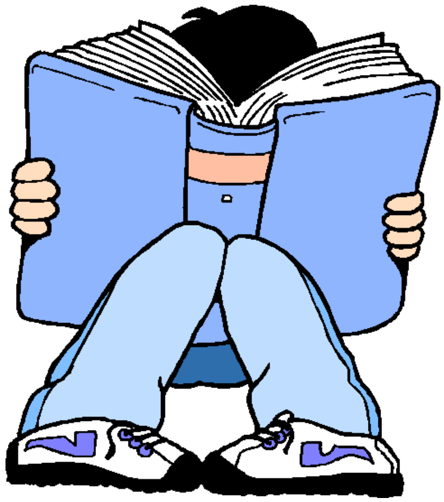


ADVANCED NETWORKING TRENDS

ADVANCED NETWORKING TRENDS (ELECTIVE - III)

RT806-5 **3+1+0**

Module 1

Ethernet Technology – Frame format – Interface Gap – CSMA/CD – 10 mbps Ethernet, Fast Ethernet, Gigabit Ethernet, Wireless Ethernet – SONET – Sonet multiplexing, Sonet frame structure

Module 2

ISDN - Definition - Protocol architecture - System architecture - Transmission channels - ISDN interface, B-ISDN.

Module 3

ATM – ATM Principles – BISDN reference model – ATM layers – ATM adaption Layer – AAL1, AAL2, AAL3/4, AAL5 – ATM addressing – UNI Signaling – PNNI Signalling

Module 4

SATELLITE COMMUNICATION: Satellite communication principles - Geo stationary satellites - block schematic of satellite earth station - VSAT - VSAT networks - applications in personnel communication. (basic ideas only)

Module 5

Wireless Lan – Infrared Vs Radio transmission – Infrastructure & ad hoc n/w – IEEE 802.11 – Hiper Law – Bluetooth – Physical Layer – MAC layer – Networking – Security

References Books

Module 1

1. An introduction to Computer Networking - Kenneth C Mansfield, Jr., James L. Antonakos, PHI

Module 1,2,3

1. Communication Networks Fundamental Concepts & Key Architecture - Leon-Garcia – Widjaja, Tata McGraw Hill
2. Mobile Communication - Jochen Schiller, Pearson Education Asia
3. Larry L Peterson and Bruce S Davie, Computer Networks - A Systems Approach, 2nd Edition, Harecourt Asia Pte. Ltd
4. Data Communications, Computer Networks & Open Systems - Fred Halsallel - Addison
   Wesley

Module 1

Ethernet Technology – Frame format – Interface Gap – CSMA/CD – 10 mbps Ethernet, Fast Ethernet, Gigabit Ethernet, Wireless Ethernet – SONET – Sonet multiplexing, Sonet frame structure.

1.1Background

The term *Ethernet* refers to the family of local-area network (LAN) products covered by the IEEE 802.3 standard that defines what is commonly known as the CSMA/CD protocol. Three data rates are currently defined for operation over optical fiber and twisted-pair cables:

- 10 Mbps—10Base-T Ethernet
- 100 Mbps—Fast Ethernet
- 1000 Mbps—Gigabit Ethernet

10-Gigabit Ethernet is under development and will likely be published as the IEEE 802.3ae supplement to the IEEE 802.3 base standard in late 2001 or early 2002.

Other technologies and protocols have been touted as likely replacements, but the market has spoken. Ethernet has survived as the major LAN technology (it is currently used for approximately 85 percent of the world's LAN-connected PCs and workstations) because its protocol has the following characteristics:

- Is easy to understand, implement, manage, and maintain
- Allows low-cost network implementations
- Provides extensive topological flexibility for network installation
- Guarantees successful interconnection and operation of standards-compliant products, regardless of manufacturer

Ethernet—A Brief History

The original Ethernet was developed as an experimental coaxial cable network in the 1970s by Xerox Corporation to operate with a data rate of 3 Mbps using a carrier sense multiple access collision detect (CSMA/CD) protocol for LANs with sporadic but occasionally heavy traffic requirements. Success with that project attracted early attention and led to the 1980 joint development of the 10-Mbps Ethernet Version 1.0 specification by the three-company consortium: Digital Equipment Corporation, Intel Corporation, and Xerox Corporation.

The original IEEE 802.3 standard was based on, and was very similar to, the Ethernet Version 1.0 specification. The draft standard was approved by the 802.3 working group in 1983 and was subsequently published as an official standard in 1985 (ANSI/IEEE Std. 802.3-1985). Since then, a number of supplements to the standard have been defined to take advantage of improvements in the technologies and to support additional network media and higher data rate capabilities, plus several new optional network access control features.

Throughout the rest of this chapter, the terms Ethernet and 802.3 will refer exclusively to network implementations compatible with the IEEE 802.3 standard.

Ethernet Network Elements

Ethernet LANs consist of network nodes and interconnecting media. The network nodes fall into two major classes:

- Data terminal equipment (DTE)—Devices that are either the source or the destination of data frames. DTEs are typically devices such as PCs, workstations, file servers, or print servers that, as a group, are all often referred to as end stations.
- Data communication equipment (DCE)—Intermediate network devices that receive and forward frames across the network. DCEs may be either standalone devices such as repeaters, network switches, and routers, or communications interface units such as interface cards and modems.

Throughout this chapter, standalone intermediate network devices will be referred to as either intermediate nodes or DCEs. Network interface cards will be referred to as NICs.

The current Ethernet media options include two general types of copper cable: unshielded twisted-pair (UTP) and shielded twisted-pair (STP), plus several types of optical fiber cable.

Ethernet Network Topologies and Structures

LANs take on many topological configurations, but regardless of their size or complexity, all will be a combination of only three basic interconnection structures or network building blocks.

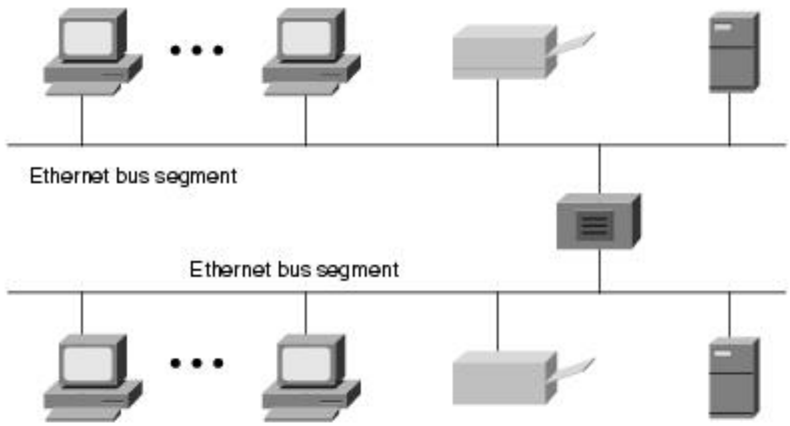
The simplest structure is the point-to-point interconnection, shown in Figure 7-1. Only two network units are involved, and the connection may be DTE-to-DTE, DTE-to-DCE, or DCE-to-DCE. The cable in point-to-point interconnections is known as a network link. The maximum allowable length of the link depends on the type of cable and the transmission method that is used.

Example Point-to-Point Interconnection

n

The original Ethernet networks were implemented with a coaxial bus structure, as shown in Figure Segment lengths were limited to 500 meters, and up to 100 stations could be connected to a single segment. Individual segments could be interconnected with repeaters, as long as multiple paths did not exist between any two stations on the network and the number of DTEs did not exceed 1024. The total path distance between the most-distant pair of stations was also not allowed to exceed a maximum prescribed value.

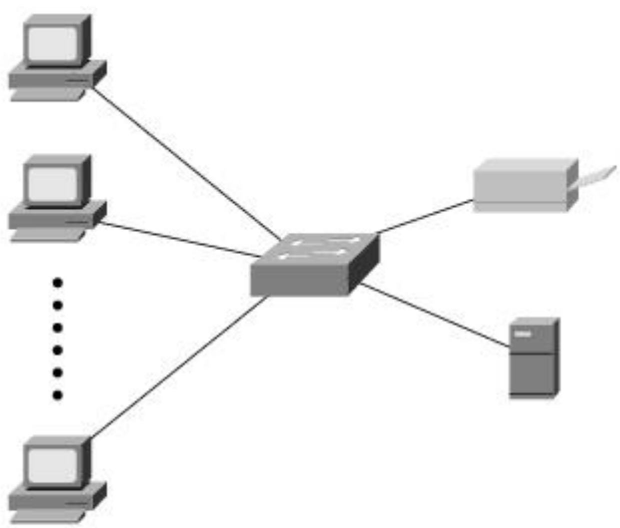
Example Coaxial Bus Topology



Although new networks are no longer connected in a bus configuration, some older bus-connected networks do still exist and are still useful.

Since the early 1990s, the network configuration of choice has been the star-connected topology. The central network unit is either a multiport repeater (also known as a hub) or a network switch. All connections in a star network are point-to-point links implemented with either twisted-pair or optical fiber cable.

