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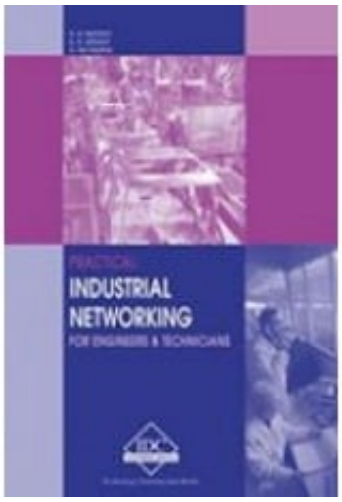
IN-E - Practical Industrial Networking for Engineers and Technicians



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Short Description

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Table of Contents

Download Chapter List

First Chapter

Introduction to Ethernet

1

Introduction to Ethernet

Objectives

In this first chapter, you will:

- Learn about the history of Ethernet
- Learn about relevant standards and standards institutions
- Have an overview of networking basics
- Become familiar with the concept of the OSI reference model
- Learn about the various IEEE and TCP/IP protocols and their position within the OSI model
- Learn about network topologies
- Learn the basics of network communications

1.1 Introduction

IDC Technologies welcomes you to this workshop on industrial networking. The workshop is aimed at practicing technicians, engineers, and others in commerce and industry that have been associated with the design, or maintenance, or administration of computer networks.

The broad objective of this workshop is to make you current and confident in the basics as well as in the practical knowledge of computer networking in general and Ethernet (including its industrial implementation) in particular. Associated topics, which do not strictly fall under Ethernet, but can be beneficial to you in your efforts to improve your knowledge in this area, have also been covered.

This first chapter starts with a brief overview of some networking history, standards, standards organizations, and semantics related to ‘Ethernet’. The remaining sections of this chapter delve into some basics, and give you a conceptual framework for computer networks.

1.1.1 The birth of Ethernet

University of Hawaii researcher Norman Abramson was one of the first to experiment with computer networking. In the late sixties, he experimented with what was called the Aloha network using a radio network for communication in the Hawaiian Islands, sharing a radio channel for communication.

The Aloha protocol allowed a station to transmit whenever it liked. The transmitting station then waited for an acknowledgment; and if none was received, a collision of the message with another sent by some other station was assumed. The transmission station then backed off for a random time and retransmitted again.

This kind of system resulted in approximately 18% utilization of the channel. Abramson next introduced assigned transmission slots synchronized by a master clock. This improved utilization to 37%.

In the early seventies Bob Metcalfe (Xerox) improved on the Aloha system by using a mechanism that listened for channel activity before transmitting, and detected collisions. He used this new system to connect Alto computers. This was called the Alto Aloha network.

In 1973 the name was changed to Ethernet to indicate that any computer could be supported, not just the Altos. Just as the ‘ether’ (i.e. space) spreads communication to all who listen, his system carried the message to all stations, hence the name ‘Ethernet’. Ethernet became a registered trademark of Xerox.

For a technology to become universally usable, its rules cannot be controlled by any single commercial enterprise. The technology, in this case LAN technology, has to be usable across a wide variety of equipment, and therefore has to be vendor-neutral.

Realizing that Ethernet had immense potential for worldwide use, Metcalfe persuaded Xerox in 1979 to join with Digital and Intel to promote a consortium for standardizing an open system. The joint efforts of Xerox, Digital and Intel, led by Metcalfe, refined the original technology into what is known as DIX Ethernet, Bluebook Ethernet or Ethernet II. Xerox agreed to let anybody use its patented technology for a small fee. Xerox even gave up its right to the trademark on the Ethernet name. Metcalfe started a company ‘3COM’ in 1979, to promote computer communications compatibility.

Thus, the Ethernet standard became the first vendor-neutral open LAN standard. The idea of sharing open source computer expertise for the benefit of everyone was a very radical notion at that time.

The success of DIX Ethernet proved that the open source environment worked. Requirements began to emerge for an even more open environment to attain more and more capabilities, cross-vendor usability, and cost reductions. Cross vendor usability required different platforms to recognize each other for

communications and sharing of data.

1.1.2 Developments through the Eighties and Nineties

The original Ethernet II used thick coaxial cable as a transmission medium, and although the use of Ethernet was spreading fast, there were problems in installing, connecting and troubleshooting thick coaxial cables. On a bus topology a cable fault will pull down the whole network.

The twisted pair cable medium was developed in the late eighties, and enabled the use of a star topology. Such systems were now much easier and quicker to install and troubleshoot. This development really gave a boost to the Ethernet market. The structured cabling standard (using twisted pairs) developed in early nineties made it possible to provide highly reliable LANs.

