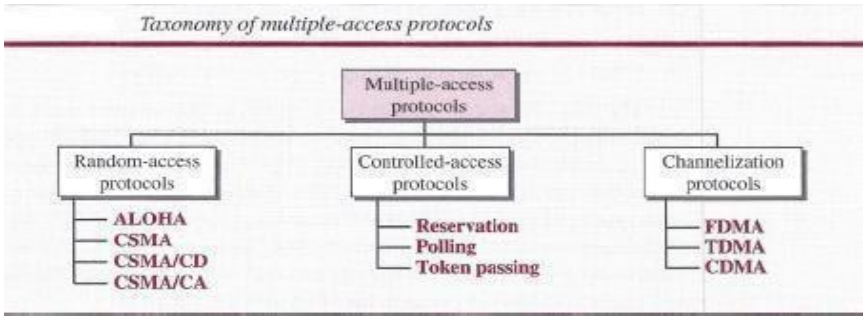
UNIT- II
Multiple Access Protocols

Multiple Access Protocols

When nodes or stations are connected and use a common link, called a *multipoint* or *broadcast link*, we need a multiple-access protocol to coordinate access to the link. Many protocols have been devised to handle access to a shared link. All of these protocols belong to a sub layer in the data-link layer called *media access control (MAC).* We categorize them into three groups:



- The first section discusses random-access protocols. Four protocols, ALOHA, CSMA, *CSMA/CD,* and *CSMA/CA,* are described in this section. These protocols are mostly used in LANs and WANs.
- The second section discusses controlled-access protocols. Three protocols, reservation, polling, and token-passing, are described in this section. Some of these protocols are used in LANs, but others have some historical value.
- The third section discusses channelization protocols. Three protocols, FDMA, TDMA, and CDMA are described in this section. These protocols are used in cellular telephony.

RANDOM ACCESS

In random-access or contention methods, no station is superior to another station and none is assigned control over another. At each instance, a station that has data to send uses a procedure defined by the protocol to make a decision on whether or not to send. This decision depends on the state of the medium (idle or busy). In other words, each station can transmit when it desires on the condition that it follows the predefined procedure, including testing the state of the medium.

ALOHA:

In the 1970s, Norman Abramson and his colleagues at the University of Hawaii devised a new and elegant method to solve the channel allocation problem. Their work has been extended by many researchers since then (Abramson, 1985).Although Abramson's work, called the ALOHA system, used ground-based radio broadcasting, the basic idea is applicable to any system in which uncoordinated users are competing for the use of a single shared channel. There are two versions of ALOHA: pure and slotted. They differ with respect to whether time is divided into discrete slots into which all frames must fit. Pure ALOHA does not require global time synchronization; slotted ALOHA does.

Pure ALOHA:

The basic idea of an ALOHA system is simple: let users transmit whenever they have data to be sent. There will be collisions, of course, and the colliding frames will be damaged. However, due to the feedback property of broadcasting, a sender can always find out whether its frame was destroyed by listening to the channel, the same way other users do. With a LAN, the feedback is immediate; with a satellite, there is a delay of 270 msec before the sender knows if the transmission was successful.

If listening while transmitting is not possible for some reason, acknowledgements are needed. If the frame was destroyed, the sender just waits a random amount of time and sends it again. The waiting time must be random or the same frames will collide over and over, in lockstep. Systems in which multiple users share a common channel in a way that can lead to conflicts are widely known as contention systems. A sketch of frame generation in an ALOHA system is given in Fig.1. We have made the frames all the same length because the throughput of ALOHA systems is maximized by having a uniform frame size rather than by allowing variable length frames.

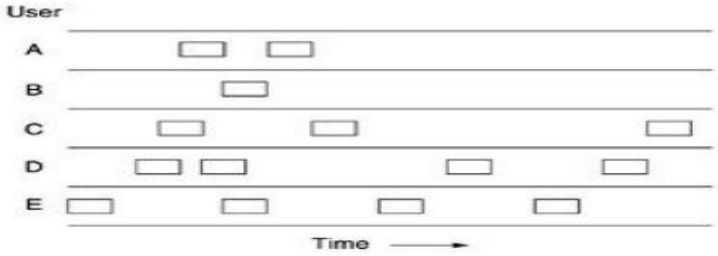


Fig.1 In pure ALOHA, frames are transmitted at completely arbitrary times.

Whenever two frames try to occupy the channel at the same time, there will be a collision and both will be garbled. If the first bit of a new frame overlaps with just the last bit of a frame almost finished, both frames will be totally destroyed and both will have to be retransmitted later. The checksum cannot (and should not) distinguish between a total loss and a near miss.Let the "frame time" denote the amount of time needed to transmit the standard, fixed length frame (i.e., the frame length divided by the bit rate). At this point we assume that the infinite population of users generates new frames according to a Poisson distribution with mean N frames per frame time. (The infinite-population assumption is needed to ensure that N does no decrease as users become blocked.) If N > 1, the user community is generating frames at a higher rate than the channel can handle, and nearly every frame will suffer a collision. For reasonable throughput we would expect 0 < N < 1. In addition to the new frames, the stations also generate retransmissions of frames that previously suffered collisions. Let us further assume that the probability of k transmission attempts per frame time, old and new combined, is also Poisson, with mean G per frame time. Clearly, G≥N. At low load (i.e., N 0), there will be few collisions, hence few retransmissions, so G N. At high load there will be many collisions, so G > N.

Under all loads, the throughput, S, is just the offered load, G, times the probability, P0, of a transmission succeeding—that i s, S = GP0, where P0 is the probability that a frame does not suffer a collision. A frame will not suffer a collision if no other frames are sent within one frame time of its start, as shown in Fig.2.

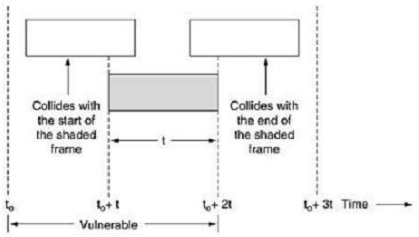


Fig.2. Vulnerable period for the shaded frame

Under what conditions will the shaded frame arrive undamaged? Let t be the time required to send a frame. If any other user has generated a frame between time t0 and t0+tr, the end of that frame will collide with the beginning of the shaded one. In fact, the shaded frame's fate was already sealed even before the first bit was sent, but since in pure ALOHA a station does not listen to the channel before transmitting, it has no way of knowing that another frame was already underway. Similarly, any other frame started between t0+t and t0+2t will bump into the end of the shaded frame.