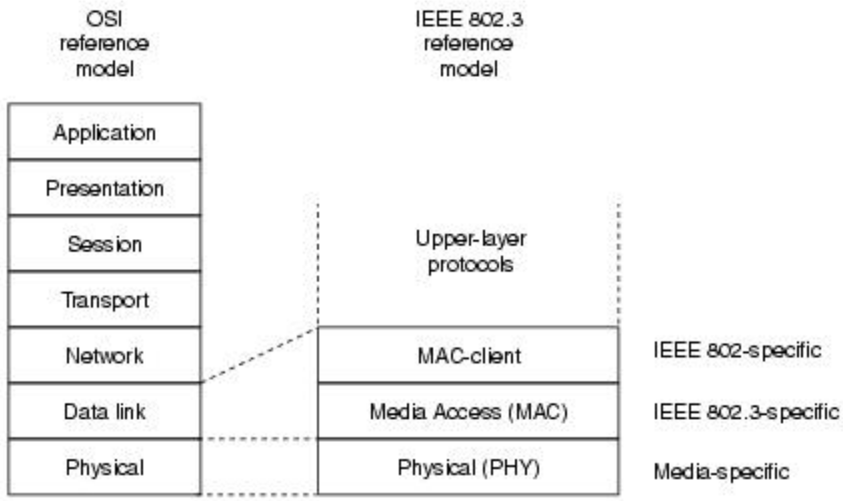
Example Star-Connected Topology



The IEEE 802.3 Logical Relationship to the ISO Reference Model

Figure the IEEE 802.3 logical layers and their relationship to the OSI reference model. As with all IEEE 802 protocols, the ISO data link layer is divided into two IEEE 802 sublayers, the Media Access Control (MAC) sublayer and the MAC-client sublayer. The IEEE 802.3 physical layer corresponds to the ISO physical layer.

Ethernet's Logical Relationship to the ISO Reference Model

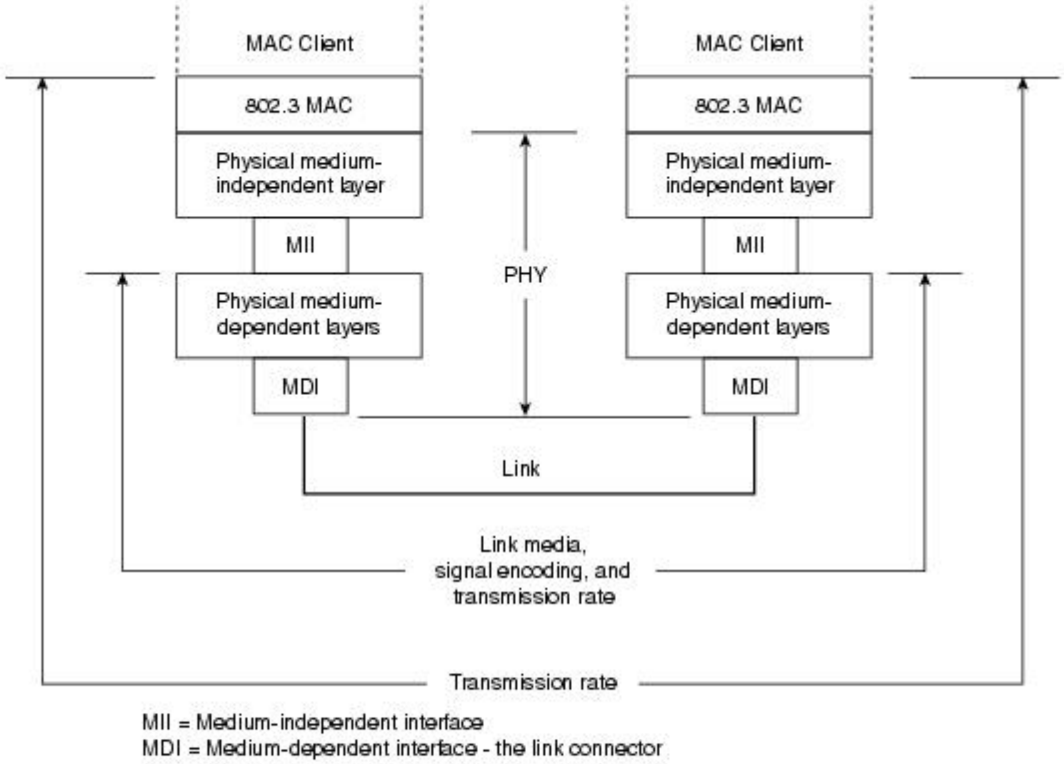


The MAC-client sublayer may be one of the following:

- Logical Link Control (LLC), if the unit is a DTE. This sublayer provides the interface between the Ethernet MAC and the upper layers in the protocol stack of the end station. The LLC sublayer is defined by IEEE 802.2 standards.
- Bridge entity, if the unit is a DCE. Bridge entities provide LAN-to-LAN interfaces between LANs that use the same protocol (for example, Ethernet to Ethernet) and also between different protocols (for example, Ethernet to Token Ring). Bridge entities are defined by IEEE 802.1 standards.

Because specifications for LLC and bridge entities are common for all IEEE 802 LAN protocols, network compatibility becomes the primary responsibility of the particular network protocol. Figure shows different compatibility requirements imposed by the MAC and physical levels for basic data communication over an Ethernet link.

MAC and Physical Layer Compatibility Requirements for Basic Data Communication



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The MAC layer controls the node's access to the network media and is specific to the individual protocol. All IEEE 802.3 MACs must meet the same basic set of logical requirements, regardless of whether they include one or more of the defined optional protocol extensions. The only requirement for basic communication (communication that does not require optional protocol extensions) between two network nodes is that both MACs must support the same transmission rate.

The 802.3 physical layer is specific to the transmission data rate, the signal encoding, and the type of media interconnecting the two nodes. Gigabit Ethernet, for example, is defined to operate over either twisted-pair or optical fiber cable, but each specific type of cable or signal-encoding procedure requires a different physical layer implementation.

The Ethernet MAC Sublayer

The MAC sublayer has two primary responsibilities:

- Data encapsulation, including frame assembly before transmission, and frame parsing/error detection during and after reception
- Media access control, including initiation of frame transmission and recovery from transmission failure

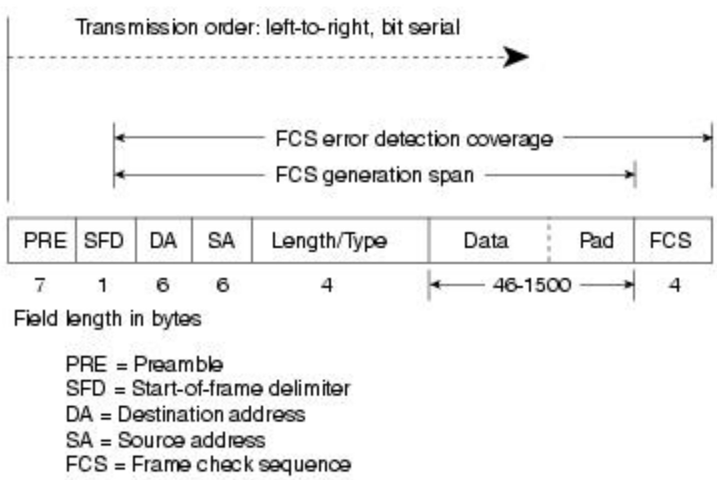
The Basic Ethernet Frame Format

The IEEE 802.3 standard defines a basic data frame format that is required for all MAC implementations, plus several additional optional formats that are used to extend the protocol's basic capability. The basic data frame format contains the seven fields shown in Figure 7-6.

- Preamble (PRE)—Consists of 7 bytes. The PRE is an alternating pattern of ones and zeros that tells receiving stations that a frame is coming, and that provides a means to synchronize the frame-reception portions of receiving physical layers with the incoming bit stream.
- Start-of-frame delimiter (SOF)—Consists of 1 byte. The SOF is an alternating pattern of ones and zeros, ending with two consecutive 1-bits indicating that the next bit is the left-most bit in the left-most byte of the destination address.
- Destination address (DA)—Consists of 6 bytes. The DA field identifies which station(s) should receive the frame. The left-most bit in the DA field indicates whether the address is an individual address (indicated by a 0) or a group address (indicated by a 1). The second bit from the left indicates whether the DA is globally administered (indicated by a 0) or locally administered (indicated by a 1). The remaining 46 bits are a uniquely assigned value that identifies a single station, a defined group of stations, or all stations on the network.
- Source addresses (SA)—Consists of 6 bytes. The SA field identifies the sending station. The SA is always an individual address and the left-most bit in the SA field is always 0.

- Length/Type—Consists of 2 bytes. This field indicates either the number of MAC-client data bytes that are contained in the data field of the frame, or the frame type ID if the frame is assembled using an optional format. If the Length/Type field value is less than or equal to 1500, the number of LLC bytes in the Data field is equal to the Length/Type field value. If the Length/Type field value is greater than 1536, the frame is an optional type frame, and the Length/Type field value identifies the particular type of frame being sent or received.
- Data—Is a sequence of n bytes of any value, where n is less than or equal to 1500. If the length of the Data field is less than 46, the Data field must be extended by adding a filler (a pad) sufficient to bring the Data field length to 46 bytes.
- Frame check sequence (FCS)—Consists of 4 bytes. This sequence contains a 32-bit cyclic redundancy check (CRC) value, which is created by the sending MAC and is recalculated by the receiving MAC to check for damaged frames. The FCS is generated over the DA, SA, Length/Type, and Data fields.

The Basic IEEE 802.3 MAC Data Frame Format



Frame Transmission

Whenever an end station MAC receives a transmit-frame request with the accompanying address and data information from the LLC sublayer, the MAC begins the transmission sequence by transferring the LLC information into the MAC frame buffer.