The original Ethernet networks were operating at 10 Mbps. This was fast at the time; in fact, Ethernet at 10 Mbps was faster than the computers connected to it. However, as computer speeds doubled every two years, traffic loads became heavy for 10 Mbps networks. This spurred development of higher speeds and the standard for 100 Mbps was adopted in 1995. This was now based on twisted pair and fiber optic media systems. The new interfaces for such systems use an Auto-Negotiation protocol to automatically set speeds in consonance with the hubs to which the workstations are attached. As computer chips raced ahead in terms of clock speeds, developments in Ethernet kept pace and the 1000 Mbps (1Gbps) standard was born around 1998.

While bars were periodically raised and jumped over as far as speed is concerned, there have been developments in other relevant fields as well. Original Half-Duplex Ethernet could either transmit or receive data at a given time, but could not do both simultaneously. The Full-Duplex standard now makes this possible – resulting in an effective bandwidth of 200 Mbps in 100 Mbps network.

The latest development, 10 Gbps Ethernet, was released at the beginning of 2002.

1.1.3 Semantics – which Ethernet are we talking about?

The term Ethernet originally referred to the original LAN implementation standardized by Xerox, Digital, and Intel. When the IEEE introduced Standard IEEE 802, its 802.3 group standardized operation of a CSMA/CD network that was functionally equivalent to (DIX) Ethernet II. There are distinct differences

between the Ethernet II and IEEE 802.3 standards. However, both are referred to as “Ethernet”. The only real legacy of Ethernet II that has to be dealt with today is the difference in the frame composition, specifically the “Type/Length” field. Although it is normally dealt with by the software, it could nevertheless confuse the novice.

Ethernet originally expanded and prospered in office environments. However, Ethernet networks in the industrial environment also gained importance. In these types of networks some of the shared and intercommunicating devices are not conventional computers, but rather industrial controllers and sensors for the measurement of process parameters. These industrial Ethernet networks have their own peculiarities, and will be dealt with under the umbrella of “Industrial Ethernet”.

Data transmission speeds of 100 Mbps (the IEEE802.3u standard, a.k.a. Fast Ethernet) and 1000 Mbps (the IEEE802.3z standard, a.k.a. Gigabit Ethernet) have been achieved. These faster versions are also included in the term ‘Ethernet’ and shall be dealt with later in this manual.

1.2 Standards and Standards Institutions

1.2.1 Standards institutions

ANSI, or the American National Standards Institute, is a nonprofit private organization with the objectives of development, coordination, and publication of voluntary national standards. Although ANSI standards are voluntary, and since ANSI participates in global bodies such as the ISO, IEC etc., noncompliance with ANSI standards becomes noncompliance with world standards.

The **IEEE**, or the Institute of Electrical and Electronic Engineers, develops standards for acceptance by ANSI, and ANSI in turn participates in the global standards bodies. The IEEE has been responsible for telecommunication and data communication standards for LANs, most relevant of which for our purposes are the IEEE 802 series of standards.

The **ISO**, or International Organization for Standardization, located in Geneva, is a UN organization chartered to define standards in virtually all subjects except electrical or electronic subjects. The acronym for this organization is derived from its French name. The OSI model has been developed by the ISO, and lays down a common organizational scheme for network standardization.

The **IAB**, or Internet Architecture Board, oversees technical development of the

Internet. The IETF and IRTF, two task forces set up by IAB, were responsible for standards such as the Internet Protocol (IP).

The **IEC**, or International Electro-technical Commission, located in Geneva sets international standards on electric and electronic subjects.

1.2.2 Compliance with standards

Not all Ethernet equipment necessarily conforms to the official IEEE standard. For example, twisted-pair media systems were vendor innovations (*de facto* standards), which later became specified media systems (*de jure* standards) in the IEEE standard. However, if you are a network manager responsible for maximum stability and predictability given different vendor equipment and traffic loads, then preference should be given to compliant and standard specified equipment.

Having said this, users often have to settle for *de facto* standards. This is often the case with equipment such as routers, switches and Industrial Ethernet where the official standard is lacking and vendors have to innovate out of necessity. This is particularly true for the leading innovators. A good example is the RJ-45 connector, which is unsuitable for Industrial applications but where a suitable alternative is not (yet) included in the standard. In this case vendors often use DB or M12 connectors.

1.3 Networking basics – an overview

An overview of the basics of networking in general will now be given so that participants of this workshop get [1] a broad perspective of the basic concepts underlying networking technologies, and [2] a sense of the interrelationships between the various aspects of networking technologies.

1.3.1 Network definition

A ‘network’ is a system comprising software, hardware, and rules that enable computers, computer peripherals, and other electronic devices to communicate with each other so as to share information and resources, collaborate on a task, and even communicate directly through individually addressed messages.

1.3.2 Classifications of computer networks

There is no generally accepted classification of computer networks, but transmission technology and physical/geographical areas of coverage are two

ways to classify them.

1.3.3 Classification based on transmission technology

There are two broad types of transmission technology. They are ‘broadcast’ and ‘point-to point’.

A broadcast network has a single communication channel, shared by all devices on that network. A sent message containing an address field specifying the device for which it is intended is received by all devices, which check the address field and discard the message if it is not specifically intended for them.