The probability that k frames are generated during a given frame time is given by the Poisson distribution:

Equation

$$\Pr[k] = \frac{G^k e^{-G}}{k!}$$

so the probability of zero frames is just e-G. In an interval two frame times long, the mean number of frames generated is 2G. The probability of no other traffic being initiated during the entire vulnerable period is thus given by P0 = e -2G. Using S = GP0, we get

$$S = Ge^{-2G}$$

The relation between the offered traffic and the throughput is shown in Fig. 4-3. The maximum throughput occurs at G = 0.5, with S = 1/2e, which is about 0.184. In other words, the best we can hope for is a channel utilization of 18 per cent. This result is not very encouraging, but with everyone transmitting at will, we could hardly have expected a 100 per cent success rate.

Slotted ALOHA:

In 1972, Roberts published a method for doubling the capacity of an ALOHA system (Robert, 1972). His proposal was to divide time into discrete intervals, each interval corresponding to one frame. This approach requires the users to agree on slot boundaries. One way to achieve synchronization would be to have one special station emit a pip at the start of each interval, like a clock. In Roberts' method, which has come to be known as slotted ALOHA, in contrast to Abramson's pure ALOHA, a computer is not permitted to send whenever a carriage return is typed. Instead, it is required to wait for the beginning of the next slot. Thus, the continuous pure ALOHA is turned into a discrete one. Since the vulnerable period is now halved, the probability of no other traffic during the same slot as our test frame is e-G which leads to

Equation

$$S = Ge^{-G}$$

As you can see from Fig.3, slotted ALOHA peaks at G = 1, with a throughput of S=1/e or about 0.368, twice that of pure ALOHA. If the system is operating at G = 1, the probability of an empty slot is 0.368. The best we can hope for using slotted ALOHA is 37 percent of the slots empty, 37 percent successes, and 26 percent collisions. Operating at higher values of G reduces the number of empties but increases the number of collisions exponentially.To see how this rapid growth of collisions with G comes about, consider the transmission of a test frame. The probability that it will avoid a collision is e-G, the probability that all the other users are silent in that slot. The probability of a collision is then just 1 – e -G. The probability of a transmission requiring exactly k attempts, (i.e., k – 1 colli sions followed by one success) is

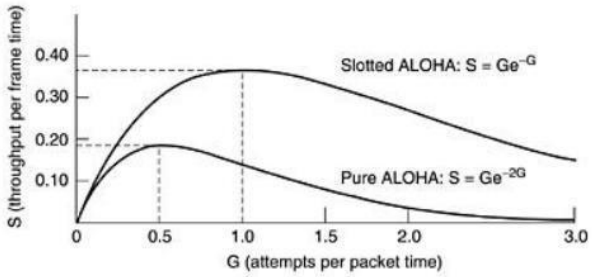


Fig.3 Throughput versus offered traffic for ALOHA systems.

$$P_k = e^{-G}(1 - e^{-G})^{k-1}$$

The expected number of transmissions, E, per carriage return typed is then

$$E = \sum_{k=1}^{\infty} kP_k = \sum_{k=1}^{\infty} ke^{-G}(1 - e^{-G})^{k-1} = e^{G}$$

As a result of the exponential dependence of E upon G, small increases in the channel load can drastically reduce its performance.

CSMA

Carrier Sense Multiple Access Protocols:

With slotted ALOHA the best channel utilization that can be achieved is 1/e. This is hardly surprising, since with stations transmitting at will, without paying attention to what the other stations are doing, there are bound to be many collisions. In local area networks, however, it is possible for stations to detect what other stations are doing, and adapt their behavior accordingly. These networks can achieve a much better utilization than 1/e. In this section we will discuss some protocols for improving performance. Protocols in which stations listen for a carrier (i.e., a transmission) and act accordingly are called carrier sense protocols. A number of them have been proposed. Kleinrock and Tobagi (1975) have analyzed several such protocols in detail. Below we will mention several versions of the carrier sense protocols

1. **1-persistent CSMA:**

The first carrier sense protocol that we will study here is called **1-persistent CSMA** (Carrier Sense Multiple Access). When a station has data to send, it first listens to the channel to see if anyone else is transmitting at that moment. If the channel is busy, the station waits until it becomes idle. When the station detects an idle channel, it transmits a frame. If a collision occurs, the station waits a random amount of time and starts all over again. The protocol is called 1-persistent because the station transmits with a probability of 1 when it finds the channel idle. The propagation delay has an important effect on the performance of the protocol. There is a small chance that just after a station begins sending, another station will become ready to send and sense the channel. If the first station's signal has not yet reached the second one, the latter will sense an idle channel and will also begin sending, resulting in a collision. The longer the propagation delay, the more important this effect becomes, and the worse the performance of the protocol. Even if the propagation delay is zero, there will still be collisions. If two stations become ready in the middle of a third station's transmission, both will wait politely until the transmission ends and then both will begin transmitting exactly simultaneously, resulting in a collision. If they were not so impatient, there would be fewer collisions. Even so, this protocol is far better than pure ALOHA because both stations have the decency to desist from interfering with the third station's frame. Intuitively, this approach will lead to a higher performance than pure ALOHA. Exactly the same holds for slotted ALOHA.

2. Non-persistent CSMA:

A second carrier sense protocol is **nonpersistent CSMA**. In this protocol, a conscious attempt is made to be less greedy than in the previous one. Before sending, a station senses the channel. If no one else is sending, the station begins doing so itself. However, if the channel is already in use, the station does not continually sense it for the purpose of seizing it immediately upon detecting the end of the previous transmission. Instead, it waits a random period of time and then repeats the algorithm. Consequently, this algorithm leads to better channel utilization but longer delays than 1-persistent CSMA.

3. P-persistent CSMA:

The last protocol is **p-persistent CSMA**. It applies to slotted channels and works as follows. When a station becomes ready to send, it senses the channel. If it is idle, it transmits with a probability p. With a probability q = 1 - p, it defers until the next slot. If that slot is also idle, It either transmits or defers again, with probabilities p and q. This process is repeated until either the frame has been transmitted or another station has begun transmitting. In the latter case, the unlucky station acts as if there had been a collision (i.e., it waits a random time and starts again). If the station initially senses the channel busy, it waits until the next slot and applies the above algorithm.

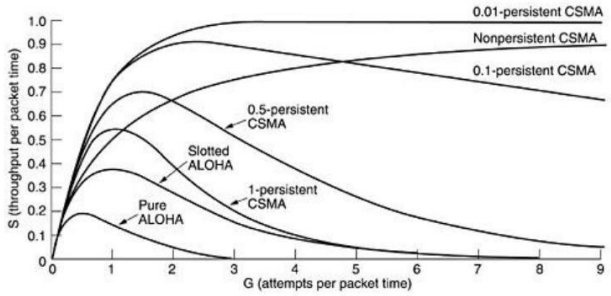


Fig.4 Comparison of the channel utilization versus load for various random access protocols

CSMA with Collision Detection:

Persistent and nonpersistent CSMA protocols are clearly an improvement over ALOHA because they ensure that no station begins to transmit when it senses the channel busy. Another improvement is for stations to abort their transmissions as soon as they detect a collision. In other words, if two stations sense the channel to be idle and begin transmitting simultaneously, they will both detect the collision almost immediately. Rather than finish transmitting their frames, which are irretrievably garbled anyway, they should abruptly stop transmitting as soon as the collision is detected. Quickly terminating damaged frames saves time and bandwidth. This protocol, known as CSMA/CD (CSMA with Collision Detection) is widely used on LANs in the MAC sub layer. In particular, it is the basis of the popular Ethernet LAN, so it is worth devoting some time to looking at it in detail. CSMA/CD, as well as many other LAN protocols, uses the conceptual model of Fig.5. At the point marked t0, a station has finished transmitting its frame. Any other station having a frame to send may now attempt to do so. If two or more stations decide to transmit simultaneously, there will be a collision. Collisions can be detected by looking at the power or pulse width of the received signal and comparing it to the transmitted signal.

