *DSL* a potentially broad consumer base, and helps minimise costs for service providers.

DSL is positioned for the Home and Small Office (SOHO) market that is looking for high speed internet access at reasonable prices. Since it also typically provides dedicated, "always on" access, it can be used for interconnecting low to mid range bandwidth servers, and provides a great access solution for small *LANs*. It is also great for those Linux power users that just want a fat pipe).

Phone companies, and other independent telecommunications providers (CLECs), are now deploying *DSL* to stay ahead of the cable companies - the main consumer and SOHO competition for *DSL* providers. This mad rush to get "a piece of the pie" is bringing much competition (a good thing!), much diversity, and some confusion, into the consumer market. The *DSL* provider (often, but not always, the phone company) will provide the *DSL* infrastructure. This will include your line, the *DSLAM*, and physical connection to the outside world. From there, it is typically picked up by an *ISP*, who provides the traditional internet services.

Consumer *DSL* plans are typically "best effort" services. While boasting speeds approaching *T1,* and even surpassing that in some cases, it is not necessarily as reliable as *T1,* however. Business class *DSL* offers more reliability at a higher cost than consumer plans, and is a good compromise where both reliability and bandwidth are at a premium. All in all, the cost of *DSL* compared to traditional telecom services, such as *T1,* is attractive and substantially more affordable for home and small business users.

DSL providers often do not have service contracts for home users, while business class *DSL* services typically do include similar *SLAs* (Service Level Agreements) to that offered for a *T1* line.

The downside is that *DSL* is not available everywhere. Availability, and available bit rate (speed), are purely a function of where you live, where the telecom firm has installed the prerequisite hardware, how far you are from the *DSLAM/CO*, and the quality of your phone line (loop). Not all loops are created equal, unfortunately. The primary limitation is distance.

3.2 History of DSL

Implementation of digital subscriber line technology originally was part of the Integrated Services Digital Network (ISDN) specification published in 1984 by the *CCITT* and *ITU* as part of *recommendation I.120,* later reused as ISDN Digital Subscriber Line (IDSL). Engineers have developed higher-speed *DSL* facilities such as High (bit rate) Digital Subscriber Line (HDSL) and Symmetric Digital Subscriber Line (SDSL) to provide traditional Digital Signal 1 (DS1) services over standard copper pair facilities. Consumer-oriented Asymmetric Digital Subscriber Line (ADSL), first tested at Bellcore in 1988, was designed to operate on existing lines already conditioned for *BRI ISDN* services. This itself is a switched digital service (non-*IP*), though most Incumbent Local Exchange Carriers (ILECs) provide Rate-Adaptive Digital Subscriber Line (RADSL) to work on virtually any available copper pair facility—whether conditioned for *BRI* or not.

The development of *DSL*, like many other forms of communication, can be traced back to Claude Shannon's 1948 seminal paper- *'A Mathematical Theory of Communication''.* Employees at Bellcore (now Telcordia Technologies) developed *ADSL* in 1988 by placing wide-band digital signals above the existing (baseband]) analog voice signal carried between Telephone Company (central offices) and customers on conventional (twisted pair) cabling facilities. A *DSL* circuit provides "digital service". The underlying technology of transport across *DSL* facilities uses high-frequency (sinusoidal - carrier wave) modulation, which is an analog signal transmission.

A *DSL* circuit terminates at each end in a (modem) which modulates patterns of Binary digit (bits) into certain high-frequency impulses for transmission to the opposing modem. Signals received from the far-end modem are demodulated to yield a corresponding bit pattern that the modem retransmits, in digital form, to its interfaced equipment, such as a computer, router, switch, etc. Unlike traditional dial-up modems, which modulate bits into signals in the 300–3400&nbsp; *Hz* baseband (voice service), *DSL* modems modulate frequencies from 4000&nbsp; Hz to as high as 4&nbsp; *MHz*. This frequency band separation enables *DSL* service and plain old telephone service (POTS) to coexist on the same copper pair facility. Generally, higher bit rate transmissions require a wider frequency band, though the ratio of bit rate to bandwidth are not linear due to significant innovations in digital signal processing.

Early *DSL* service required a dedicated dry loop, but when the U.S. Federal Communications Commission (FCC) required *ILECs* to lease their lines to competing *DSL* service providers, shared-line *DSL* became available. Also known as *DSL* over (Unbundled Network Element), this unbundling of services allows a single subscriber to receive two separate services from two separate providers on one cable pair. The *DSL* service provider's equipment is co-located in the same central office as that of the *ILEC* supplying the customer's pre-existing voice service. The subscriber's circuit is then rewired to interface with hardware supplied by the *ILEC* which combines a *DSL* frequency and *POTS* frequency on a single copper pair facility.

On the subscriber's end of the circuit, inline low-pass *DSL* filters (splitters) are installed on each telephone to filter the high-frequency "hiss" that would otherwise be heard. Conversely, High-pass filters already incorporated in the circuitry of *DSL* modems filter out voice frequencies. Although *ADSL* and *RADSL* modulation do not use the voice-frequency band, nonlinear elements in the phone could otherwise generate audible inter-modulation and may impair the operation of the data modem in the absence of low-pass filters

3.3 Features of *DSL*

Let us look at these in relation to the following

- Language feature
- Transformation feature
- Tool feature
- Process feature

At the root level, language and transformation are mandatory features because they are parts of the *DSL* definition. Tool is also mandatory because it serves to automate transformation from a domain, the problem space, down to lower abstraction levels, the solution space. Process is optional because it can be undefined or implicit.