- The preamble and start-of-frame delimiter are inserted in the PRE and SOF fields.

- The destination and source addresses are inserted into the address fields.
- The LLC data bytes are counted, and the number of bytes is inserted into the Length/Type field.
- The LLC data bytes are inserted into the Data field. If the number of LLC data bytes is less than 46, a pad is added to bring the Data field length up to 46.
- An FCS value is generated over the DA, SA, Length/Type, and Data fields and is appended to the end of the Data field.

After the frame is assembled, actual frame transmission will depend on whether the MAC is operating in half-duplex or full-duplex mode.

The IEEE 802.3 standard currently requires that all Ethernet MACs support half-duplex operation, in which the MAC can be either transmitting or receiving a frame, but it cannot be doing both simultaneously. Full-duplex operation is an optional MAC capability that allows the MAC to transmit and receive frames simultaneously.

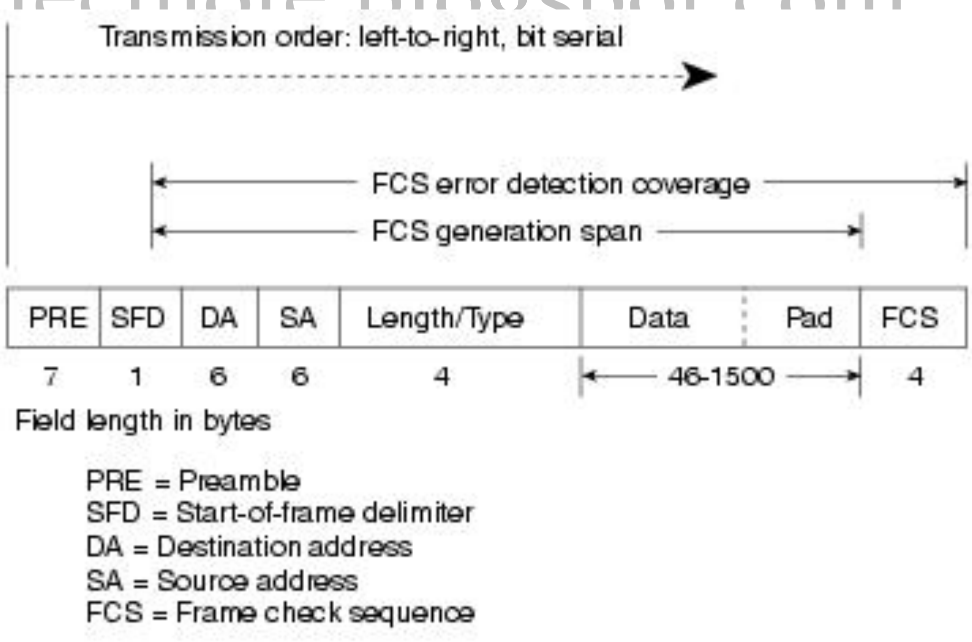
1.2The Basic Ethernet Frame Format

The IEEE 802.3 standard defines a basic data frame format that is required for all MAC implementations, plus several additional optional formats that are used to extend the protocol's basic capability. The basic data frame format contains the seven fields shown in Figure 7-6.

- Preamble (PRE)—Consists of 7 bytes. The PRE is an alternating pattern of ones and zeros that tells receiving stations that a frame is coming, and that provides a means to synchronize the frame-reception portions of receiving physical layers with the incoming bit stream.
- Start Frame Delimiter (SFD)— Consists of 1 byte. The SOF is an alternating pattern of ones and zeros, ending with two consecutive 1-bits indicating that the next bit is the left-most bit in the left-most byte of the destination address.
- Destination Address (DA)—consists of 6 bytes. The DA field identifies which station(s) should receive the frame. Every ETHERNET device manufactured contains a unique 48-bit MAC address assigned by the manufacturer. An example of MAC

address is 00-C0-F0-27-64-E2.The first 24-bit are the manufacturer code.(00-C0-F0 is Kingston) .The last 24-bits are chosen by the manufacturer.

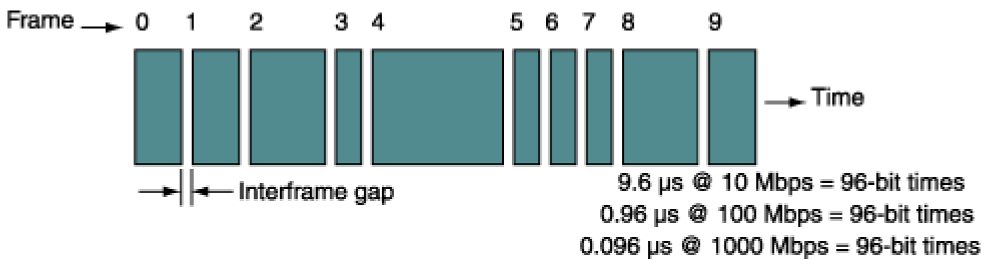
- Source Addresses (SA)—consists of 6 bytes. The SA field identifies the sending station. The SA is always an individual address and the left-most bit in the SA field is always 0.
- Length/Type—consists of 2 bytes. This field indicates no of bytes in the data field.
- Data—Is a sequence of n bytes of any value, where n is less than or equal to 1500. If the length of the Data field is less than 46, the Data field must be extended by adding a filler (a pad) sufficient to bring the Data field length to 46 bytes.
- Frame check sequence (FCS)—Consists of 4 bytes. This sequence contains a 32-bit cyclic redundancy check (CRC) value, which is created by the sending MAC and is recalculated by the receiving MAC to check for damaged frames. The FCS is generated over the DA, SA, Length/Type, and Data fields.



1.3THE INERFRAME GAP

The interframe gap is a self-imposed quiet time appended to the end of every frame. This idle time gives the network media a chance to stabilize, and other network components

time to process the frame.For 10-Mbps Ethernet ,the interframe gap is 9.6 micro-seconds.This
corresponds to 96-bit times



1.4Half-Duplex Transmission—The CSMA/CD Access Method

The CSMA/CD protocol was originally developed as a means by which two or more stations could share a common media in a switch-less environment when the protocol does not require central arbitration, access tokens, or assigned time slots to indicate when a station will be allowed to transmit. Each Ethernet MAC determines for itself when it will be allowed to send a frame.

The CSMA/CD access rules are summarized by the protocol's acronym:

- Carrier sense—Each station continuously listens for traffic on the medium to determine when gaps between frame transmissions occur.
- Multiple access—Stations may begin transmitting any time they detect that the network is quiet (there is no traffic).
- Collision detect—If two or more stations in the same CSMA/CD network (collision domain) begin transmitting at approximately the same time, the bit streams from the transmitting stations will interfere (collide) with each other, and both transmissions will be unreadable. If that happens, each transmitting station must be capable of detecting that a collision has occurred before it has finished sending its frame.
  Each must stop transmitting as soon as it has detected the collision and then must wait a quasirandom length of time (determined by a back-off algorithm) before attempting to retransmit the frame.

The worst-case situation occurs when the two most-distant stations on the network both need to send a frame and when the second station does not begin transmitting until just before the frame from the first station arrives. The collision will be detected almost immediately by the second station, but it will not be detected by the first station until the corrupted signal has propagated all the way back to that station. The maximum time that is required to detect a collision (the collision window, or "slot time") is approximately equal to twice the signal propagation time between the two most-distant stations on the network.

This means that both the minimum frame length and the maximum collision diameter are directly related to the slot time. Longer minimum frame lengths translate to longer slot times and larger collision diameters; shorter minimum frame lengths correspond to shorter slot times and smaller collision diameters.

The trade-off was between the need to reduce the impact of collision recovery and the need for network diameters to be large enough to accommodate reasonable network sizes. The compromise was to choose a maximum network diameter (about 2500 meters) and then to set the minimum frame length long enough to ensure detection of all worst-case collisions.

The compromise worked well for 10 Mbps, but it was a problem for higher data-rate Ethernet developers. Fast Ethernet was required to provide backward compatibility with earlier Ethernet networks, including the existing IEEE 802.3 frame format and error-detection procedures, plus all applications and networking software running on the 10-Mbps networks.

Although signal propagation velocity is essentially constant for all transmission rates, the time required to transmit a frame is inversely related to the transmission rate. At 100 Mbps, a minimum-length frame can be transmitted in approximately one-tenth of the defined slot time, and any collision that occurred during the transmission would not likely be detected by the transmitting stations. This, in turn, meant that the maximum network diameters specified for 10-Mbps networks could not be used for 100-Mbps networks. The solution for Fast Ethernet was to reduce the maximum network diameter by approximately a factor of 10 (to a little more than 200 meters).

The same problem also arose during specification development for Gigabit Ethernet, but decreasing network diameters by another factor of 10 (to approximately 20 meters) for 1000-Mbps operation was simply not practical. This time, the developers elected to maintain approximately the same maximum collision domain diameters as 100-Mbps networks and to increase the apparent minimum frame size by adding a variable-length nondata extension field to frames that are shorter than the minimum length (the extension field is removed during frame reception).

Figure 7-7 shows the MAC frame format with the gigabit extension field, and Table 7-1 shows the effect of the trade-off between the transmission data rate and the minimum frame size for 10-Mbps, 100-Mbps, and 1000-Mbps Ethernet.