If the address in the address field matches with the device address, the message is processed. LAN systems using coaxial cable as medium are examples of a broadcast networks.

A point-to-point network involves dedicated transmission paths between individual pairs of devices. A message or a packet on this network may have to go to its destination via one or more intermediate devices. Multiple routes to the destination are possible, and routing algorithms often play an important role. IBM Token Ring and FDDI are examples of networks using point-to-point technology.

1.3.4 Classification based on geographical area covered

Networks can be classified based on the geographical area that they cover. Physical locations and distances are important because different methods are used for different distances.

Local area networks, LANs, are located in a single building or on a campus of a few kilometers in radius. LANs differ from other networks in their size, transmission technology, and their topology. The limits of worst-case transmission time are known in advance, as the LAN is restricted in size. The transmission technique is primarily based on broadcast methods and topologies can be bus, star or ring. In these configurations at any given time, only one machine is allowed to broadcast. An arbitration mechanism, either centrally located or distributed, resolves conflicts when two or more stations want to transmit at the same time.

Metropolitan area networks, MANs, cover a greater area than LANs and can extend to a whole city or, say, a circle with a radius of 50 km. A MAN has

typically two fiber optic rings or unidirectional buses to which all computers are connected. The direction of transmission on the two buses could be opposite to each other. The MAN does not consist of switching elements, and can support both data and voice.

Wide area networks or WANs incorporate LANs that are great distances apart; distances ranging from a few kilometers to thousands of kilometers. WANs normally use public telecommunication systems to provide cost-effective communication between LANs. Since such communication links are provided by independent third party utilities, they are referred as a communications cloud. Special equipment called routers store messages at LAN speed and transmit them across the communication cloud at the speed of the communication link to the LAN at the other end. The remote LAN then takes over and passes on the message at LAN speed once more. For mission critical and time critical applications, WANs are considered less reliable due to the delays in transmission through the communication cloud.

Virtual Private Networks or VPNs are WANs that use the Internet infrastructure to connect two or more LANs. Since Internet traffic is visible to all other users, encryption technology has to be used to maintain privacy of communication.

1.4 Interoperability and internetworking

Interoperability refers to the ability of users of a network to transfer information between different communication systems; irrespective of the way those systems are supported. It has also been defined as the capability of using similar devices from different manufacturers as effective replacements for each other without losing functionality or sacrificing the degree of integration with the host system.

In other words, it is the capability of software and hardware systems on different devices to communicate with each other, the user thus being able to choose the right devices for an application, independent of supplier, control system and protocol.

Internetworking is a term that is used to describe the interconnection of differing networks so that they retain their own status as a network. What is important in this concept is that internetworking devices be made available so that the exclusivity of each of the linked networks is retained, but that the ability to share information and physical resources, if necessary, becomes both seamless and transparent to the end user.

1.5 Network architecture and protocols

A protocol is defined as a set of rules for exchanging data in a manner that is understandable to both the transmitter and the receiver. There must be a formal and agreed set of rules if the communication is to be successful. The rules generally relate to such responsibilities as error detection and correction methods, flow control methods, and voltage and current standards. In addition, other properties such as the size of data packets are important in LAN protocols.

Another important responsibility of protocols is the method of routing the packet, once it has been assembled. In a self-contained LAN, i.e., intranetwork, this is not a problem since all packets will eventually reach their destinations. However, if the packet is to be routed across networks, i.e., on an internetwork (such as a WAN) then a routing decision must be made.

Network protocols are conceptually organized as a series of levels, or layers, one above each other. The names, number of layers and function(s) of each layer can vary from one type of network type to another. In any type of network, however, the purpose of any layer is to provide certain services to higher layers, hiding from these higher layers the details of how these services are implemented.

Layer ‘n’ of a computer communicates with layer ‘n’ of another computer, the communication being carried out as per the rules of the layer ‘n’ protocol.

The communication carried out between the two ‘n’ layers takes place on a logical plane. Physically, data is not directly transferred between these layers. Instead, each layer passes data and control information to the layer immediately below it, until the lowest layer is reached. The data is then transmitted across the medium to the receiving side, where it is passed upwards layer by layer until it reaches layer ‘n’.

Between each pair of layers is an interface that specifies which services the lower layer offers to the upper layer. A clean and unambiguous interface simplifies replacement, substituting implementation of one layer with a completely different implementation (if the need arises), because all that is required of the new implementation is that it offers exactly the same services to its upper layer as was done in the previous case.

A network architecture is nothing but a set of layers and protocols. The specifications of the architecture must contain sufficient information to allow implementers to write code or specify hardware for each layer so that it will correctly obey the appropriate protocol. Details of implementation and specification of interfaces are, however, not part of the network architecture. It is

not even necessary that interfaces on all machines be the same, as long as each machine correctly uses and obeys all protocols.

A group of protocols, one per layer, is called a protocol stack.