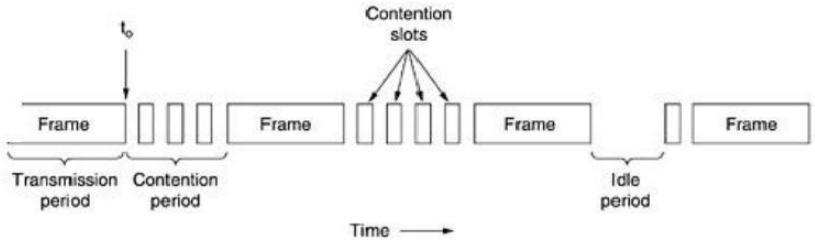
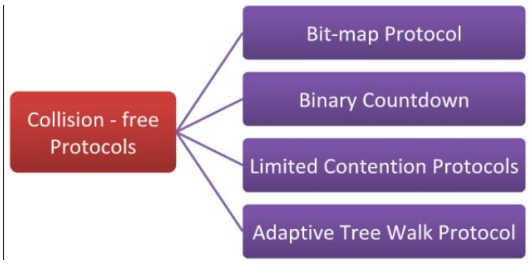


Fig.5. CSMA/CD can be in one of three states: contention, transmission, or idle

After a station detects a collision, it aborts its transmission, waits a random period of time, and then tries again, assuming that no other station has started transmitting in the meantime. Therefore, our model for CSMA/CD will consist of alternating contention and transmission periods, with idle periods occurring when all stations are quiet (e.g., for lack of work). Suppose that two stations both begin transmitting at exactly time t0. How long will it take them to realize that there has been a collision? The answer to this question is vital to determining the length of the contention period and hence what the delay and throughput will be. The minimum time to detect the collision is then just the time it takes the signal to propagate from one station to the other. Based on this reasoning, you might think that a station not hearing a collision for a time equal to the full cable propagation time after starting its transmission could be sure it had seized the cable. By "seized," we mean that all other stations knew it was transmitting and would not interfere. This conclusion is wrong. Consider the following worst-case scenario. Let the time for a signal to propagate between the two farthest stations be. At t0, one station begins transmitting. At, an instant before the signal arrives at the most distant station, that station also begins transmitting. Of course, it detects the collision almost instantly and stops, but the little noise burst caused by the collision does not get back to the original station until time. In other words, in the worst case a station cannot be sure that it has seized the channel until it has transmitted for without hearing a collision. For this reason we will model the contention interval as a slotted ALOHA system with slot width. On a 1-km long coaxial cable, . For simplicity we will assume that each slot contains just 1 bit. Once the channel has been seized, a station can transmit at any rate it wants to, of course, not just at 1 bit per sec.

Collision-Free Protocols

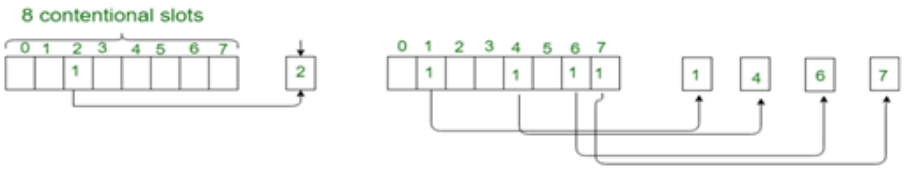
- Collision – free protocols are devised so that collisions do not occur. Protocols like CSMA/CD and CSMA/CA nullifies the possibility of collisions once the transmission channel is acquired by any station. However, collision can still occur during the contention period if more than one stations starts to transmit at the same time.



1. Bit – map Protocol

Bit map protocol is collision free Protocol in In bitmap protocol method, each contention period consists of exactly N slots. if any station has to send frame, then it transmits a 1 bit in the respective slot. For example if station 2 has a frame to send, it transmits a 1 bit during the second slot.In general Station 1 Announce the fact that it has a frame questions by inserting a 1 bit into slot 1.

In this way, each station has complete knowledge of which station wishes to transmit. There will never be any collisions because everyone agrees on who goes next. Protocols like this in which the desire to transmit is broadcasting for the actual transmission are called *Reservation Protocols*.



A Bit-map Protocol.

For analyzing the performance of this protocol, We will measure time in units of the contention bits slot, with a data frame consisting of *d* time units. Under low load conditions, the bitmap will simply be repeated over and over, for lack of data frames. All the stations have something to send all the time at high load, the N bit contention period is prorated over N frames, yielding an overhead of only 1 bit per frame.Generally, high numbered stations have to wait for half a scan before starting to transmit low numbered stations have to wait for half a scan(N/2 bit slots) before starting to transmit, low numbered stations have to wait on an average 1.5 N slots.

2. Binary Countdown

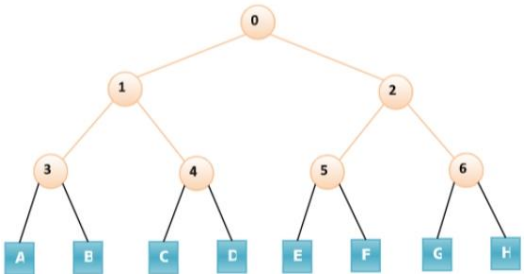
This protocol overcomes the overhead of 1 bit per station of the bit – map protocol. Here, binary addresses of equal lengths are assigned to each station. For example, if there are 6 stations, they may be assigned the binary addresses 001, 010, 011, 100, 101 and 110. All stations wanting to communicate broadcast their addresses. The station with higher address gets the higher priority for transmitting.

3. Limited Contention Protocols

These protocols combines the advantages of collision based protocols and collision free protocols. Under light load, they behave like ALOHA scheme. Under heavy load, they behave like bitmap protocols.

4. Adaptive Tree Walk Protocol

Initially all nodes (A, B ……. G, H) are permitted to compete for the channel. If a node is successful in acquiring the channel, it transmits its frame. In case of collision, the nodes are divided into two groups (A, B, C, D in one group and E, F, G, H in another group). Nodes belonging to only one of them is permitted for competing. This process continues until successful transmission occurs.