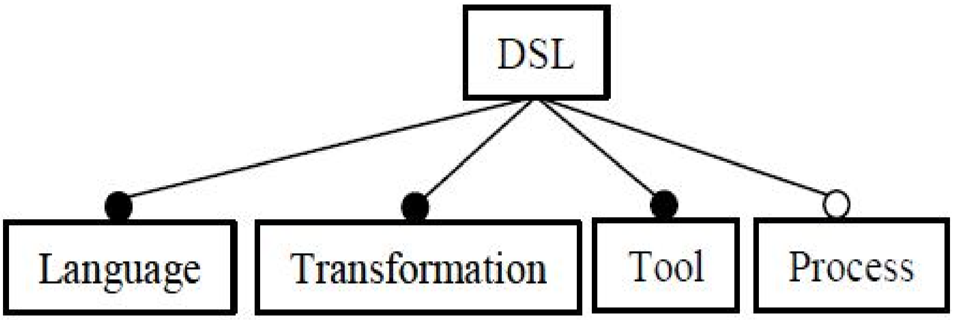

Figure 2.1: Root of the DSL Feature Model

3.4 Benefits of *DSL*

Fast *DSL* access allows you to download images, videos and other large files at lightning-fast speeds.

It is always on- there is no waiting to get connected, no busy signals, no dialing required with *DSL*.

No extra phone line necessary- *DSL* technology uses your existing phone line, allowing you to share phone and Internet on the same line — at the same time. You can also use other devices as usual on your telephone line. *DSL* won't interfere.

Fast internet connection- you will be able to download information, graphics and video from the internet at speeds up to 2.3 Mb/s; and, you can upload information as fast as 2.3 Mb/s.

Dedicated line- you will be the only person using your connection and line. Unlike cable or modems, you do you not share your connection with other users, so your Internet speed will be more private, stable and will have fewer delays.

Available in limited areas- *DSL* users must be within a certain radius of the area telephone switch.

In essence, *DSL* has the following advantages:

- the speed is much higher than a regular modem
- *DSL* does not necessarily require new wiring
- the company that offers *DSL* will usually provide the modem as part of the installation

Disadvantages of *DSL*

- The service is not available everywhere
- The connection is fast for receiving data than it is for sending data over the internet
- A *DSL* works better when you are closer to the provider’s central office

3.5 How *DSL* Works

DSL is an innovative technology that allows your computer to transmit information over existing phone line, but at a higher frequency than telephones, fax machines and other devices. Since it utilises the higher frequencies, you can use the same phone line for telephone calls and your internet connection at the same time, without any interference. Your telephone calls will still be clear and crisp, and your Internet connection will seem like lightning compared to a 56K modem connection.

4.0 CONCLUSION

A *DSL* is a problem-oriented language, which- combined to transformation tools, such as generators, serves to raise the abstraction level of software and ease software development; but beyond this general definition, *DSL* and *DSL* tool variants are numerous. The reason of a *DSL* feature model is to formalise *DSL* and *DSL* tool variants. A first application of this feature model is a *DSL* tool factory, which applies variations during production of *DSL* tools. A second application is the selection of pertinent *DSL* families among all possible families from the feature model. A third application is the definition of *DSL* tool foundations. A fourth usage is the selection of *DSL* tools. The feature model, combined with classification criteria, contains needed information to evaluate *DSL* tools. *DSL* feature model is in the scope of domain analysis of *DSLs*. Its clarification becomes a prerequisite for long-term and large scale *DSL* developments.

5.0 SUMMARY

In this unit, you have learnt that:

- the implementation of Digital Subscriber Line technology originally was part of the Integrated Services Digital Network (ISDN) specification published in 1984 by the *CCITT* and ITU as part of recommendation I.120, later reused as *ISDN* Digital Subscriber Line (IDSL)

- *DSL* is a telephone loop technology that uses existing copper phones lines, and provides a dedicated, high speed Internet connection
- *DSL* feature model consist of language, transformation, tool, process
- *DSL* is an innovative technology that allows your computer to transmit information over existing phone line, but at a higher frequency than telephones, fax machines and other devices
- A *DSL* is a problem-oriented language, which- combined with transformation tools, such as generators, serves to raise the abstraction level of software and ease software development

6.0 TUTOR-MARKED ASSIGNMENT

What are the advantages and disadvantages of *DSL?*

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UNIT 3 SYNCHRONOUS OPTICAL NETWORK (SONET)

CONTENTS

1.0 INTRODUCTION

In today's business world, each industry is looking for different ways to create competitive advantages to deliver information, products and services in a more timely and cost effective manner. End-to-end SONET (Synchronous Optical Networking) network solutions are one important ingredient in creating a competitive edge. As a convergence technology, *SONET* provides for the unification of voice, data and video over the same transport service.