1.6 Layer design parameters

There are some parameters or design aspects that are common to the design of several layers. Some of these parameters are as follows:

- In a network having many computers, some of which may be carrying out multiple processes, a method is needed for a process on one machine to identify the process on another machine with whom it wishes to communicate. A **method of addressing**is therefore needed.
- **Direction of data transfer**is another design aspect that needs to be addressed. Data may travel in one direction only (as in simplex transmission), or it may travel in both directions but not simultaneously (as in half-duplex transmission), or in both directions simultaneously (as in full duplex transmission). The number of logical channels and their prioritizing also needs to be determined
- **Error control**is the third design aspect that is important. This involves the methods of error detection, error correction, informing the sender about correct or incorrect receipt of the message, and re-transmitting of the message if one has been ‘lost’
- **Sequencing**parts of a message in correct order while sending and their correct reassembly on receipt is another design issue. The matching of a fast sender of data and a slow receiver of data by methods such as agreed upon transmission rate also needs to be taken into account
- **Routing decisions** in the case of multiple paths between source and destination is another aspect of layer design. Sometimes this decision is split over two or three layers

1.7 Entities, SAPs, IDUs/ SDUs

Entities are active elements in each layer. They can be software entities (such as processes) or hardware entities (such as an intelligent I/O chip). Entities on the same layer on different machines are called peer entities. Entities in layer ‘n’ implement services used by layer n+1. Here layer n is the service provider and layer n+1 is the service user. Services are rendered and received at **SAPs** (Service access points) that have unique addresses. When two layers

exchange information, layer n+1 entity passes on an **IDU** (Interface Data Unit) to the entity in layer n. An IDU consists of an **SDU** (Service Data Unit) and control information. It is this SDU that is passed on to the peer entity of the destination machine.

1.8 Connectionless and connection-oriented service

These are types of services offered to a given layer by the layer below it. A connection-oriented service is like a telephone system that involves establishing a connection, using the connection, and releasing the connection. A connectionless service, on the other hand is like a postal service where messages are ‘sent and forgotten’.

These services are classified according to quality or reliability. A reliable service involves getting an acknowledgment from the receiver, which may be necessary in some cases, but will cause more traffic and delay, for example as in the case of a registered post parcel. An unreliable service is not necessarily “bad”, but simply one in which acknowledgements are not obtained.

1.9 Reference Models

Having discussed some universally applicable basics and terminology, two important reference models of network architectures, namely the open system interconnection (OSI) model and TCP/IP model will be dealt with.

1.9.1 Open systems interconnection (OSI) model

Computer networks are complex systems made of different hardware and software, each of which performs different functions. These functions have to be performed in conformity with certain rules so that functional conflicts are avoided. The rules have to be such that the interfacing interconnections between all these hardware and software components are open, transparent, and vendor neutral.

A model or framework is necessary to understand how various hardware and software parts mesh with each other. The open systems interconnection (OSI) reference model was developed for this purpose. The International Organization for Standardization (OSI) developed this model in 1977–1978 and it predates efforts by the IEEE on the Ethernet standard.

The OSI is not a physical model. It is a purely intangible and conceptual framework to explain how the complex interactions take place among the various subsystems of a network.

The OSI model defines rules governing the following issues:

- Methods by which network components contact and communicate with each other
- Methods by which a network component knows when to transmit a message
- Methods to ensure correct receipt of a message by a specific recipient
- Logical and physical arrangement of the physical transmission media and its connections
- Ensuring a proper flow of data
- Arrangement of bits that make up the data

The OSI model defines seven conceptual layers arranged hierarchically, one above the other (see Figure 1.1). The functional jurisdiction of each layer is made separate and distinct from that of any other layer. Each layer performs a task or some sub-task. The rule-based computer processes that perform these tasks are called ‘protocols’.

Since the layers are arranged hierarchically and vertically above each other, the protocols corresponding to each of the layers also are arranged in similar hierarchical stacks. When two computers are connected to a network to talk to each other intelligently, each of the computers must have the same stack of protocols running on it, even if the computers are dissimilar to each other in other aspects like operating system, hardware configuration etc.

When a message is sent from an application (e.g. a client) on one computer to an application (e.g. a server) on another computer, the client passes the message down the stack of protocols from top to bottom on the first computer, then the message passes across the medium between the two computers, and finally it proceeds up the stack on the receiving computer, where it is delivered to the server.

Figure 1.1

OSI layers

A message is composed in the sending application and then passed down to the application layer, from where it travels down to the bottom layer of the stack. As it

travels down, each intermediate layer adds its own header (consisting of relevant control information) until it reaches the lowest layer. From here, the message with headers travels to the destination machine. As it proceeds up the stack of the receiving machine, each successive layer strips off the header of its corresponding peer layer on the sending computer until it reaches the application layer. From there it is delivered to the receiving application. This is schematically shown in Figure 1.2. Please note that application itself does NOT reside at the application layer, but above it.