Figure 7-7 MAC Frame with Gigabit Carrier Extension

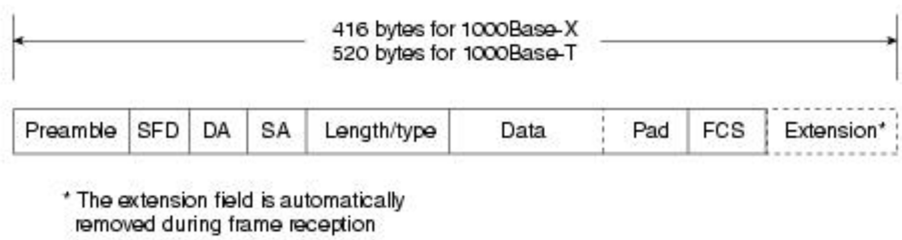


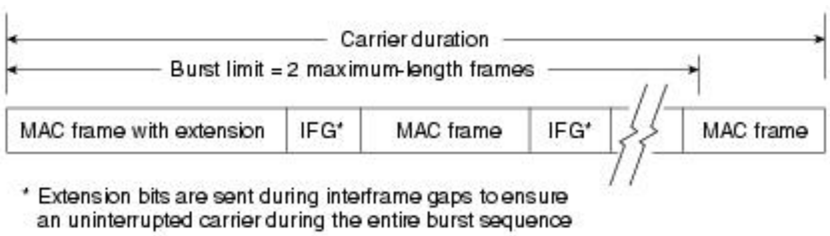
Table 7-1 Limits for Half-Duplex Operation			
Parameter	10 Mbps	100 Mbps	1000 Mbps
Minimum frame size	64 bytes	64 bytes	520 bytes[1] (with extension field added)
Maximum collision diameter, DTE to DTE	100 meters UTP	100 meters UTP 412 meters	100 meters UTP 316 meters fiber

		fiber	
Maximum collision diameter with repeaters	2500 meters	205 meters	200 meters
Maximum number of repeaters in network path	5	2	1

[1] 520 bytes applies to 1000Base-T implementations. The minimum frame size with extension field for 1000Base-X is reduced to 416 bytes because 1000Base-X encodes and transmits 10 bits for each byte.

Another change to the Ethernet CSMA/CD transmit specification was the addition of frame bursting for gigabit operation. Burst mode is a feature that allows a MAC to send a short sequence (a burst) of frames equal to approximately 5.4 maximum-length frames without having to relinquish control of the medium. The transmitting MAC fills each interframe interval with extension bits, as shown in Figure 7-8, so that other stations on the network will see that the network is busy and will not attempt transmission until after the burst is complete.

Figure 7-8 A Gigabit Frame-Burst Sequence



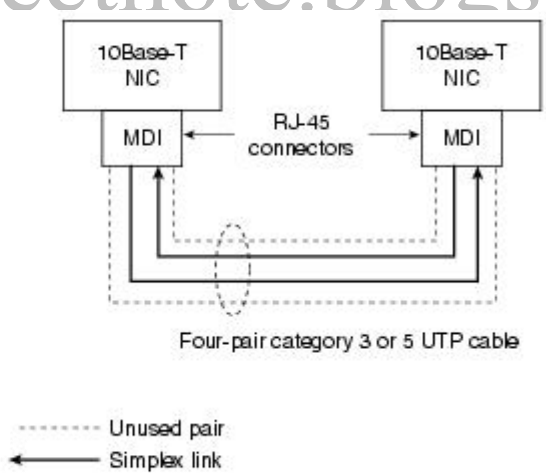
If the length of the first frame is less than the minimum frame length, an extension field is added to extend the frame length to the value indicated in Table 7-1. Subsequent frames in a frame-burst sequence do not need extension fields, and a frame burst may continue as long as the burst limit has not been reached. If the burst limit is reached after a frame transmission has begun, transmission is allowed to continue until that entire frame has been sent.

Frame extension fields are not defined, and burst mode is not allowed for 10 Mbps and 100 Mbps transmission rates.

10.5 10-Mbps Ethernet—10Base-T

10Base-T provides Manchester-encoded 10-Mbps bit-serial communication over two unshielded twisted-pair cables. Although the standard was designed to support transmission over common telephone cable, the more typical link configuration is to use two pair of a four-pair Category 3 or 5 cable, terminated at each NIC with an 8-pin RJ-45 connector (the MDI), as shown in Figure 7-15. Because each active pair is configured as a simplex link where transmission is in one direction only, the 10Base-T physical layers can support either half-duplex or full-duplex operation.

Figure 7-15 The Typical 10Base-T Link Is a Four-Pair UTP Cable in Which Two Pairs Are Not Used



Although 10Base-T may be considered essentially obsolete in some circles, it is included here because there are still many 10Base-T Ethernet networks, and because full-duplex operation has given 10BaseT an extended life.

10Base-T was also the first Ethernet version to include a link integrity test to determine the health of the link. Immediately after powerup, the PMA transmits a normal link pulse

(NLP) to tell the NIC at the other end of the link that this NIC wants to establish an active link connection:

- If the NIC at the other end of the link is also powered up, it responds with its own NLP.
- If the NIC at the other end of the link is not powered up, this NIC continues sending an NLP about once every 16 ms until it receives a response.

The link is activated only after both NICs are capable of exchanging valid NLPs.

1.6 100 Mbps—Fast Ethernet

Increasing the Ethernet transmission rate by a factor of ten over 10Base-T was not a simple task, and the effort resulted in the development of three separate physical layer standards for 100 Mbps over UTP cable: 100Base-TX and 100Base-T4 in 1995, and 100Base-T2 in 1997. Each was defined with different encoding requirements and a different set of media-dependent sublayers, even though there is some overlap in the link cabling. Table 7-2 compares the physical layer characteristics of 10Base-T to the various 100Base versions.

Summary of 100Base-T Physical Layer Characteristics				
Ethernet Version	Transmit Symbol Rate[1]	Encoding	Cabling	Full-Duplex Operation
10Base-T	10 MBd	Manchester	Two pairs of UTP Category -3 or better	Supported
100Base-TX	125 MBd	4B/5B	Two pairs of UTP Category -5 or Type 1 STP	Supported

100Base-T4	33 MBd	8B/6T	Four pairs of UTP Category -3 or better	Not supported
100Base-T2	25 MBd	PAM5x5	Two pairs of UTP Category -3 or better	Supported

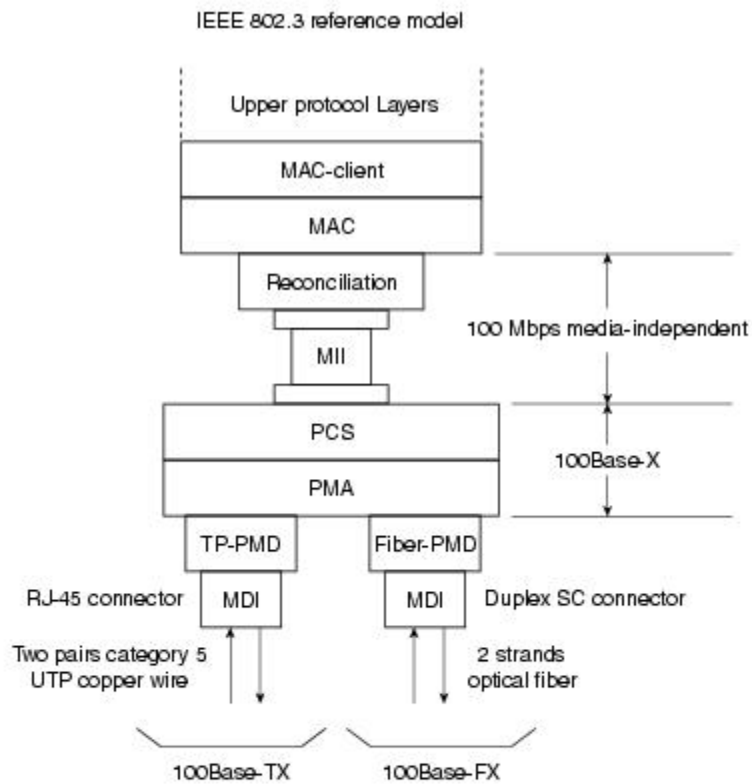
[1] One baud = one transmitted symbol per second, where the transmitted symbol may contain the equivalent value of 1 or more binary bits.

Although not all three 100-Mbps versions were successful in the marketplace, all three have been discussed in the literature, and all three did impact future designs. As such, all three are important to consider here.

100Base-X

100Base-X was designed to support transmission over either two pairs of Category 5 UTP copper wire or two strands of optical fiber. Although the encoding, decoding, and clock recovery procedures are the same for both media, the signal transmission is different—electrical pulses in copper and light pulses in optical fiber. The signal transceivers that were included as part of the PMA function in the generic logical model of Figure 7-14 were redefined as the separate physical media-dependent (PMD) sublayers shown in Figure.

Figure The 100Base-X Logical Model



The 100Base-X encoding procedure is based on the earlier FDDI optical fiber physical media-dependent and FDDI/CDDI copper twisted-pair physical media-dependent signaling standards developed by ISO and ANSI. The 100Base-TX physical media-dependent sublayer (TP-PMD) was implemented with CDDI semiconductor transceivers and RJ-45 connectors; the fiber PMD was implemented with FDDI optical transceivers and the Low Cost Fibre Interface Connector (commonly called the duplex SC connector).