This unit gives you an operational overview of *SONET* (Synchronous Optical Networking). Here, an important point to remember is this- *SONET* is a powerful, highly scalable technology. Although it may appear to be complex, most of what goes on in a *SONET* network is transparent to the user. Also, this unit briefly discusses Wave Division Multiplexing (WDM) for awareness purposes only, since *WDM* is another high performance transport technology that also leverages fiber optics.

SONET is a *transport* technology, designed to provide enterprise and government users –as well as service providers – a network infrastructure with survivability characteristics, so that business operations can continue uninterrupted. *SONET's* self-healing fiber optic ring functionality enables automatic network recovery due to failures that can be caused by a fiber optic cable cut, lost signal, or degraded signal (e.g. due to aging laser) or node/system failure. *SONET* is also a technology that is designed to ensure network traffic is restored within 60 milliseconds in the event of a failure.

2.0 OBJECTIVES

At the end of this unit, you should be able to:

- define the term *SONET*
- state the standards of *SONET*
- discuss the topology of *SONET*
- describe the *SONET* equipment layers
- explain *SONET* synchronisation and timing

3.0 MAIN CONTENT

3.1 Definition of *SONET*

Synchronous optical network (SONET) is a standard for optical telecommunications transport formulated by the Exchange Carriers Standards Association (ECSA) for the American National Standards Institute (ANSI), which sets industry standards in the U.S. for telecommunications and other industries. The comprehensive *SONET* standard is expected to provide the transport infrastructure for worldwide telecommunications for at least the next two or three decades.

3.2 Inception of *SONET*

SONET was conceived of and written about back in the early 1980's, when submitted to the members of American National Standard Institute (ANSI) *T1* Committee as a universal transport system. In the mid-1980's, the *T1* Committee further enhanced the standard to arrive at the Synchronous Transport Signal One (STS-1) as the base-signaling rate. Around this time, the ITU-T (International Telecommunication Union-Telecommunications standard) (formerly CCITT) adopted *SONET* as the basis of or its international standard referred to as SDH (Synchronous Digital Hierarchy) transport system, where the *STS-1* rate (51.84Mbps) was to be a factor of 3 in terms of the European base rate of 155.52Mbps.

3.3 *SONET* Standards

The base standard for *SONET* is the *T1.105-1991* American National Standard (ANSI) for Telecommunications- Digital Hierarchy-Optical Interface Rates and Formats Specification (SONET). SONET standards define rates and formats as well as optical interfaces. The following American National Standards Institute (ANSI) specifications provide the primary standards, which define SONET:

- *ANSI T1*.106-1998 specification for optical parameters
- *ANSI T1*.102-1993specification for electrical parameters
- *ANSI T1*.105-1991 specification for multiplexing methods to map existing Digital Signals (e.g. DS1) into *SONET* payload signals
- *ANSI T1*.105-1991 specification for criteria for optical line automatic
- Protection switching
- *ANSI T1*.105-1991 specification for overhead channels to support standard
- Operation, Administration and Provisioning (OAM&P)

SELF-ASSESSMENT EXERCISE 1

Describe the structure of SONET

3.4 *SONET* Topology

SONET technology enables a number of different network topologies to solve networking requirements, including survivability, cost, and bandwidth efficiencies. The following provides a description of 3 different *SONET* configurations, which are deployed in a variety of enterprise situations. *SONET* configurations include:

- point-to-point configuration
- *hubbed* configuration
- linear add/drop configuration
- ring configuration

3.4.1 Point-to-Point Configuration

SONET point-to-point configurations create a simple topology that terminates a *SONET* payload at each point of a fiber optic cable span. Point-to-point configurations are typically deployed in transport applications, which require a single *SONET* multiplexer in a single route. Point-to-point configurations can be enhanced to increase survivability by deploying a protection path (second fiber span) over a different path between two or more *SONET* multiplexers.

3.4.2 *Hubbed* Configuration

Hubbed configurations consolidate traffic from multiple sites onto a single optical channel, which then can be forwarded to another site. This topology is often used in applications where the user wants to consolidate traffic from multiple satellite sites to a single site such as corporate headquarters, before extending it, in some cases to a central office. This topology helps to reduce the number of hops as well as the equipment required to create a multisite topology.

3.4.3 Linear Add/Drop Configuration

In the asynchronous digital signal hierarchy environment, every time a digital signal is accessed the entire signal needs to be multiplexed/demultiplexed, costing time and money at each site along a given path. However, a linear add/drop configuration enables direct access to *VTS/STS* channels at each intermediate site along a fiber optic path. Therefore the linear add/drop configuration eliminates the need to process (multiplex/demultiplex) the entire optical signal for pass-through traffic.