Figure 1.2

Adding and peeling of headers

As the sending application, e.g. the client (‘process A’), communicates with the receiving application, e.g. the server (‘process B’), the actual information flows in a ‘U’ fashion down the stack on the one side, across the medium, and then up the stack on the other side. However, each protocol in the stack is ‘horizontally’ (logically) communicating with its peer in the opposite stack by means of header information. This is illustrated in Figure 1.3.

Figure 1.3

Peer layers talking to each other

1.9.2 Functions of the OSI model layers

Application layer

This is the uppermost layer and the end user really ‘sees’ the output of this layer only, although all other layers are working in the background to bring the communication to its final form. This layer provides services directly for user applications such as clients and servers for file transfers, web services, e-mail, etc. It allows applications on one machine to talk to applications on another machine. The application layer services are more varied than those in other layers are because the entire range of application possibilities is available here.

Presentation layer

This layer is responsible for presenting information in a manner suitable for the application or the user dealing with the information. It translates data between the formats the network requires and the format the user expects. Its services include protocol conversion, data translation, encryption, compression, character set conversion, interpretation of graphic commands etc. In practice, this layer rarely appears in a pure form, and is not very well defined amongst the OSI layers. Some of its functions are encroached upon by the application and session layers.

Session layer

This layer provides services such as synchronization and sequencing of the packets in a network connection, maintaining the session until transmission is complete, and inserting checkpoints so that in the event of a network failure, only the data sent after the point of failure need be re-sent.

Transport layer

The transport layer is responsible for providing data transfer at an agreed-upon level of quality such as transmission rates and error rates. To ensure delivery, outgoing packets could be assigned numbers in sequence. The numbers would be included in packets that are transmitted by the lower layers. The transport layer at the receiving end would then check the packet numbers to ensure that all packets have been delivered and put the packets in the correct sequence for the recipient. Thus, this layer ensures that packets are delivered error free, in sequence, and with no losses or duplications. It breaks large messages from the session layer into packets to be sent to the destination and reassembles these packets to be presented to the session layer.

This layer also typically sends acknowledgments to the originator for messages received.

This layer uses the network layer below to establish a route between the source and destination. The transport layer is crucial in many ways, because it sits between the heavily application-dependent upper layers and the application-independent lower layers. It provides an end-to-end check of the message integrity, above the level of the routing and packet handling.

Network layer

Layers below the transport layer are called subnet layers, and the network layer

is the first among these.

The network layer decides on routes and sends packets to destinations that are farther than a single link. (Two devices connected by a link communicate directly with each other and not through a switching device). This layer determines addresses, translating logical ones into machine addresses, and decides on priorities.

Routing decisions are implemented here. This layer may choose a certain route to avoid other routes that are experiencing heavy traffic.

Routers and gateways operate in the network layer. Circuit, message and packet switching, network layer flow control, network layer error control, and packet sequence control are functions of the network layer.

Data link layer

The data link layer is responsible for creating, transmitting, and receiving data packets. It provides services for the various protocols at the network layer, and uses the physical layer to transmit or receive messages. It creates packets appropriate for the network architecture being used. Requests and data from the network layer are part of data in these packets (or, frames as they are called at this layer). Network architectures such as Ethernet, ARCnet, Token Ring, and FDDI encompass the data link and physical layers, which is why these architectures are said to support services at the data link level.

This layer provides for error-free transfer of frames from one computer to another. A cyclic redundancy check (CRC) added to the data frame can help in identifying damaged frames, and the corresponding data link layer in the destination computer care request that the information be resent. Detection of lost frames is also carried out and requests made to send them again.

In Ethernet, implementations using shared media (such as coax or a shared hub) receive all transmitted data. The data link layers in all these devices find out whether the destination ID matches the machine address, and discards the packets whose addresses do not match.

The IEEE has split this layer in two sub-layers called the logical link layer and media access layer. These layers will be discussed later at length.

Physical layer

The physical layer (the lowest of the seven layers) gets data packets from the data link layer and converts them into a series of electrical signals (or optical signals in case of optical fiber cabling) that represent ‘0’ and ‘1’ values in a digital transmission. These signals are sent via a transmission medium to the physical layer at the destination computer, which converts the signals back into a series of bit values. These values are grouped into packets and passed up to the data link layer of the same machine:

Matters managed or addressed at the physical layer include:

- Physical topology, i.e. physical layout of networks such as bus, star, ring, or hybrid network, type of media etc.,
- Connection types such as point-to-point and multipoint, pin assignments in connections
- Bit synchronization
- Baseband and broadband transmissions which decide how available media bandwidth is used
- Multiplexing, involving the combining of more than one data channel into one
- Termination of cables to prevent reflection of signals
- Encoding schemes for ‘0’ and ‘1’values, etc

1.9.3 The TCP/IP reference model

The US Department of Defense had sponsored a research network called ARPANET. ARPANET eventually connected hundreds of government establishments and universities together, first through leased telephone lines and later through radio networks and satellite links.