The 4B/5B encoding procedure is the same as the encoding procedure used by FDDI, with only minor adaptations to accommodate Ethernet frame control. Each 4-bit data nibble (representing half of a data byte) is mapped into a 5-bit binary code-group that is transmitted bit-serial over the link. The expanded code space provided by the 32 5-bit code-groups allow separate assignment for the following:

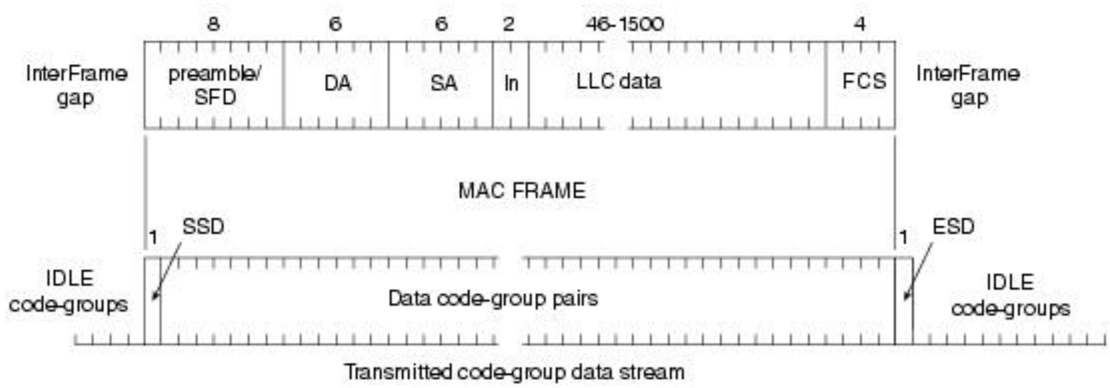
- The 16 possible values in a 4-bit data nibble (16 code-groups).
- Four control code-groups that are transmitted as code-group pairs to indicate the start-of-stream delimiter (SSD) and the end-of-stream delimiter (ESD). Each MAC frame is "encapsulated" to mark both the beginning and end of the frame. The first byte of

preamble is replaced with SSD code-group pair that precisely identifies the frame's code-group boundaries. The ESD code-group pair is appended after the frame's FCS field.

- A special IDLE code-group that is continuously sent during interframe gaps to maintain continuous synchronization between the NICs at each end of the link. The receipt of IDLE is interpreted to mean that the link is quiet.
- Eleven invalid code-groups that are not intentionally transmitted by a NIC (although one is used by a repeater to propagate receive errors). Receipt of any invalid code-group will cause the incoming frame to be treated as an invalid frame.

Figure shows how a MAC frame is encapsulated before being transmitted as a 100Base-X code-group stream.

Figure The 100Base-X Code-Group Stream with Frame Encapsulation

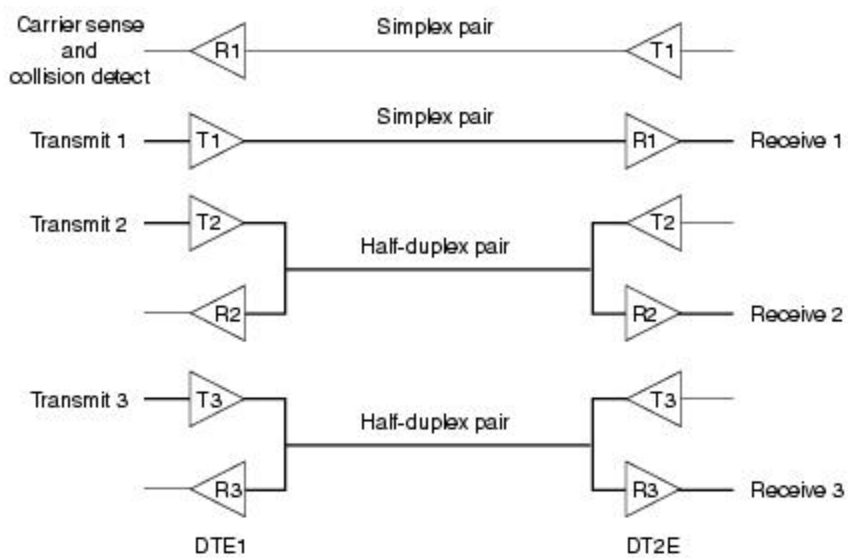


100Base-TX transmits and receives on the same link pairs and uses the same pin assignments on the MDI as 10Base-T. 100Base-TX and 100Base-FX both support half-duplex and full-duplex transmission.

100Base-T4

100Base-T4 was developed to allow 10BaseT networks to be upgraded to 100-Mbps operation without requiring existing four-pair Category 3 UTP cables to be replaced with the newer Category 5 cables. Two of the four pairs are configured for half-duplex operation and can support transmission in either direction, but only in one direction at a time. The other two pairs are configured as simplex pairs dedicated to transmission in one direction only. Frame transmission uses both half-duplex pairs, plus the simplex pair that is appropriate for the transmission direction, as shown in Figure 7-18. The simplex pair for the opposite direction provides carrier sense and collision detection. Full-duplex operation cannot be supported on 100Base-T4.

The 100Base-T4 Wire-Pair Usage During Frame Transmission

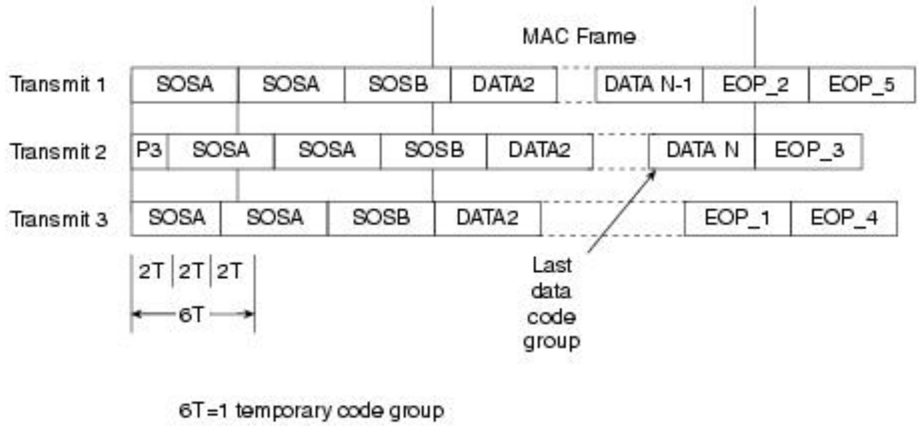


Transmission paths for DTE1 shown in bold

100Base-T4 uses an 8B6T encoding scheme in which each 8-bit binary byte is mapped into a pattern of six ternary (three-level: +1, 0, -1) symbols known as 6T code-groups. Separate 6T code-groups are used for IDLE and for the control code-groups that are necessary for frame transmission. IDLE received on the dedicated receive pair indicates that the link is quiet.

During frame transmission, 6T data code-groups are transmitted in a delayed round-robin sequence over the three transmit wire-pairs, as shown in Figure 7-19. Each frame is encapsulated with start-of-stream and end-of-packet 6T code-groups that mark both the beginning and end of the frame, and the beginning and end of the 6T code-group stream on each wire pair. Receipt of a non-IDLE code-group over the dedicated receive-pair any time before the collision window expires indicates that a collision has occurred.

Figure The 100Base-T4 Frame Transmission Sequence



100Base-T2

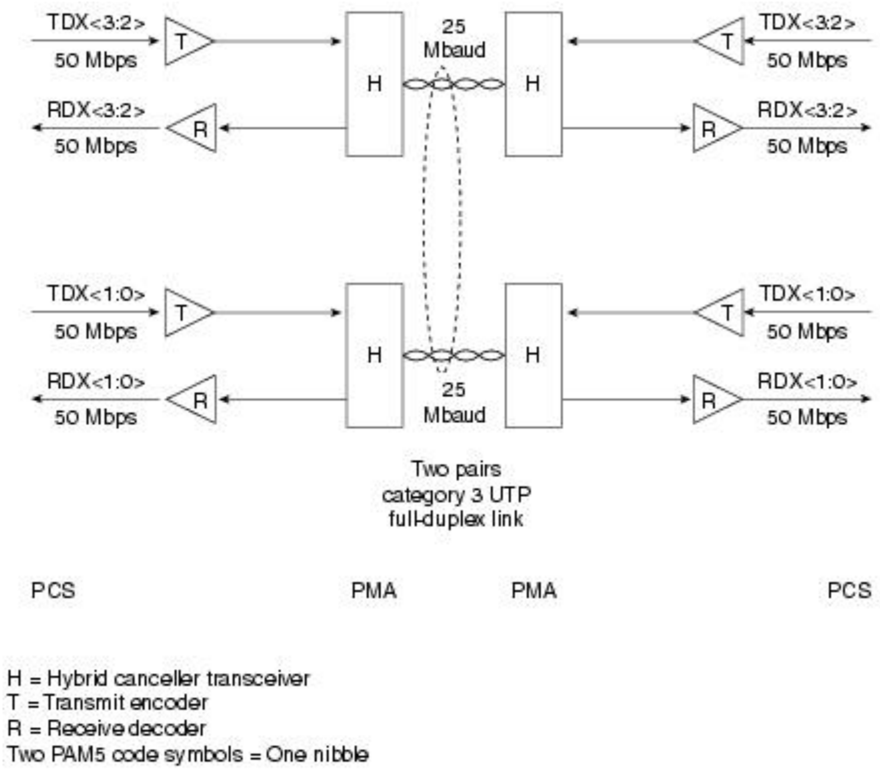
The 100Base-T2 specification was developed as a better alternative for upgrading networks with installed Category 3 cabling than was being provided by 100Base-T4. Two important new goals were defined:

- To provide communication over two pairs of Category 3 or better cable

- To support both half-duplex and full-duplex operation

100Base-T2 uses a different signal transmission procedure than any previous twisted-pair Ethernet implementations. Instead of using two simplex links to form one full-duplex link, the 100Base-T2 dual-duplex baseband transmission method sends encoded symbols simultaneously in both directions on both wire pairs, as shown in Figure 7-20. The term "TDX<3:2>" indicates the 2 most significant bits in the nibble before encoding and transmission. "RDX<3:2>" indicates the same 2 bits after receipt and decoding.