3.4.4 Self-Healing Ring Configuration

In a self-healing ring configuration, a mechanism referred to as automatic protection switching is employed. There are two types of protection ring-the first is UPSR (Unidirectional Path Switched Ring), the other is BLSR (Bi-Directional Line Switched Ring). Each of these is discussed later. The self-healing ring configuration is the most commonly deployed *SONET* topology in mission critical government and enterprise backbones, due to its survivability characteristics. Automatic protection switching is a mechanism provided within the *SONET* specification that is designed to provide duplicate finer span paths. In this configuration, a backup fiber span (protection ring) is enabled when and if there is a failure within the fiber span currently carrying traffic on a *SONET* network. It should be noted that during normal operating conditions, both fiber spans are always active, and a *SONET* multiplexer selects which fiber span to receive traffic, based on an internal algorithm (e.g. based on which fiber module was installed in the multiplexer first). The *SONET* standard specifies that the protection ring should automatically become the fiber span (ring) the *SONET* multiplexer receives traffic from within 60 milliseconds (unnoticeable to the user) in the event of a failure on the other fiber span.

3.5 SONET Equipment Layers

SONET defines the end-to-end connection as being made up of 3 different equipment layers, including Path Terminating Equipment (PTE), Line Terminating Equipment (LTE), and Section Terminating Equipment (STE).

a. Path Terminating Equipment (PTE)

STS (Synchronous Transport Signal) path terminating equipment provides the multiplexing and demultiplexing functions within a *SONET* network. Path terminating equipment can originate access, modify, or terminate path overhead in any combination.

b. Line Terminating Equipment (LTE)

SONET line terminating equipment provides the function that originates and terminates line signals. *SONET* line terminating equipment can originate, access, modify, or terminate line overhead in any combination

c. Section Terminating Equipment (STE)

A *SONET* “section” is any two neighboring *SONET* network elements. *SONET* section terminating equipment can be a network element or a *SONET* regenerator. *SONET* section terminating equipment can originate, access, modify, or terminate section overhead in any combination

3.6 Importance of Synchronised Timing

Network timing between *SONET* devices is an integral part of maintaining accurate information transmitted over a *SONET* network. In the earlier days of networking, the method used in timing was Asynchronous. In Asynchronous timing each switch runs its own clock. In Synchronous timing, switches can use a single common clock to maintain timing. This single common clock is referred to as a Primary Reference Source (PRS) or Master Clock. Synchronous timing maintains better accuracy than Asynchronous timing because it uses a single common clocking scheme to maintain timing. This accurate timing often becomes important to government and enterprise applications, particularly when they are running time sensitive applications (e.g. video streaming applications). There are three methods typically used in obtaining synchronous timing in *SONET* multiplexers (e.g. Lucent *DDM*-2000 Multiplexer), they are:

- timing from an onboard internal oscillator
- timing from an incoming optical signal from a high-speed interface
- timing from an external source coming from a DS1 timing reference that can be stratum 3 or higher clocking

SELF-ASSESSMENT EXERCISE 2

What do you understand by synchronous timing?

4.0 CONCLUSION

In this unit, you have been introduced to the fundamental concepts of *SONET*. You also learnt the historical background of *SONET*, the various standards of *SONET*. You were also introduced to various topologies of *SONET*.

5.0 SUMMARY

In this unit, you have learnt that:

- SONET is a *transport* technology, designed to provide enterprise and government users as well as service providers
- SONET is a network infrastructure with survivability characteristics, so that business operations continue uninterrupted
- Synchronous optical network (SONET) is a standard for optical telecommunications transport formulated by the Exchange Carriers Standards Association (ECSA) for the American National Standards Institute (ANSI), which sets industry standards in the U.S. for telecommunications and other industries
- State the standards of SONET
- The base standard for SONET is the T1.105-1991 American National Standard (ANSI) for Telecommunications- Digital Hierarchy-Optical Interface Rates and Formats Specification (SONET)
- SONET consist of 3 different equipment layers, including Path Terminating Equipment (PTE), Line Terminating Equipment (LTE), and Section Terminating Equipment (STE).

6.0 TUTOR-MARKED ASSIGNMENT

Explain SONET topologies.

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UNIT 4 PACKET SWITCHING

CONTENTS

1.0 INTRODUCTION

Packet switching is the basis for the Internet Protocol (IP). In packet switching, information flows are broken into variable-size packets (or fixed-size cells as in the case of ATM). These packets are sent, one by one, to the nearest router, which will look up the destination address, and then forward them to the corresponding next hop. This process is repeated until the packet reaches its destination. The routing of the information is thus done locally. Routing decisions are independent of other decisions in the past and in other routers; however, they are based on network state and topology information that is exchanged among routers using BGP, IS-IS or OSPF. The network does not need to keep any state to operate, other than the routing tables. The forwarding mechanism is called store-and-forward because IP packets are completely received and stored in the router while being processed, and then transmitted. Additionally, packets may need to be buffered locally to resolve CONTENTSion for resources. If the system runs out of buffers, packets are dropped.

2.0 OBJECTIVES

At the end of this unit, you should be able to:

- define the term Packet Switching
- state the advantages and disadvantages of packet switching
- state various packet switching methods.

3.0 MAIN CONTENT

3.1 History

The concept of switching small blocks of data was first explored by Paul Baran in the early 1960s. Independently, Donald Davies at the National Physical Laboratory in the UK had developed the same ideas. Leonard Kleinrock conducted early research in queueing theory which would be important in packet switching, and published a book in the related field of digital message switching (without the packets) in 1961. The also later played a leading role in building and management of the world's first packet switched network, the ARPANET.