Figure 1.4

OSI and TCP/IP layers

Whereas the OSI model was developed in Europe by the **International Organization for Standardization** (ISO), the ARPA model (also known as the DOD (Department of Defense) or TCP/IP model) was developed in the USA by ARPA. Although they were developed by different bodies and at different points in time, both serve as models for a communications infrastructure and hence provide ‘abstractions’ of the same reality. The remarkable degree of similarity is therefore not surprising.

Whereas the OSI model has 7 layers, the ARPA model has 4 layers. The OSI layers map onto the ARPA model as follows:

- The OSI Session, Presentation and Applications layers are contained in the ARPA Process/Application Layer (nowadays referred to by the Internet community simply as the Application Level)
- The OSI Transport Layer maps onto the ARPA Host-to-Host Layer (nowadays referred to by the Internet community as the Host Level)
- The OSI Network Layer maps onto the ARPA Internet Layer (nowadays referred to by the Internet community as the Network Level)
- The OSI Physical and Data link Layers map onto the ARPA Network Interface Layer

1.10 OSI layers and IEEE layers

The Ethernet standard concerns itself with the data link layer in the OSI model and the physical layer below it. The IEEE thought it fit to further include sub-layers in the data link and the physical layers to add more clarity.

The mapping of the IEEE/Ethernet sub-layers to the OSI layers is shown schematically in Figure.1.5.

Figure 1.5

Implementation of OSI layers and IEEE sub-layers

The data link layer is divided into two sub-layers, namely, the logical link control (LLC), and the media access control (MAC) sub-layers.

The LLC layer is an IEEE standard (802.2) for identifying the data carried in a frame and is independent of the other 802 LAN standards. It will not vary with the LAN system used. The LLC control fields are intended for all IEEE 802 LAN systems and not just Ethernet.

The MAC sub-layer provides for shared access to the network and communicates directly with network interface cards, which each have a unique 48-bit MAC address, typically assigned by the manufacturer of the card. These MAC addresses are used to establish connections between computers on all IEEE 802 LAN systems.

The physical layer is divided into physical signaling sub-layers and media specifications. These sub-layers vary depending on whether 10-, 100-, or, 1000 Mbps Ethernet is being specified. Each IEEE 802.3 physical sub-layer specification has a three-part name that identifies its characteristics. The three parts of the name are (A) LAN speed in Mbps, (B) whether transmission is baseband or broadband and (C) the physical media type.

1.11 Network topologies

1.11.1 Broadcast and point-to-point topologies

The way the nodes are connected to form a network is known as its topology. There are many topologies available but they form two basic types, broadcast and point-to-point.

Broadcast topologies are those where the message ripples out from the transmitter to reach all nodes. There is no active regeneration of the signal by the nodes and so signal propagation is independent of the operation of the network electronics. This then limits the size of such networks.

Figure.1.6 shows an example of a broadcast topology.

Figure 1.6

Broadcast topology

In a point-to-point communications network, however, each node is communicating directly with only one node. That node may actively regenerate the signal and pass it on to its nearest neighbor. Such networks have the capability of being made much larger. Figure 1.7 shows some examples of point-to-point topologies.

Figure 1.7

Point-to-point topology

1.11.2 Logical and physical topologies

A logical topology defines how the elements in the network communicate with each other, and how information is transmitted through a network. The different types of media access methods determine how a node gets to transmit information along the network. In a bus topology, information is broadcast, and every node gets the same information within the amount of time it actually takes a signal to cover the entire length of cable (traveling at approximately two-thirds the speed of light). This time interval limits the maximum speed and size for the network. In a ring topology, each node hears from exactly one node and talks to exactly one other node. Information is passed sequentially, in an order determined by the physical interconnection of the nodes. A token mechanism is used to determine who has transmission rights, and a node can transmit only when it has this right.

On the other hand a physical topology defines the wiring layout for a network. This specifies how the elements in the network are connected to each other electrically. This arrangement will determine what happens if a node on the network fails. Physical topologies fall into three main categories: bus, star, and ring. Combinations of these can be used to form hybrid topologies in order to overcome weaknesses or restrictions in one or other of these three component topologies.

Bus (multidrop) topology

A bus refers to both a physical and a logical topology. As a physical topology, a bus describes a network in which each node is connected to a common single communication channel or ‘bus’. This bus is sometimes called a backbone, as it provides the spine for the network. Every node can hear each message packet as it goes past.

Logically, a passive bus is distinguished by the fact that packets are broadcast and every node gets the message at the same time. Transmitted packets travel in both directions along the bus, and need not go through the individual nodes, as in a point-to-point system. Rather, each node checks the destination address that is included in the message packet to determine whether that packet is intended for the specific node. When the signal reaches the end of the bus, an electrical terminator absorbs it to keep it from reflecting back again along the bus cable, possibly interfering with other messages already on the bus. Each end of a bus cable must be terminated, so that signals are removed from the bus when they reach the end.