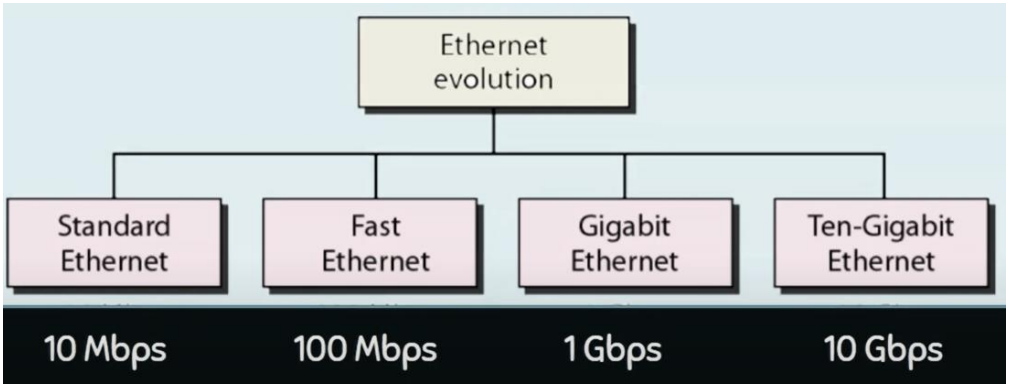


Difference between CSMA/CA and CSMA/CD?

Key	CSMA/CA	CSMA/CD
Effectiveness	CSMA/CA is effective before a collision.	CSMA/CD is effective after a collision.
Network Type	CSMA/CA is generally used in wireless networks.	CSMA/CD is generally used in wired networks.
Recovery Time	CSMA/CA minimizes the risk of collision.	CSMA/CD reduces the recovery time.
Conflict Management	CSMA/CA initially transmits the intent to send the data. Once an acknowledgment is received, the sender sends the data.	CSMA/CD resends the data frame in case a conflict occurs during transmission.
IEEE Standards	CSMA/CA is part of the IEEE 802.11 standard.	CSMA/CD is part of the IEEE 802.3 standard.
Efficiency	CSMA/CA is similar in efficiency as CSMA.	CSMA/CD is more efficient than CSMA.

Ethernet

Ethernet is a set of technologies and protocols that are used primarily in LANs. However, Ethernet can also be used in MANs and even WANs. It was first standardized in the 1980s as IEEE 802.3 standard.



Characteristics

Let us first discuss some characteristics of the Standard Ethernet.

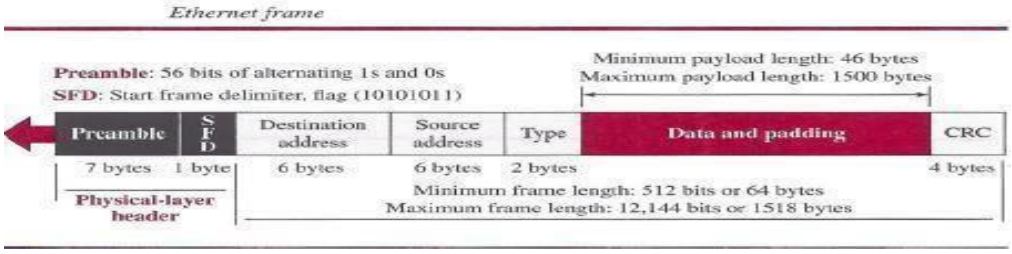
Connectionless and Unreliable Service

Ethernet provides a connectionless service, which means each frame sent is independent of the previous or next frame. Ethernet has no connection establishment or connection termination phases. The sender sends a frame whenever it has it; the receiver mayor may not be ready for it. The sender may overwhelm the receiver with frames, which may result in dropping frames. If a frame drops, the sender will not know about it. Since IP, which is using the service of Ethernet, is also connectionless, it will not know about it either. If the transport layer is also a connectionless protocol, such as UDP, the frame is lost and salvation may only come from the application layer.

However, if the transport layer is TCP, the sender TCP does not receive acknowledgment for its segment and sends it again. Ethernet is also unreliable like IP and UDP. If a frame is corrupted during transmission and the receiver finds out about the corruption, which has a high level of probability of happening because of the CRC-32, the receiver drops the frame silently. It is the duty of high-level protocols to find out about it.

Frame Length

Ethernet has imposed restrictions on both the minimum and maximum lengths of a frame, a shown in Figure 5.



Preamble This field contains 7 bytes (56 bits) of alternating Os and Is that alert the receiving system to the coming frame and enable it to synchronize its clock if it's out of synchronization. The pattern provides only an alert and a timing pulse. The 56-bit pattern allows the stations to miss some bits at the beginning of the frame. The *preamble* is actually added at the physical layer and is not (formally) part of the frame.

Start frame delimiter (SFD) This field (1 byte: 10101011) signals the beginning of the frame. The SFD warns the station or stations that this is the last chance for synchronization. The last 2 bits are (llh and alert the receiver that the next field is the destination address. This field is actually a flag that defines the beginning of the frame. We need to remember that an Ethernet frame is a variable-length frame. It needs a flag to define the beginning of the frame. The SFD field is also added at the physical layer.

Destination address (DA) This field is six bytes (48 bits) and contains the link layer address of the destination station or stations to receive the packet. When the receiver sees its own link-layer address, or a multicast address for a group that the receiver is a member of, or a broadcast address, it decapsulates the data from the frame and passes the data to the upper layer protocol defined by the value of the type field.

Source address (SA) This field is also six bytes and contains the link-layer address of the sender of the packet.

Type This field defines the upper-layer protocol whose packet is encapsulated in the frame. This protocol can be IP, ARP, OSPF, and so on. In other words, it serves the same purpose as the protocol field in a datagram and the port number in a segment or user datagram. It is used for multiplexing and de-multiplexing.

Data This field carries data encapsulated from the upper-layer protocols. It is a minimum of 46 and a maximum of 1500 bytes. We discuss the reason for these minimum and maximum values shortly. If the data coming from the upper layer is more than 1500 bytes, it should be fragmented and encapsulated in more than one frame. If it is less than 46 bytes, it needs to be padded with extra Os. A padded data frame is delivered to the upper-layer protocol as it is (without removing

the padding), which means that it is the responsibility of the upper layer to remove or, in the case of the sender, to add the padding. The upper-layer protocol needs to know the length of its data. For example, a datagram has a field that defines the length of the data. *CRC* The last field contains error detection information, in this case a CRC-32. The CRC is calculated over the addresses, types, and data field. If the receiver calculates the CRC and finds that it is not zero (corruption in transmission), it discards the frame.