Baran developed the concept of message block switching during his research at the RAND Corporation for the US Air Force into survivable communications networks, first presented to the Air Force in the summer of 1961 as briefing B-265Baran's P-2626 paper described a general architecture for a large-scale, distributed, survivable communications network. The paper focuses on three key ideas: first, use of a decentralized network with multiple paths between any two points; and second, dividing complete user messages into what he called "message blocks" (later called packets); then third, delivery of these messages by store and forward switching.

Licklider at the Information Processing Technology Office, both wide-area network evangelists, and it helped influence Lawrence Roberts to adopt the technology when Taylor put him in charge of development of the ARPANET. Baran's work was similar to the research performed independently by Donald Davies at the National Physical Laboratory, UK. In 1965, Davies developed the concept of packet-switched networks and proposed development of a UK wide network. He gave a talk on the proposal in 1966, after which a person from the Ministry of Defense told him about Baran's work. A member of Davies' team met Lawrence Roberts at the 1967 Association for Computing Machinery (ACM) Symposium on Operating System Principles.

Interestingly, Davies had chosen some of the same parameters for his original network design as Baran, such as a packet size of 1024 bits. In 1966 Davies proposed that a network should be built at the laboratory to serve the needs of NPL and prove the feasibility of packet switching. The NPL Data Communications Network entered service in 1970. Roberts and the ARPANET team took the name "packet switching" itself from Davies's work.The first computer network and packet switching network deployed for computer resource sharing was the Octopus Network at the Lawrence Livermore National Laboratory that began connecting four CDC 6600 (Control Data 6600 computers) to several shared storage devices including an IBM 2321 Data Cell.

3.2 Packet Switching Definition

Packet switching is dividing of messages into *packets* before they are sent, transmitting each packet individually, and then reassembling them into the original message once all of them have arrived at the intended destination.

Packets are the fundamental unit of information transport in all modern computer networks, and increasingly in other communications networks as well. Each packet, which can be of fixed or variable size depending on the protocol, consists of a *header*, body (also called a *payload*) and a *trailer*. The body contains a segment of the message being transmitted.

3.3 Advantages of Packet Switching

The following are some advantages of packet switching:

- Share link usage and greater link efficiency
- Links can transmit at different rates
- Under heavy load, calls can still be accepted but with greater delay
- Can prioritize packet transmission

3.4 Disadvantages of Packet Switching

Despite its great advantages, packet switching has the following disadvantages:

- Packet delay at each node (Processing, Queuing, Transmission)
- A packet can acquire jitter while passing through the network
- Packets may contain metadata i.e Higher overhead than circuit Switching (e.g., telephone system)
- More processing required at each node in the network.

3.5 Packet Switching Methods

Datagram (Connectionless)

This method has the following characteristics:

- Each packet carries a header with full destination address.
- Each packet is treated independent of other packets.
- Each network node chooses the next hop for each packet.

Virtual Circuit Switching (Connection-Oriented)

This method has the following characteristics:

- Each packet header contains a virtual circuit identifier (VCI)
- Each node routes packets based on the VCI field
- A preplanned route is established before sending packets
- Faster routing

Source Routing: Full path stored in packet.

Exercise 1: What are the advantages and disadvantages of the packet switching methods?

4.0 CONCLUSION

In this unit, you have been introduced to the fundamental concepts of Packet switching. You also learnt the advantages and disadvantages of packet switching. You were also introduced to various packet switching methods.

5.0 SUMMARY

In this unit, you have learnt that:

- The concept of switching small blocks of data was first explored by Paul Baran in the early 1960s
- Packet switching is dividing of messages into *packets* before they are sent, transmitting each packet individually, and then reassembling them into the original message once all of them have arrived at the intended destination
- Packet switching methods datagram, virtual circuit and source routing.

6.0 TUTOR-MARKED ASSIGNMENT

1. State the advantages and the disadvantages of Packet Switching.
2. What do you understand by the term “Datagram”

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UNIT 5 INTERNET AND *TCP/IP*

CONTENTS

1.0 INTRODUCTION

The '''internet protocol suite''' is the set of communication protocols used for the internet and other similar networks. It is commonly also known as *'''TCP/IP'''*, named from two of the most important protocols in it- the Transmission Control Protocol (TCP) and the Internet Protocol (IP), which were the first two networking protocols defined in this standard. Modern *IP* networking represents a synthesis of several developments that began to evolve in the 1960s and 1970s, namely the internet and local area networks, which emerged during the 1980s, together with the advent of the World Wide Web(WWW) in the early 1990s. The internet protocol suite consists of four abstraction layers. From the lowest to the highest layer, these are the link layer, the internet layer, the transport layer, and the application layer.

The link layer contains communication technologies for the local network the host is connected to directly, the link. It provides the basic connectivity functions interacting with the networking hardware of the computer and the associated management of interface-to-interface messaging. The internet layer provides communication methods between multiple links of a computer and facilitates the interconnection of networks. As such, this layer establishes the internet. It contains primarily the internet protocol, which defines the fundamental addressing namespaces, *IPv4* Internet Protocol Version 4 (IPv4) and IPv6|Internet Protocol Version 6 (IPv6) used to identify and locate hosts on the network. Direct host-to-host communication tasks are handled in the transport layer, which provides a general framework to transmit data between hosts using protocols like the transmission control protocol and the user datagram.