In a bus topology, nodes should be far enough apart so that they do not interfere with each other. However, if the backbone bus cable is too long, it may be

necessary to boost the signal strength using some form of amplification, or repeater. The maximum length of the bus is limited by the size of the time interval that constitutes ‘simultaneous’ packet reception. Figure 1.8 illustrates the bus topology.

Figure 1.8

Bus topology

Bus topologies offer the following advantages:

- A bus uses relatively little cable compared to other topologies, and arguably has the simplest wiring arrangement
- Since nodes are connected by high impedance taps across a backbone cable, it is easy to add or remove nodes from a bus. This makes it easy to extend a bus topology
- Architectures based on this topology are simple and flexible
- The broadcasting of messages is advantageous for one-to-many data transmissions

The bus topology disadvantages are:

- There can be a security problem, since every node may see every message, even those that are not destined for it
- Troubleshooting is difficult, since the fault can be anywhere along the bus
- There is no automatic acknowledgement of messages, since messages get absorbed at the end of the bus and do not return to the sender
- The bus cable can be a bottleneck when network traffic gets heavy. This is because nodes can spend much of their time trying to access the network

Star (hub) topology

A star topology is a physical topology in which multiple nodes are connected to a central component, generally known as a hub. The hub usually is just a wiring center – that is, a common termination point for the nodes, with a single connection continuing from the hub. In some cases, the hub may actually be a file server (a central computer that contains a centralized file and control system),

with all its nodes attached directly to the server. As a wiring center, a hub may be connected to the file server or to another hub.

All message packets going to and from each node must pass through the hub to which the node is connected. The telephone system is the best-known example of a star topology, with lines to individual customers coming from a central telephone exchange location. An example of a star topology is shown in Figure 1.9.

Figure 1.9

Star topology

The star topology advantages are:

- Troubleshooting and fault isolation is easy
- It is easy to add or remove nodes, and to modify the cable layout
- Failure of a single node does not isolate any other node
- The inclusion of an intelligent hub allows remote monitoring of traffic for management purposes

The star topology disadvantages are:

- If the hub fails, the entire network fails. Sometimes a backup hub is included, to make it possible to deal with such a failure
- A star topology requires a lot of cable

Ring topology

A ring topology is both a logical and a physical topology. As a logical topology, a ring is distinguished by the fact that message packets are transmitted sequentially from node to node, in the order in which the nodes are connected, and as such, it is an example of a point-to-point system. Nodes are arranged in a closed loop, so that the initiating node is the last one to receive a packet. As a physical topology, a ring describes a network in which each node is connected to exactly two other nodes.

Information traverses a one-way path, so that a node receives packets from

exactly one node and transmits them to exactly one other node. A message packet travels around the ring until it returns to the node that originally sent it. In a ring topology, each node acts as a repeater, boosting the signal before sending it on. Each node checks whether the message packet's destination node matches its address. When the packet reaches its destination, the destination node accepts the message, and then sends it onwards to the sender, to acknowledge receipt.

Ring topologies use token passing to control access to the network, therefore, the token is returned to the sender with the acknowledgement. The sender then releases the token to the next node on the network. If this node has nothing to say, the node passes the token on to the next node, and so on. When the token reaches a node with a packet to send, that node sends its packet. Physical ring networks are rare, because this topology has considerable disadvantages compared to a more practical star-wired ring hybrid, which is described later.

Figure 1.10

Ring topology

The ring topology advantages are:

- A physical ring topology has minimal cable requirements
- No wiring center or closet is needed
- The message can be acknowledged automatically
- Each node can regenerate the signal

The ring topology disadvantages are:

- If any node goes down, the entire ring goes down
- Troubleshooting is difficult because communication is only one-way
- Adding or removing nodes disrupts the network
- There is a limit on the distance between nodes, depending on the transmission medium and driver technology used

1.11.3 Hybrid technologies

Besides these three main topologies, some of the more important variations will

now be considered. Once again, it should be clear that these are just variations, and should not be considered as topologies in their own right.

Star-wired ring topology

A star-wired ring topology, also a physical hub topology, is a hybrid physical topology that combines features of the star and ring topologies. Individual nodes are connected to a central hub, as in a star network. Within the hub, however, the connections are arranged into an internal ring. Thus, the hub constitutes the ring, which must remain intact for the network to function. The hubs, known as multi-station access units (MAUs) in IBM token ring network terminology, may be connected to other hubs. In this arrangement, each internal ring is opened and connected to the attached hubs, to create a larger, multi-hub ring.

The advantage of using star wiring instead of simple ring wiring is that it is easy to disconnect a faulty node from the internal ring. The IBM data connector is specially designed to close a circuit if an attached node is disconnected physically or electrically. By closing the circuit, the ring remains intact, but with one less node. In Token Ring networks, a secondary ring path can be established and used if part of the primary path goes down. The star-wired ring is illustrated in Figure 1.11.