Frame Length

Ethernet has imposed restrictions on both the minimum and maximum lengths of a frame. The minimum length restriction is required for the correct operation of CSMAlCD. An Ethernet frame needs to have a minimum length of 512 bits or 64 bytes. Part of this length is the header and the trailer. If we count 18 bytes of header and trailer (6 bytes of source address, 6 bytes of destination address, 2 bytes of length or type, and 4 bytes of CRC), then the minimum length of data from the upper layer is 64 - 18 = 46 bytes. If the upper-layer packet is less than 46 bytes, padding is added to make up the difference. The standard defines the maximum length of a frame (without preamble and SFD field) as 1518 bytes. If we subtract the 18 bytes of header and trailer, the maximum length of the payload is 1500 bytes. The maximum length restriction has two historical reasons. First, memory was very expensive when Ethernet was designed; a maximum length restriction helped to reduce the size of the buffer. Second, the maximum length restriction prevents one station from monopolizing the shared medium, blocking other stations that have data to send.

Minimum frame length: 64 bytes
Maximum frame length: 1518 bytes
Minimum data length: 46 bytes
Maximum data length: 1500 bytes

Addressing

Each station on an Ethernet network (such as a PC, workstation, or printer) has its own network interface card (NIC). The NIC fits inside the station and provides the station with a link-layer address. The Ethernet address is 6 bytes (48 bits), normally written in hexadecimal notation, with a colon between the bytes. For example, the following shows an Ethernet MAC address:

4A:30:10:21:10:1A

Transmission of Address Bits

The way the addresses are sent out online is different from the way they are written in hexadecimal notation. The transmission is left to right, byte by byte; however, for each byte, the least significant bit is sent first and the most significant bit is sent last. This means that the bit that defines an address as unicast or multicast arrives first at the receiver. This helps the receiver to immediately know if the packet is unicast or multicast.

Physical Layer

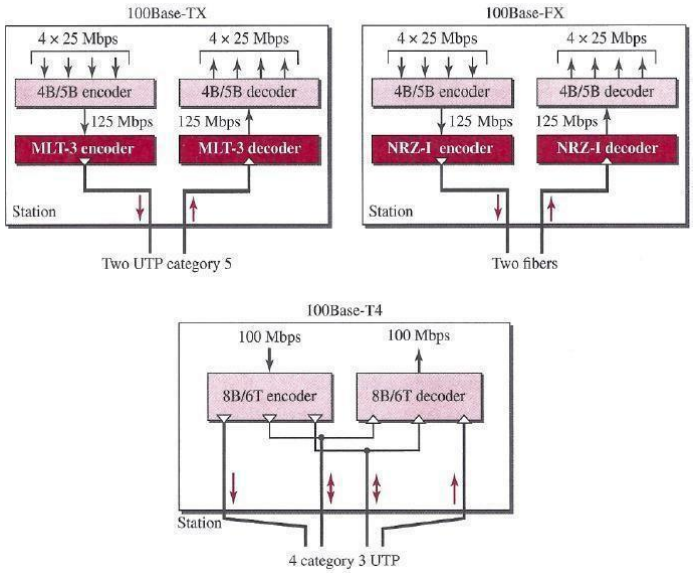
To be able to handle a 100 Mbps data rate, several changes need to be made at the physical layer.

Topology

Fast Ethernet is designed to connect two or more stations. If there are only two stations, they can be connected point-to-point. Three or more stations need to be connected in a star topology with a hub or a switch at the center.

Encoding

Manchester encoding needs a 200-Mbaud bandwidth for a data rate of 100 Mbps, which makes it unsuitable for a medium such as twisted-pair cable. For this reason, the Fast Ethernet designers sought some alternative encoding/decoding scheme. However, it was found that one scheme would not perform equally well for all three implementations. Therefore, three different encoding schemes were chosen



100Base- TX uses two pairs of twisted-pair cable (either category S UTP or STP). For this implementation, the MLT-3 scheme was selected since it has good bandwidth performance. (However, since MLT-3 is not a self-synchronous line coding scheme, 4B/SB block coding is used to provide bit synchronization by preventing the occurrence of a long sequence of Os and Is. This creates a data rate of 125 Mbps, which is fed into MLT-3 for encoding.

100Base-FX uses two pairs of fiber-optic cables. Optical fiber can easily handle high bandwidth requirements by using simple encoding schemes. The designers of 100Base-FX selected the NRZ-I encoding scheme for this implementation. However, NRZ-I has a bit synchronization problem for long sequences of 0s (or Is, based on the encoding). To overcome this problem, the designers used 4B/5B block encoding, as we described for 100Base- TX. The block encoding increases the bit rate from 100 to 125 Mbps, which can easily be handled by fiber-optic cable. A 100Base-TX network can provide a data rate of 100 Mbps, but it requires the use of category 5 UTP or STP cable.

This is not cost-efficient for buildings that have already been wired for voice-grade twisted-pair (category 3). A new standard, called ***lOOBase-T4,*** was designed to use category 3 or higher UTP. The implementation uses four pairs of UTP for transmitting 100 Mbps. Encoding/decoding in 100Base-T4 is more complicated. As this implementation uses category 3 UTP, each twisted-pair cannot easily handle more than 25 Mbaud. In this design, one pair switches between sending and receiving. Three pairs of UTP category 3, however, can handle only 75 Mbaud (25 Mbaud) each. We need to use an encoding scheme that converts 100 Mbps to a 75 Mbaud signal. 8B/6T satisfies this requirement. In 8B/6T, eight data elements are encoded as six signal elements. This means that 100 Mbps uses only (6/8) x 100 Mbps, or 75 Mbaud.

Network Devices (Hub, Repeater, Bridge, Switch, Router, Gateways and B router)

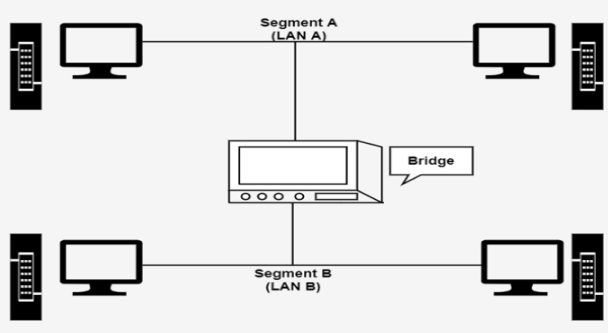
1. **Repeater** – A repeater operates at the physical layer. Its job is to regenerate the signal over the same network before the signal becomes too weak or corrupted so as to extend the length to which the signal can be transmitted over the same network. An important point to be noted about repeaters is that they do not amplify the signal. When the signal becomes weak, they copy the signal bit by bit and regenerate it at the original strength.
2. **Hub** – A hub is basically a multiport repeater. A hub connects multiple wires coming from different branches, for example, the connector in star topology which connects different stations. Hubs cannot filter data, so data packets are sent to all connected devices. In other words, collision domain of all hosts connected through Hub remains one. Also, they do not have intelligence to find out best path for data packets which leads to inefficiencies and wastage.