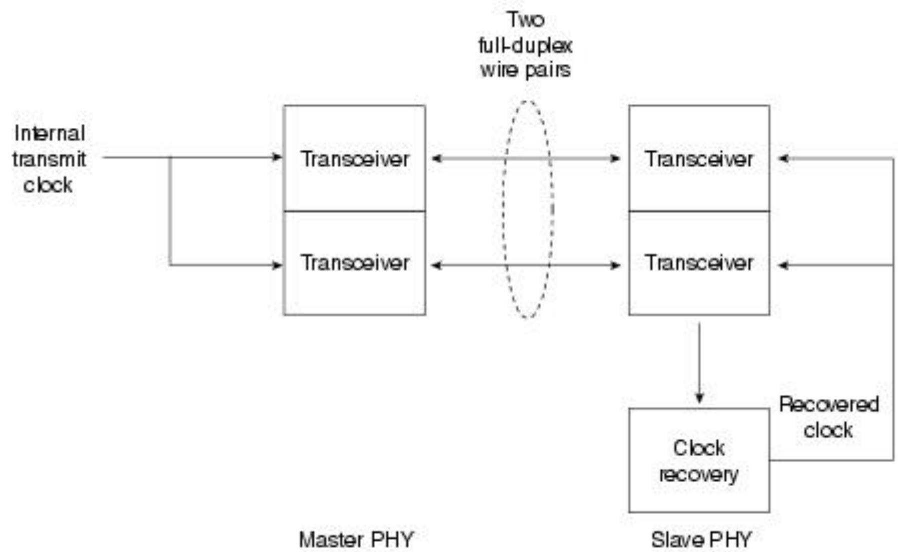
Figure The 100Base-T2 Link Topology



Dual-duplex baseband transmission requires the NICs at each end of the link to be operated in a master/slave loop-timing mode. Which NIC will be master and which will be slave

is determined by autonegotiation during link initiation. When the link is operational, synchronization is based on the master NIC's internal transmit clock. The slave NIC uses the recovered clock for both transmit and receive operations, as shown in Figure 7-21. Each transmitted frame is encapsulated, and link synchronization is maintained with a continuous stream of IDLE symbols during interframe gaps.

Figure The 100Base-T2 Loop Timing Configuration



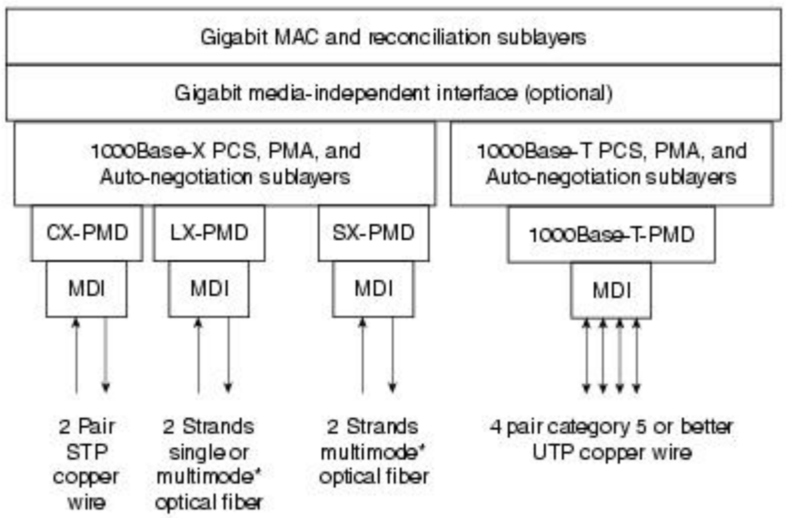
The 100Base-T2 encoding process first scrambles the data frame nibbles to randomize the bit sequence. It then maps the two upper bits and the two lower bits of each nibble into two five-level (+2, +1, 0, -1, -2) pulse amplitude-modulated (PAM5) symbols that are simultaneously transmitted over the two wire pairs (PAM5x5). Different scrambling procedures for master and slave transmissions ensure that the data streams traveling in opposite directions on the same wire pair are uncoordinated.

Signal reception is essentially the reverse of signal transmission. Because the signal on each wire pair at the MDI is the sum of the transmitted signal and the received signal, each receiver subtracts the transmitted symbols from the signal received at the MDI to recover the symbols in the incoming data stream. The incoming symbol pair is then decoded, unscrambled, and reconstituted as a data nibble for transfer to the MAC.

1.7 1000 Mbps—Gigabit Ethernet

The Gigabit Ethernet standards development resulted in two primary specifications: 1000Base-T for UTP copper cable and 1000Base-X STP copper cable, as well as single and multimode optical fiber .

Figure Gigabit Ethernet Variations



1000Base-T

1000Base-T Ethernet provides full-duplex transmission over four-pair Category 5 or better UTP cable. 1000Base-T is based largely on the findings and design approaches that led to the development of the Fast Ethernet physical layer implementations:

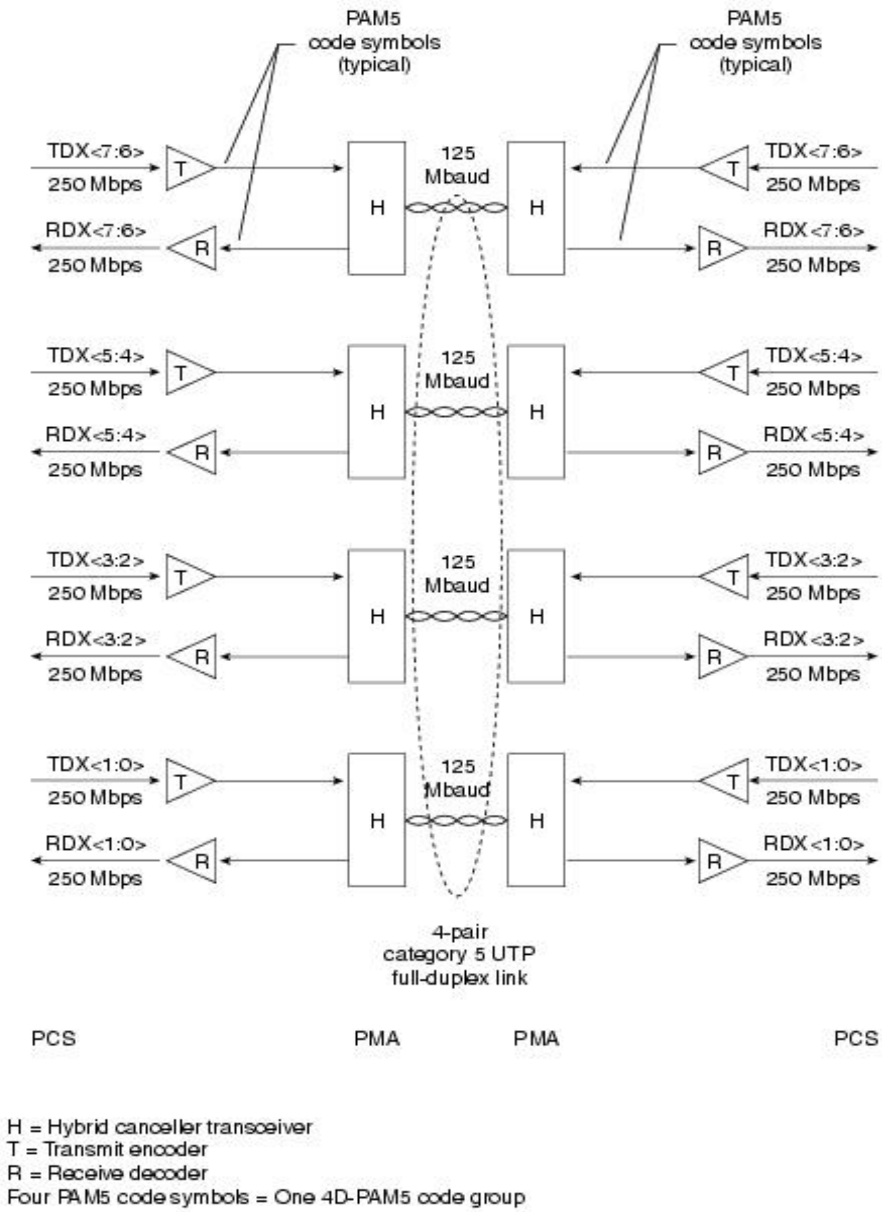
- 100Base-TX proved that binary symbol streams could be successfully transmitted over Category 5 UTP cable at 125 MBd.
- 100Base-T4 provided a basic understanding of the problems related to sending multilevel signals over four wire pairs.
- 100Base-T2 proved that PAM5 encoding, coupled with digital signal processing, could handle both simultaneous two-way data streams and potential crosstalk problems resulting from alien signals on adjacent wire pairs.