2.0 OBJECTIVES

At the end of this unit, you should be able to:

- describe the internet standards process
- define common terms used in *TCP/IP*
- state the advantages of including *TCP/IP* components in windows
- explain how the *TCP/IP* protocol suite maps to the Department of Defense Advanced Research Projects Agency (DARPA) and Open System Interconnection (OSI) models.

3.0 MAIN CONTENT

3.1 Historical Background

The internet protocol suite resulted from research and development conducted by the Defense Advanced Research Projects Agency (DARPA) in the early 1970s. After initiating the pioneering *ARPANET* in 1969, *DARPA* started work on a number of other data transmission technologies. In 1972, Robert E. Kahn joined the *DARPA* information processing technology office, where he worked on both satellite packet networks and ground-based radio packet networks, and recognised the value of being able to communicate across both. In the spring of 1973, Vinton Cerf, the developer of the existing *ARPANET* Network Control Program (NCP) protocol, joined Kahn to work on open-architecture interconnection models with the goal of designing the next protocol generation for the *ARPANET*.

By the summer of 1973, Kahn and Cerf had worked out a fundamental reformulation, where the differences between network protocols were hidden by using a common [[internetwork protocol]], and, instead of the network being responsible for reliability, as in the *ARPANET*, the hosts became responsible.

The design of the network included the recognition that it should provide only the functions of efficiently transmitting and routing traffic between end nodes and that all other intelligence should be located at the edge of the network, in the end nodes. Using a simple design, it became possible to connect almost any network to the *ARPANET*, irrespective of their local characteristics, thereby solving Kahn's initial problem. One popular expression is that *TCP/IP*, the eventual product of Cerf and Kahn's work, will run over "two tin cans and a string". A computer, called a router, is provided with an interface to each network.

It forwards packets back and forth between them. Originally a router was called "gateway", but the term was changed to avoid confusion with other types of gateway (computer networking) computers.

The idea was worked out in more detailed form by Cerf's networking research group at Stanford within 1973–74, resulting in the first *TCP* specification. The internet protocol suite resulted from research and development conducted by the Defense Advanced Research Projects Agency (DARPA) in the early 1970s. After initiating the pioneering *ARPANET* in 1969, *DARPA* started work on a number of other data transmission technologies. In 1972, Robert E. Kahn joined the *DARPA* information processing technology office, where he worked on both satellite packet networks and ground-based radio packet networks, and recognized the value of being able to communicate across both. In the spring of 1973, Vinton Cerf, the developer of the existing *ARPANET* Network Control Program (NCP) protocol, joined Kahn to work on open-architecture interconnection models with the goal of designing the next protocol.

3.2 Internet Standard Process

TCP/IP is the protocol of the Internet, it has evolved based on fundamental standards that have been created and adopted over more than 30 years. The future of *TCP/IP* is closely associated with the advances and administration of the Internet as additional standards continue to be developed. Although, no one organisation owns the Internet or its technologies, several organizations oversee and manage these new standards, such as the Internet Society and the Internet Architecture Board.

The Internet Society (ISOC) was created in 1992 and is a global organisation responsible for the internetworking technologies and applications of the internet. Although the society’s principal purpose is to encourage the development and availability of the internet, it is also responsible for the further development of the standards and protocols that allow the Internet to function. The *ISOC* sponsors the Internet Architecture Board (IAB), a technical advisory group that sets internet standards, publishes *RFCs*, and oversees the Internet standards process. The *IAB* governs the following bodies:

- the Internet Assigned Number Authority (IANA) oversees and coordinates the assignment of protocol identifiers used on the Internet.
- the Internet Research Task Force (IRTF) coordinates all TCP/IP-related research projects.

- the Internet Engineering Task Force (IETF) solves technical problems and needs as they arise on the internet and develops internet standards and protocols. *IETF* working groups define standards known as *RFC.*

3.3 *TCP/IP* Terminology

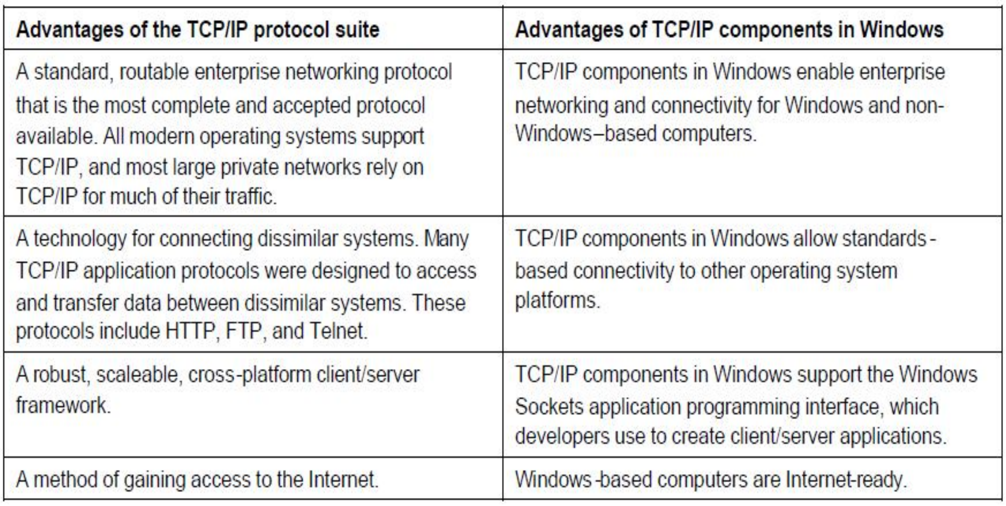
The internet standards use a specific set of terms when referring to network elements and concepts related to *TCP/IP* networking. Common terms and concepts in *TCP/IP* are defined as follows:

- **Node-** any device, including routers and hosts, which runs an implementation of *IP*.
- **Router-** a node that can forward *IP* packets not explicitly addressed to itself. On an *IPv6* network, a Router also typically advertises its presence and host configuration information.
- **Host-** a node that cannot forward *IP* packets not explicitly addressed to itself (a non-router). A host is typically the source and the destination of IP traffic. A host silently discards traffic that it receives but that is not explicitly addressed to it.
- **Upper-layer protocol:** a protocol above *IP* that uses *IP* as its transport. Examples include internet layer protocols such as the Internet Control Message Protocol (ICMP) and transport layer protocols such as the Transmission Control Protocol (TCP) and User Datagram Protocol (UDP). (However, application layer protocols that use *TCP* and *UDP* as their transports are not considered upper-layer protocols. File Transfer Protocol [FTP] and Domain Name System [DNS] fall into this category).
- **LAN segment-a** portion of a subnet consisting of a single medium that is bounded by bridges or layer 2 switches.
- **Subnet-** one or more *LAN* segments that are bounded by routers and use the same *IP* address prefix. Other terms for subnet are network segment and link.
- **Network-** two or more subnets connected by routers. Another term for network is internetwork.
- **Neighbour-** a node connected to the same subnet as another node.
- **Interface-t**he representation of a physical or logical attachment of a node to a subnet. An example of a physical interface is a network adapter. An example of a logical interface is a tunnel

interface that is used to send IPv6 packets across an IPv4 network.

- **Address-** an identifier that can be used as the source or destination of *IP* packets and that is assigned at the Internet layer to an interface or set of interfaces.
- **Packet-**The protocol Data Unit (PDU) that exists at the internet layer and comprises an *IP* head payload. a common [[internetwork protocol]], and, instead of the network being responsible for reliability, as in the *ARPANET*, the hosts became responsible.

Table 4.1: The Advantages of the TCP/IP Protocol Suite and the Inclusion of TCP/IP Components in Windows.



Advantages of the TCP/IP protocol suite	Advantages of TCP/IP components in Windows
A standard, routable enterprise networking protocol that is the most complete and accepted protocol available. All modern operating systems support TCP/IP, and most large private networks rely on TCP/IP for much of their traffic.	TCP/IP components in Windows enable enterprise networking and connectivity for Windows and non-Windows–based computers.
A technology for connecting dissimilar systems. Many TCP/IP application protocols were designed to access and transfer data between dissimilar systems. These protocols include HTTP, FTP, and Telnet.	TCP/IP components in Windows allow standards-based connectivity to other operating system platforms.
A robust, scaleable, cross-platform client/server framework.	TCP/IP components in Windows support the Windows Sockets application programming interface, which developers use to create client/server applications.
A method of gaining access to the Internet.	Windows-based computers are Internet-ready.

3.4 The *TCP/IP* Protocol Suite

The *TCP/IP* protocol suite maps to a four-layer conceptual model known as the *DARPA* model, which was named after the U.S. government agency that initially developed *TCP/IP*. The four layers of the *DARPA* model are: Application, Transport, Internet, and Network Interface. Each layer in the *DARPA* model corresponds to one or more layers of the seven-layer *OSI* model.