   Types of Hub

   1. **Active Hub:-** These are the hubs that have their power supply and can clean, boost, and relay the signal along with the network. It serves both as a repeater as well as a wiring center. These are used to extend the maximum distance between nodes.
   2. **Passive Hub:-** These are the hubs that collect wiring from nodes and power supply from the active hub. These hubs relay signals onto the network without cleaning and boosting them and can't be used to extend the distance between nodes.

   3. Intelligent Hub:- It works like an active hub and includes remote management capabilities. They also provide flexible data rates to network devices. It also enables an administrator to monitor the traffic passing through the hub and to configure each port in the hub.
3. **Switch** – A switch is a multi port bridge with a buffer and a design that can boost its efficiency(large number of ports imply less traffic) and performance. Switch is data link layer device. Switch can perform error checking before forwarding data, that makes it very efficient as it does not forward packets that have errors and forward good packets selectively to correct port only.
4. **Routers** – A router is a device like a switch that routes data packets based on their IP addresses. Router is mainly a Network Layer device. Routers normally connect LANs and WANs together and have a dynamically updating routing table based on which they make decisions on routing the data packets. Router divide broadcast domains of hosts connected through it.

5. **Gateway** – A gateway, as the name suggests, is a passage to connect two networks together that may work upon different networking models. They basically works as the messenger agents that take data from one system, interpret it, and transfer it to another system. Gateways are also called protocol converters and can operate at any network layer. Gateways are generally more complex than switch
6. **Bridge**
   A bridge in a computer network is a device used to connect multiple LANs together with a larger Local Area Network (LAN). The mechanism of network aggregation is known as bridging. The bridge is a physical or hardware device but operates at the OSI model's data link layer and is also known as a layer of two switches. The primary responsibility of a switch is to examine the incoming traffic and determine whether to filter or forward it.
   Basically, a bridge in computer networks is used to divide network connections into sections, now each section has separate bandwidth and a separate collision domain. Here bridge is used to improve network performance.



MAC Layer

The Media Access Control (MAC) data communication Networks protocol sub-layer, also known as the Medium Access Control, is a sub-layer of the data link layer specified in the seven-layer OSI model. The medium access layer was made necessary by systems that share a common communications medium. Typically these are local area networks. The MAC layer is the "low" part of the second OSI layer, the layer of the "data link". In fact, the IEEE divided this layer into two layers "above" is the control layer the logical connection (Logical Link Control, LLC) and "down" the control layer The medium access (MAC). The LLC layer is standardized by the IEEE as the 802.2 since the beginning 1980 Its purpose is to allow level 3 network protocols (for eg IP) to be based on a single layer (the LLC layer) regardless underlying protocol used, including WiFi, Ethernet or Token Ring, for example. All WiFi data packets so carry a pack LLC, which contains itself packets from the upper network layers. The header of a packet LLC indicates the type of layer 3 protocol in it: most of the time, it is IP protocol, but it could be another protocol, such as IPX (Internet Packet Exchange) for example. Thanks to the LLC layer, it is possible to have at the same time, on the same network, multiple Layer 3 protocols.

Functions of the MAC Layer

The MAC layer also defines the network addresses: all devices have an identifier of 48 bits (6 bytes) known as the "MAC address". The first three bytes identify the manufacturer of the network equipment. For example, in hexadecimal notation, 00-00-0c corresponds to Cisco constructor, 00-04-23 corresponds to Intel Corporation, etc. The following three bytes define an identifier one chosen by the manufacturer, for example 8B-B5-0B. An address will look like for example: 00-

04-23-8B-B5-0B. Any network adapter (WLAN, Ethernet or other) therefore has in principle a MAC address, supposed to be unique. One can communicate with a device by sending packets on the network, denominated in its MAC address. Other standardized protocols by the IEEE, such as Ethernet or Token Ring, have the same definition of the MAC address. This allows stations to different types of networks to communicate with each other: it suffices to connect different networks together with "bridges" (bridge).

DATA LINK LAYER SWITCHING

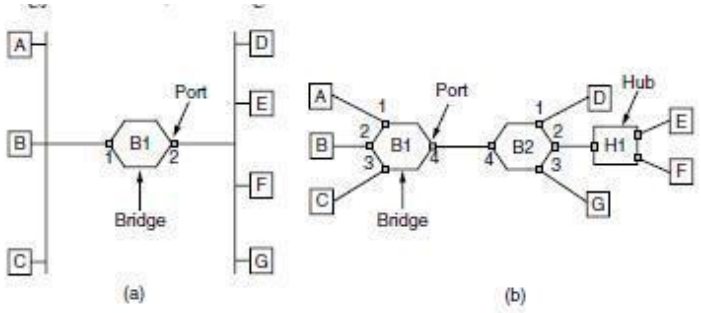
Many organizations have multiple LANs and wish to connect them. Would it not be convenient if we could just join the LANs together to make a larger LAN? In fact, we can do this when the connections are made with devices called **bridges**. are a modern name for bridges; they provide functionality that goes beyond classic Ethernet and Ethernet hubs to make it easy to join multiple LANs into a larger and faster network.We shall use the terms ''bridge'' and ''switch'' interchangeably. Bridges operate in the data link layer, so they examine the data link layer addresses to forward frames. Since they are not supposed to examine the payload field of the frames they forward, they can handle IP packets as well as other kinds of packets, such as AppleTalk packets. In contrast, *routers* examine the addresses in packets and route based on them, so they only work with the protocols that they were designed to handle. physical LANs into a single logical LAN. We will also look at how to do the reverse and treat one physical LAN as multiple logical LANs, called **VLANs** (**Virtual LANs**). Both technologies provide useful flexibility for managing networks. For a comprehensive treatment of bridges, switches, and related topics, see Seifert and Edwards (2008) and Perlman (2000).

Uses of Bridges

Before getting into the technology of bridges, let us take a look at some common situations in which bridges are used. We will mention three reasons why a single organization may end up with multiple LANs. First, many university and corporate departments have their own LANs to connect their own personal computers, servers, and devices such as printers. Since the goals of the various departments differ, different departments may set up different LANs, without regard to what other departments are doing. Sooner or later, though, there is a need for interaction, so bridges are needed. In this example, multiple LANs come into existence due to the autonomy of their owners. Second, the organization may be geographically spread over several buildings separated by considerable distances. It may be cheaper to have separate LANs in each building and connect them with bridges and a few long-distance fiber optic links than to run all the cables to a single central switch. Even if laying the cables is easy to do, there are limits on their lengths (e.g., 200 m for twisted-pair gigabit Ethernet). The network would not work for longer cables due to the excessive signal attenuation or round-trip delay. The only solution is to partition the LAN and install bridges to join the pieces to increase the total physical distance that can be covered. Third, it may be necessary to split what is logically a single LAN into separate LANs (connected by bridges) to accommodate the load. At many large universities, for example, thousands of workstations are available for student and faculty computing. Companies may also have thousands of employees. The scale of this system precludes putting all the workstations on a single LAN—there are more computers than ports on any Ethernet hub and more stations than allowed on a single classic Ethernet. Even if it were possible to wire all the workstations together, putting more stations on an Ethernet hub or classic Ethernet would not add capacity. All of the stations share the same, fixed amount of bandwidth.