1000Base-T scrambles each byte in the MAC frame to randomize the bit sequence before it is encoded using a 4-D, 8-State Trellis Forward Error Correction (FEC) coding in which four PAM5 symbols are sent at the same time over four wire pairs. Four of the five levels in each PAM5 symbol represent 2 bits in the data byte. The fifth level is used for FEC coding, which enhances symbol recovery in the presence of noise and crosstalk. Separate scramblers for the master and slave PHYs create essentially uncorrelated data streams between the two opposite-travelling symbol streams on each wire pair.

The1000Base-T link topology is shown in Figure 7-23. The term "TDX<7:6>" indicates the 2 most significant bits in the data byte before encoding and transmission. "RDX<7:6>" indicates the same 2 bits after receipt and decoding.

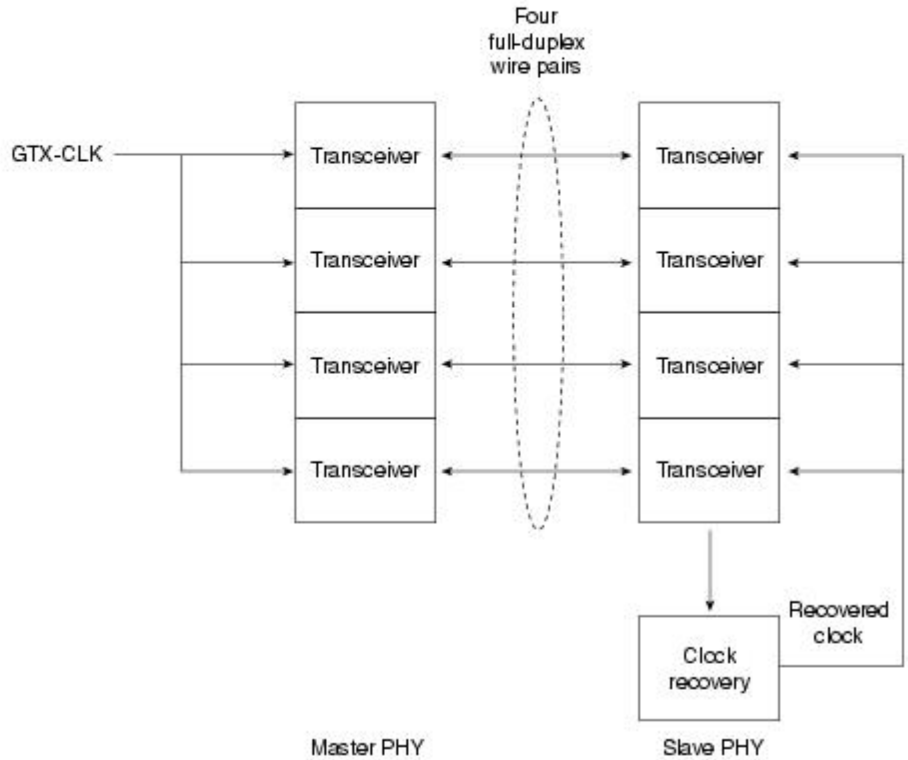
FigureThe 1000Base-T Link

Topology



The clock recovery and master/slave loop timing procedures are essentially the same as those used in 100Base-T2 (see Figure). Which NIC will be master (typically the NIC in a multiport intermediate network node) and which will be slave is determined during autonegotiation.

Figure 1000Base-T Master/Slave Loop Timing Configuration

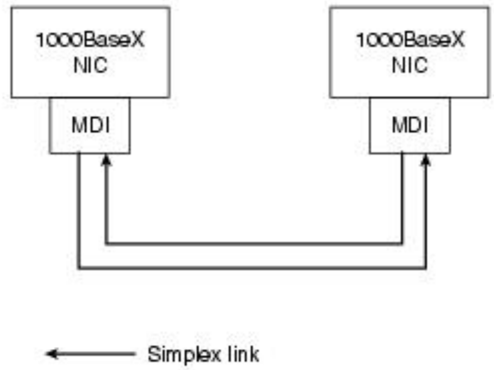


Each transmitted frame is encapsulated with start-of-stream and end-of-stream delimiters, and loop timing is maintained by continuous streams of IDLE symbols sent on each wire pair during interframe gaps. 1000Base-T supports both half-duplex and full-duplex operation.

1000Base-X

All three 1000Base-X versions support full-duplex binary transmission at 1250 Mbps over two strands of optical fiber or two STP copper wire-pairs, as shown in Figure 7-25. Transmission coding is based on the ANSI Fibre Channel 8B/10B encoding scheme. Each 8-bit data byte is mapped into a 10-bit code-group for bit-serial transmission. Like earlier Ethernet versions, each data frame is encapsulated at the physical layer before transmission, and link synchronization is maintained by sending a continuous stream of IDLE code-groups during interframe gaps. All 1000Base-X physical layers support both half-duplex and full-duplex operation.

Figure 1000Base-X Link Configuration



The principal differences among the 1000Base-X versions are the link media and connectors that the particular versions will support and, in the case of optical media, the wavelength of the optical signal (see Table).

Table 1000Base-X Link Configuration Support

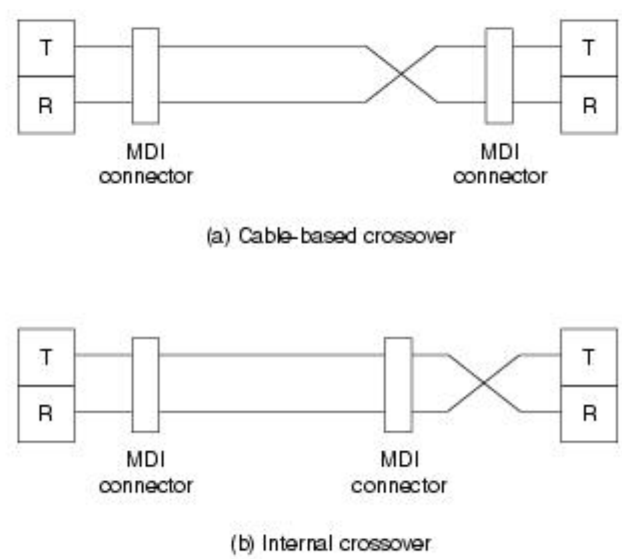
Link Configuration	**1000Base-CX**	**1000Base-SX (850 nm Wavelength)**	**1000Base-LX (1300 nm Wavelength)**
150 Ω STP copper	Supported	Not supported	Not supported
125/62.5 µm multimode optical fiber[1]	Not supported	Supported	Supported
125/50 µm multimode optical fiber	Not supported	Supported	Supported
125/10 µm single mode optical fiber	Not supported	Not supported	Supported
Allowed connectors	IEC style 1 or Fibre Channel style 2	SFF MT-RJ or Duplex SC	SFF MT-RJ or Duplex SC

1.8 Network Cabling—Link Crossover Requirements

Link compatibility requires that the transmitters at each end of the link be connected to the receivers at the other end of the link. However, because cable connectors at both ends of the link are keyed the same, the conductors must cross over at some point to ensure that transmitter outputs are always connected to receiver inputs.

Unfortunately, when this requirement first came up in the development of 10Base-T, IEEE 802.3 chose not to make a hard rule as to whether the crossover should be implemented in the cable as shown in Figure or whether it should be implemented internally as shown in Figure .

Figure Alternative Ways for Implementing the Link Crossover Requirement



Instead, IEEE 802.3 defined two rules and made two recommendations:

- There must be an odd number of crossovers in all multiconductor links.
- If a PMD is equipped with an internal crossover, its MDI must be clearly labeled with the graphical X symbol.
- Implementation of an internal crossover function is optional.
- When a DTE is connected to a repeater or switch (DCE) port, it is recommended that the crossover be implemented within the DCE port.

The eventual result was that ports in most DCEs were equipped with PMDs that contained internal crossover circuitry and that DTEs had PMDs without internal crossovers. This led to the following oft-quoted de facto "installation rule":

- Use a straight-through cable when connecting DTE to DCE. Use a crossover cable when connecting DTE to DTE or DCE to DCE.

Unfortunately, the de facto rule does not apply to all Ethernet versions that have been developed subsequent to 10Base-T. As things now stand, the following is true:

- All fiber-based systems use cables that have the crossover implemented within the cable.
- All 100Base systems using twisted-pair links use the same rules and recommendations as 10Base-T.
- 1000Base-T NICs may implement a selectable internal crossover option that can be negotiated and enabled during autonegotiation. When the selectable crossover option is not implemented, 10Base-T rules and recommendations apply.

10.9 Synchronous optical networking`(SONET)

Synchronous optical networking, is a method for communicating digital information using lasers or light-emitting diodes (LEDs) over optical fiber. The method was developed to replace the Plesiochronous Digital Hierarchy(PDH) system for transporting large amounts of telephone and data traffic and to allow for interoperability between equipment from different vendors.