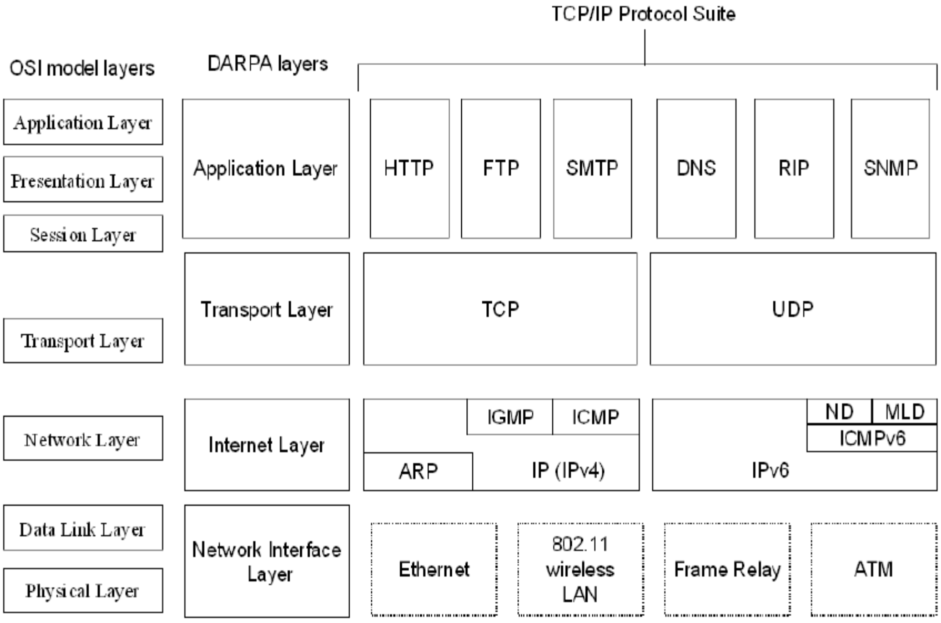


Figure 4.1: The Architecture of the TCP/IP model

The *TCP/IP* protocol suite has two sets of protocols at the internet layer:

a. *IPv4*, also known as *IP*, is the Internet layer in common use today on private intranets and the Internet
b. *IPv6* is the new Internet layer that will eventually replace the existing *IPv4* Internet layer.

SELF-ASSESSMENT EXERCISE 1

What are the specific problems with *IPv4*?

3.5 Network Interface Layer

The network interface layer (also called the network access layer) sends *TCP/IP* packets on the network medium and receives *TCP/IP* packets off the network medium. *TCP/IP* was designed to be independent of the network access method, frame format, and medium. Therefore, you can use *TCP/IP* to communicate across differing network types that use *LAN* technologies—such as ethernet and 802.11 wireless *LAN*—and *WAN* technologies—such as Frame Relay and Asynchronous Transfer Mode (ATM). By being independent of any specific network technology, *TCP/IP* can be adapted to new technologies. The network interface layer of the *DARPA* model encompasses the data link and physical layers of the *OSI* model. The internet layer of the *DARPA* model does not take advantage of sequencing and acknowledgment services that might be present in the data link layer of the *OSI* model. The internet layer assumes an unreliable network interface layer and that reliable communications through session establishment and the sequencing and acknowledgment of packets is the responsibility of either the transport layer or the application layer.

3.6 Internet Layer

The internet layer responsibilities include addressing, packaging, and routing functions. The internet layer is analogous to the network layer of the *OSI* model. The core protocols for the *IPv4* internet layer consist of the following:

i. the Address Resolution Protocol (ARP) resolves the internet layer address to a network interface layer address such as a hardware address.
ii. the Internet Protocol (IP) is a routable protocol that addresses, routes, fragments, and reassembles packets.
iii. the Internet Control Message Protocol (ICMP) reports errors and other information to help you diagnose unsuccessful packet delivery.

SELF-ASSESSMENT EXERCISE 2

What is the difference between internet layer and network layer of the *OSI* model?

3.7 Application Layer Interface

The application layer allows applications to access the services of the other layers, and it defines the protocols that applications use to exchange data. The application layer contains many protocols, and more are always being developed.

The most widely known application layer protocols help users exchange information. Some of these are:

i. the Hypertext Transfer Protocol (HTTP) transfers files that make up pages on the World Wide Web(WWW).
ii. the File Transfer Protocol (FTP) transfers individual files, typically for an interactive user session
iii. the Simple Mail Transfer Protocol (SMTP) transfers mail messages and attachments.

Additionally, the following application layer protocols help you use and manage *TCP/IP* networks:

i. the Domain Name System (DNS) protocol resolves a host name, such as ‘www.microsoft.com’, to an *IP* address and copies name information between DNS servers.

ii. the Routing Information Protocol (RIP) is a protocol that routers use to exchange routing information on an *IP* network., called a router, is provided with an interface to each network.

4.0 CONCLUSION

In this unit, you have been introduced to the fundamental concepts of internet *TCP/IP*. You also learnt the internet standard process and some *TCP/IP* terminologies. You were also introduced to Various *TCP/IP* protocol suites.

5.0 SUMMARY

In this unit, you have learnt that:

- the "internet protocol suite" is the set of communications protocols used for the internet and other similar networks
- the *TCP/IP* is the protocol of the internet; it has evolved based on fundamental standards that have been created and adopted over more than 30 years
- the *TCP/IP* protocol suite maps to a four-layer conceptual model known as the *DARPA* model, which was named after the U.S. government agency that initially developed *TCP/IP*
- the *TCP/IP* protocol suite has two sets of protocols at the internet layer- namely *IPv4 and IPv6*
- the network interface layer (also called the network access layer) sends *TCP/IP* packets on the network medium and receives *TCP/IP* packets off the network medium
- the internet layer responsibilities include addressing, packaging, and routing functions
- the application layer allows applications to access the services of the other layers, and it defines the protocols that applications use to exchange data.

6.0 TUTOR-MARKED ASSIGNMENT

1. Discuss the history of the *TCP/IP* model.
2. What is the function of the internet and the application layer?

7.0 REFERENCES/FURTHER READING

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