The more stations there are, the less average bandwidth per station. However, two separate LANs have twice the capacity of a single LAN. Bridges let the LANs be joined together while keeping this capacity. The key is not to send traffic onto ports where it is not needed, so that each LAN can run at full speed. This behavior also increases reliability, since on a single LAN a defective node that keeps outputting a continuous stream of garbage can clog up the entire LAN. By deciding what to forward and what not to forward, bridges act like fire doors in a building, preventing a single node that has gone berserk from bringing down the entire system. To make these benefits easily available, ideally bridges should be completely transparent. It should be possible to go out and buy bridges, plug the LAN cables into the bridges, and have everything work perfectly, instantly.

There should be no hardware changes required, no software changes required, no setting of address switches, no downloading of routing tables or parameters, nothing at all. Just plug in the cables and walk away. Furthermore, the operation of the existing LANs should not be affected by the bridges at all. As far as the stations are concerned, there should be no observable difference whether or not they are part of a bridged LAN. It should be as easy to move stations around the bridged LAN as it is to move them around a single LAN. Surprisingly enough, it is actually possible to create bridges that are transparent. Two algorithms are used: a backward learning algorithm to stop traffic being sent where it is not needed; and a spanning tree algorithm to break loops that may be formed when switches are cabled together willy-nilly. Let us now take a Learning Bridges The topology of two LANs bridged together is shown in figure. On the left-hand side, two multidrop LANs, such as classic Ethernets, are joined by a special station—the bridge—that sits on both LANs. On the right-hand side, LANs with point-to- point cables, including one hub, are joined together. The bridges are the devices to which the stations and hub are attached. If the LAN technology is Ethernet, the bridges are better known as Ethernet switches.



(a) Bridge connecting two multidrop LANs. (b) Bridges (and a hub) connecting seven point-to-point stations

Bridges were developed when classic Ethernets were in use, so they are often shown in topologies with multidrop cables, as in Fig. 4-41(a). However, all the topologies that are encountered today are comprised of point-to-point cables and switches. The bridges work the same way in both settings. All of the stations attached to the same port on a bridge belong to the same collision domain, and this is different than the collision domain for other ports. If there is more than one station, as in a classic Ethernet, a hub, or a half-duplex link, the CSMA/CD protocol is used to send frames.

There is a difference, however, in how the bridged LANs are built. To bridge multidrop LANs, a bridge is added as a new station on each of the multidrop LANs, as in Fig. 4-41(a). To bridge point-to-point LANs, the hubs are either connected to a bridge or, preferably, replaced with a

bridge to increase performance. In Fig. bridges have replaced all but one hub. Different kinds of cables can also be attached to one bridge. For example, the cable connecting bridge *B1* to bridge *B2* in Fig. might be a long-distance fiber optic link, while the cable connecting the bridges to stations might be a short-haul twisted-pair line. This arrangement is useful for bridging LANs in different buildings.

Now let us consider what happens inside the bridges. Each bridge operates in promiscuous mode, that is, it accepts every frame transmitted by the stations attached to each of its ports. The bridge must decide whether to forward or discard each frame, and, if the former, on which port to output the frame. This decision is made by using the destination address. As an example, consider the topology of Fig. If station *A* sends a frame to station *B*, bridge *B1* will receive the frame on port This frame can be immediately discarded without further ado because it is already on the correct port. However, in the topology of Fig. suppose that *A* sends a frame to *D*. Bridge *B1* will receive the frame on port 1 and output it on port 4. Bridge *B2* will then receive the frame on its port 4 and output it on its port 1. A simple way to implement this scheme is to have a big (hash) table inside the bridge. The table can list each possible destination and which output port it belongs on. For example, in Fig, the table at *B1* would list *D* as belonging to port 4, since all *B1* has to know is which port to put frames on to reach *D*. That, in fact, more forwarding will happen later when the frame hits *B2* is not of interest to *B1*. As mentioned above, the bridges operate in promiscuous mode, so they see every frame sent on any of their ports. By looking at the source addresses, they can tell which machines are accessible on which ports. For example, if bridge *B1* in Fig sees a frame on port 3 coming from *C*, it knows that *C* must be reachable via port 3, so it makes an entry in its hash table. Any subsequent frame addressed to *C* coming in to *B1* on any other port will be forwarded to port 3.The topology can change as machines and bridges are powered up and down and moved around. To handle dynamic topologies, whenever a hash table entry is made, the arrival time of the frame is noted in the entry. Whenever a frame whose source is already in the table arrives, its entry is updated with the current time. Thus, the time associated with every entry tells the last time a frame from that machine was seen. Periodically, a process in the bridge scans the hash table and purges all entries more than a few minutes old. In this way, if a computer is unplugged from its LAN, moved around the building, and plugged in again somewhere else, within a few minutes it will be back in normal operation, without any manual intervention. This algorithm also means that if a machine is quiet for a few minutes, any traffic sent to it will have to be flooded until it next sends a frame itself. The routing procedure for an incoming frame depends on the port it arrives on (the source port) and the address to which it is destined (the destination address).

The procedure is as follows:

1. If the port for the destination address is the same as the source port, discard the frame.
2. If the port for the destination address and the source port are different, forward the frame on to the destination port.
3. If the destination port is unknown, use flooding and send the frame on all ports except the source port.

You might wonder whether the first case can occur with point-to-point links. The answer is that it can occur if hubs are used to connect a group of computers to a bridge. An example is shown in Fig. 4-41(b) where stations *E* and *F* are connected to hub *H*1, which is in turn connected to bridge *B2*. If *E* sends a frame to *F*, the hub will relay it to *B2* as well as to *F*. That is what hubs do—they wire all ports together so that a frame input on one port is simply output on all other ports. The frame will arrive at *B2* on port 4, which is already the right output port to reach the destination.

Bridge *B2* need only discard the frame. As each frame arrives, this algorithm must be applied, so it is usually implemented with special-purpose VLSI chips.
The chips do the lookup and update the table entry, all in a few microseconds. Because bridges only look at the MAC addresses to decide how to forward frames, it is possible to start forwarding as soon as the destination header field has come in, before the rest of the frame has arrived (provided the output line is available, of course). This design reduces the latency of passing through the bridge, as well as the number of frames that the bridge must be able to buffer. It is referred to as cut-through switching or wormhole routing and is usually handled in hardware.
We can look at the operation of a bridge in terms of protocol stacks to understand what it means to be a link layer device. Consider a frame sent from station *A* to station *D* in the configuration of Fig), in which the LANs are Ethernet. The frame will pass through one bridge. The protocol stack view of processing is shown in Fig .The packet comes from a higher layer and descends into the Ethernet MAC layer. It acquires an Ethernet header (and also a trailer, not shown in the figure). This unit is passed to the physical layer, goes out over the cable, and is picked up by the bridge. In the bridge, the frame is passed up from the physical layer to the Ethernet MAC layer. This layer has extended processing compared to the Ethernet MAC layer at a station. It passes the frame to a relay, still within the MAC layer. The bridge relay function uses only the Ethernet MAC header to determine how to handle the frame. In this case, it passes the frame to the Ethernet MAC layer of the port used to reach station *D*, and the frame continues on its way. In the general case, relays at a given layer can rewrite the headers for that layer. VLANs will provide an example shortly. In no case should the bridge look inside the frame and learn that it is carrying an IP packet; that is irrelevant to the bridge processing and would violate protocol layering. Also note that a bridge with *k* ports will have *k* instances of MAC and physical layers. The value of *k* is 2 for our simple example.