Figure 1.11

Star wired ring

The advantages of a star-wired ring topology include:

- Troubleshooting, or fault isolation, is relatively easy
- The modular design makes it easy to expand the network, and makes layouts extremely flexible
- Individual hubs can be connected to form larger rings
- Wiring to the hub is flexible

The disadvantages of a star-wired ring topology include:

- Configuration and cabling may be complicated because of the extreme flexibility of the arrangement

Distributed star topology

A distributed star topology is a physical topology that consists of two or more hubs, each of which is the center of a star arrangement. This type of topology is common, and it is generally known simply as a star topology. A good example of such a topology is an ARCnet network with at least one active hub and one or more active or passive hubs. The 100VG Any LAN utilizes a similar topology as shown in Figure.1.12.

Figure 1.12

Cascaded hubs

Mesh topology

A mesh topology is a physical topology in which there are at least two paths to and from every node. This type of topology is advantageous in hostile environments in which connections are easily broken. If a connection is broken, at least one substitute path is always available. A more restrictive definition requires each node to be connected directly to every other node. Because of the severe connection requirements, such restrictive mesh topologies are feasible only for small networks.

Figure 1.13

Mesh topology

Tree topology

A tree topology, also known as a distributed bus or a branching tree topology, is a hybrid physical topology that combines features of star and bus topologies. Several buses may be daisy-chained together, and there may be branching at the connections (which will be hubs). The starting end of the tree is known as the root or head end. This type of topology is used in delivering cable television services.

The advantages of a tree topology are:

- The network is easy to extend by just adding another branch, and that fault isolation is relatively easy

The disadvantages include:

- If the root goes down, the entire network goes down
- If any hub goes down, all branches of that hub go down
- Access becomes a problem if the entire network becomes too big

Figure 1.14

Tree topology

1.12 Network communication

There are two basic types of communications processes for transferring data across networks viz. circuit switched and packet switched. These are illustrated in Figure 1.15.

Figure 1.15

Circuit switched data and packet switched data

1.12.1 Circuit switched data

In a circuit switched process, a continuous connection is made across the network between the two different points. This is a temporary connection, which remains in place as long as both parties wish to communicate, i.e. until the connection is terminated. All the network resources are available for the exclusive use of these two parties whether they are sending data or not. When the connection is terminated, the network resources are released for other users. A telephone call is an example of a circuit switched connection.

The advantage of circuit switching is that the users have an exclusive channel available for the transfer of their data at any time while the connection is made. The obvious disadvantage is the cost of maintaining the connection when there is little or no data being transferred. Such connections can be very inefficient for the bursts of data that are typical of many computer applications.

Packet switched data

Packet switching systems improve the efficiency of the transfer of bursts of data by sharing one communications channel with other similar users. This is analogous to the efficiencies of the mail system.

When you send a letter by mail, you post the stamped, addressed envelope containing the letter in your local mailbox. At regular intervals the mail company collects all the letters from your locality and takes them to a central sorting facility where the letters are sorted in accordance with the addresses of their destinations. All the letters for each destination are sent off in common mailbags to those locations, and are subsequently delivered in accordance with their addresses. Here we have economies of scale where many letters are carried at one time and are delivered by the one visit to the recipient's street/locality. Here efficiency is more important than speed, and some delay is normal – within acceptable limits. To complete the analogy, a courier service gives us faster, exclusive delivery, with the courier delivering our message door-to-door at a much higher cost, and is equivalent to the circuit switched connection.

Packet switched messages are broken into a series of packets of certain maximum size, each containing the destination and source addresses and a packet sequence number. The packets are sent over a common communications channel, possibly interleaved with those of other users. All the receivers on the channel check the destination addresses of all packets and accept only those carrying their address. Messages sent in multiple packets are reassembled in the correct order by the destination node.

Packets for a specific destination do not necessarily all follow the same path. As they travel through the network they may be separated and handled independently from each other, but eventually arrive at their correct destination. For this reason, packets often arrive at the destination node out of their transmitted sequence. Some packets may even be held up temporarily (stored) at a node, due to unavailable lines or technical problems that might arise on the network. When the time is right, the node then allows the packet to pass or to be

‘forwarded’.

Datagrams and virtual circuits

Packet switched services generally support two types of service – datagrams and virtual circuits.

Datagram service sends each packet as a self-contained entity, which is neither pre-arranged nor acknowledged. This is similar to the mail service described above.

Virtual circuits are used where a large message is being sent in many packets. The loss of any one of these packets would invalidate the whole message, so a higher quality delivery system is used. A unique logical circuit is established for the duration of the transfer of the message. Sequence numbers are used to identify the individual packets making up the message and each packet is acknowledged to confirm successful delivery. Missing or damaged packets can be re-transmitted. It provides similar service to circuit switching and is analogous to use of a registered mail service.