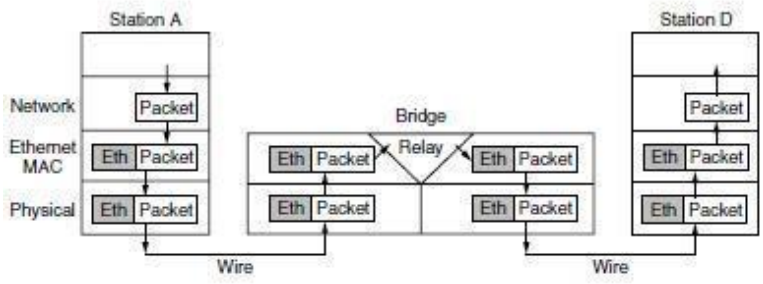
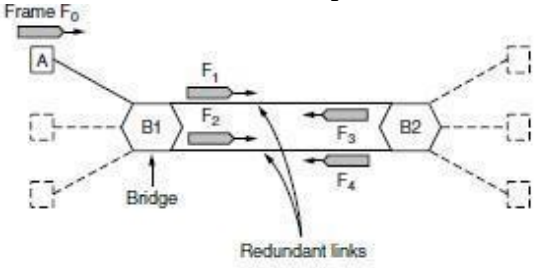


Figure 4-42. Protocol processing at a bridge.

Spanning Tree Bridges

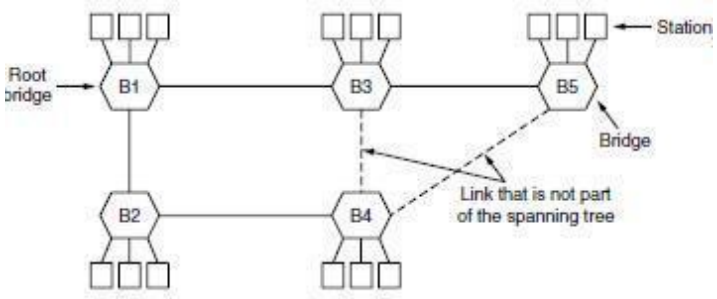
To increase reliability, redundant links can be used between bridges. In the example of Fig, there are two links in parallel between a pair of bridges. This design ensures that if one link is cut, the network will not be partitioned into two sets of computers that cannot talk to each other.



However, this redundancy introduces some additional problems because it creates loops in the topology. An example of these problems can be seen by looking at how a frame sent by *A* to a previously unobserved destination is handled in Fig.. Each bridge follows the normal rule for handling unknown destinations, which is to flood the frame. Call the frame from *A* that reaches bridge *B1* frame *F*0. The bridge sends copies of this frame out all of its other ports. We will only consider the bridge ports that connect *B1* to *B2* (though the frame will be sent out the other ports, too). Since there are two links from *B1* to *B2*, two copies of the frame will reach *B2*. They are shown in Fig as *F*1 and *F*2. Shortly thereafter, bridge *B2* receives these frames. However, it does not (and cannot) know that they are copies of the same frame, rather than two different frames sent one after the other. So bridge *B2* takes *F*1 and sends copies of it out all the other ports, and it also takes *F*2 and sends copies of it out all the other ports. This produces frames *F*3 and *F*4 that are sent along the two links back to *B1*. Bridge *B1* then sees two new frames with unknown destinations and copies them again. This cycle goes on forever. The solution to this difficulty is for the bridges to communicate with each other and overlay the actual topology with a spanning

tree that reaches every bridge. In effect, some potential connections between bridges are ignored in the interest of constructing a fictitious loop-free topology that is a subset of the actual topology.

For example, in Fig. we see five bridges that are interconnected and also have stations connected to them. Each station connects to only one bridge. There are some redundant connections between the bridges so that frames will be forwarded in loops if all of the links are used. This topology can be thought of as a graph in which the bridges are the nodes and the point-to-point links are the edges. The graph can be reduced to a spanning tree, which has no cycles by definition, by dropping the links shown as dashed lines in Fig. Using this spanning tree, there is exactly one path from every station to every other station. Once the bridges have agreed on the spanning tree, all forwarding between stations follows the spanning tree. Since there is a unique path from each source to each destination, loops are impossible.



A spanning tree connecting five bridges. The dashed lines are links that are not part of the spanning tree.

To build the spanning tree, the bridges run a distributed algorithm. Each bridge periodically broadcasts a configuration message out all of its ports to it neighbors and processes the messages it receives from other bridges, as described next. These messages are not forwarded, since their purpose is to build the tree, which can then be used for forwarding. The bridges must first choose one bridge to be the root of the spanning tree. To make this choice, they each include an identifier based on their MAC address in the configuration message, as well as the identifier of the bridge they believe to be the root. MAC addresses are installed by the manufacturer and guaranteed to be

unique worldwide, which makes these identifiers convenient and unique. The bridges choose the bridge with the lowest identifier to be the root. After enough messages have been exchanged to spread the news, all bridges will agree on which bridge is the root. In Fig., bridge *B1* has the lowest identifier and becomes the root. Next, a tree of shortest paths from the root to every bridge is constructed. In Fig. bridges *B2* and *B3* can each be reached from bridge *B1* directly, in one hop that is a shortest path. Bridge *B4* can be reached in two hops, via either *B2* or *B3*. To break this tie, the path via the bridge with the lowest identifier is chosen, so *B4* is reached via *B2*. Bridge *B5* can be reached in two hops via *B3*.

To find these shortest paths, bridges include the distance from the root in their configuration messages. Each bridge remembers the shortest path it finds to the root. The bridges then turn off ports that are not part of the shortest path. Although the tree spans all the bridges, not all the links (or even bridges) are necessarily present in the tree. This happens because turning off the ports prunes some links from the network to prevent loops. Even after the spanning tree has been established, the algorithm continues to run during normal operation to automatically detect topology changes and update the tree. The algorithm for constructing the spanning tree was invented by Radia Perlman. Her job was to solve the problem of joining LANs without loops. She was given a week to do it, but she came up with the idea for the spanning tree algorithm in a day.