There are multiple very closely related standards that describe synchronous optical networking:

- SDH or Synchronous Digital Hierarchy standard developed by the International Telecommunication Union(ITU), documented in standard G.707 and its extension G.708
- SONET or Synchronous Optical Networking standard as defined by GR-253-CORE from Telcordia and T1.105 from American National Standards Institute

Both SDH and SONET are widely used today: SONET in the U.S. and Canada and SDH in the rest of the world. Although the SONET standards were developed before SDH, their relative penetrations in the worldwide market dictate that SONET now be considered the variation.

Synchronous networking differs from PDH in that the exact rates that are used to transport the data are tightly synchronized across the entire network, made possible by atomic clocks. This synchronization system allows entire inter-country networks to operate synchronously, greatly reducing the amount of buffering required between elements in the network.

Both SONET and SDH can be used to encapsulate earlier digital transmission standards, such as the PDH standard, or used directly to support either ATM or so-called Packetover SONET/SDH(POS) networking. As such, it is inaccurate to think of SDH or SONET as communications protocols in and of themselves, but rather as generic and all-purpose transport containers for moving both voice and data. The basic format of an SDH signal allows it to carry many different services in its Virtual Container (VC) because it is bandwidth-flexible.

Structure of SONET/SDH signals

SONET and SDH often use different terms to describe identical features or functions, sometimes leading to confusion that exaggerates their differences. With a few exceptions, SDH can be thought of as a superset of SONET. The two main differences between the two:

- SONET can use either of two basic units for framing while SDH has one
- SDH has additional mapping options which are not available in SONET.

The basic unit of transmission

The basic unit of framing in SDH is an STM-1(Synchronous Transport Module level - 1), which operates at 155.52 Mbit/s. SONET refers to this basic unit as an STS-

3c(Synchronous Transport Signal - 3, concatenated), but its high-level functionality, frame size, and bit-rate are the same as STM-1.

SONET offers an additional basic unit of transmission, the STS-1 (Synchronous Transport Signal - 1), operating at 51.84 Mbit/s - exactly one third of an STM-1/STS-3c.

Framing

In packet oriented data transmission such as Ethernet, a packet frame usually consists of a header and a payload, with the header of the frame being transmitted first, followed by the payload (and possibly a trailer, such as a CRC). In synchronous optical networking, this is modified slightly. The header is termed the overhead and the payload still exists, but instead of the overhead being transmitted before the payload, it is interleaved, with part of the overhead being transmitted, then part of the payload, then the next part of the overhead, then the next part of the payload, until the entire frame has been transmitted. In the case of an STS-1, the frame is 810 octets in size; whereas in an STM-1 or STS-3c, the frame is 2430 octets in size. For STS-1, the frame is transmitted as 3 octets of overhead, followed by 87 octets of payload. This is repeated nine times over until 810 octets have been transmitted, taking 125 microseconds. In the case of an STS-3c/STM-1 which operates three times faster than STS-1, 9 octets of overhead are transmitted, followed by 261 octets of payload. This is also nine times over until 2,430 octets have been transmitted, also taking 125 microseconds. For both SONET and SDH, this is normally represented by the frame being displayed graphically as a block: of 90 columns and 9 rows for STS-1; and 270 columns and 9 rows for SDH/STS-3c. This representation aligns all the overhead columns, so the overhead appears as a contiguous block, as does the payload.

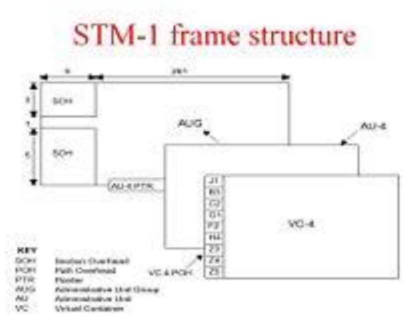
The internal structure of the overhead and payload within the frame differs slightly between SONET and SDH, and different terms are used in the standards to describe these structures. However, the standards are extremely similar in implementation, such that it is easy to interoperate between SDH and SONET at particular bandwidths.

It is worth noting that the choice of a 125 microsecond interval is not an arbitrary one. What it means is that the same octet position in each frame comes past every 125 microseconds. If one octet is extracted from the bitstream every 125 microseconds, this gives a data rate of 8 bits per 125 microseconds - or 64 kbit/s, the basic DS0 telecommunications rate. This relation allows an extremely useful behaviour of synchronous optical networking, which is that low data rate channels or streams of data can be extracted from high data rate streams by simply extracting octets at regular time intervals - there is no need to understand or decode the entire frame. This is not possible in PDH networking. Furthermore, it shows that a relatively simple device is all that is needed to extract a datastream from an SDH framed connection and insert it into a SONET framed connection and vice versa.

In practice, the terms STS-1 and OC-1 are sometimes used interchangeably, though the OC-N format refers to the signal in its optical form. It is therefore incorrect to say that an OC-3 contains 3 OC-1s: an OC-3 can be said to contain 3 STS-1s.

SDH Frame

A STM-1 Frame. The first 9 columns contain the overhead and the pointers. For the sake of simplicity, the frame is shown as a rectangular structure of 270 columns and 9 rows



For the sake of simplicity, the frame is shown as a rectangular structure of 270 columns and 9 rows. The first 3 rows and 9 columns contain Regenerator Section Overhead (RSOH) and the last 5 rows and 9 columns contain Multiplex Section Overhead (MSOH. The 4 row from the top contains pointers

The STM-1 (Synchronous Transport Module level - 1) frame is the basic transmission format for SDH or the fundamental frame or the first level of the synchronous digital hierarchy. The STS-1 frame transmitted in exactly 125 microseconds therefore, there are 8000 frames per second on a fiber-optic circuit designated OC-1 (optical carrier one). The STM-1 frame consists of overhead plus a virtual container capacity. The first 9 columns of each frame make up the Section Overhead, and the last 261 columns make up the Virtual Container (VC) capacity. The VC plus the pointers (H1, H2, H3 bytes) is called the AU (Administrative Unit).

Carried within the VC capacity, which has its own frame structure of nine rows and 261 columns, is the Path Overhead and the Container. The first column is for Path Overhead; it's followed by the payload container, which can itself carry other containers. Virtual Containers can have any phase alignment within the Administrative Unit, and this alignment is indicated by the Pointer in row four,

The Section overhead of an STM-1 signal (SOH) is divided into two parts: the Regenerator Section Overhead (RSOH) and the Multiplex Section Overhead (MSOH). The overheads contain information from the system itself, which is used for a wide range of management functions, such as monitoring transmission quality, detecting failures, managing alarms, data communication channels, service channels, etc.

The STM frame is continuous and is transmitted in a serial fashion, byte-by-byte, row-by-row.

STM–1 frame contains

- Total content : 9 x 270 bytes = 2430 bytes
    - overhead : 9 rows x 9 bytes
    - payload : 9 rows x 261 bytes
- Period : 125 μsec
- Bitrate : 155,520 Mbit/s (2430 x 8 bits x 8000 frames/sec)

- payload capacity : 150,336 Mbit/s (2349 x 8 bits x 8000 frames/sec)

The transmission of the frame is done row by row, from the top left corner Framing Structure

The frame consists of two parts, the transport overhead and the path virtual envelope.

Transport overhead

The transport overhead is used for signaling and measuring transmission error rates composed of 27 bytes/octets as follows:

- Section overhead- called RSOH (Regenerator Section Overhead) in SDH terminology: 27 octets containing information about the frame structure required by the terminal equipment.
- Line overhead - called MSOH (Multiplex Section Overhead) in SDH: 45 octets containing information about alarms, maintenance and error correction as may be required within the network.
- Pointer – these point to the location of the J1 byte in the payload overhead.

Path virtual envelope

Data transmitted from end to end is referred to as path data. It is composed of two components:

- Payload overhead (POH): 9 bytes used for end to end signaling and error measurement.
- Payload: user data (774 bytes for STS-1, or 2340 bytes for STM-1/STS-3c)

For STS-1, the payload is referred to as the synchronous payload envelope (SPE), which in turn has 18 stuffing bytes, leading to the STS-1 payload capacity of 756 bytes.[1]

The STS-1 payload is designed to carry a full PDH DS3 frame. When the DS3 enters a SONET network, path overhead is added, and that SONET network element (NE) is said to be a path generator and terminator. The SONET NE is said to be line terminating if it processes the line overhead. Note that wherever the line or path is terminated, the section is terminated also. SONET Regenerators terminate the section but not the paths or line.

SONET/SDH and relationship to 10 Gigabit Ethernet

Another circuit type amongst data networking equipment is 10 Gigabit Ethernet (10GbE). This is similar to the line rate of OC-192/STM-64 (9.953 Gbit/s). The Gigabit Ethernet Alliance created two 10 Gigabit Ethernet variants: a local area variant (LAN PHY), with a line rate of exactly 10,000,000 kbit/s and a wide area variant (WAN PHY), with the same line rate as OC-192/STM-64 (9,953,280 kbit/s). The Ethernet wide area variant encapsulates its data using a light-weight SDH/SONET frame so as to be compatible at low level with equipment designed to carry those signals.

However, 10 Gigabit Ethernet does not explicitly provide any interoperability at the bitstream level with other SDH/SONET systems. This differs from WDM System Transponders, including both Coarse- and Dense-WDM systems (CWDM, DWDM) that currently support OC-192 SONET Signals, which can normally support thin-SONET framed 10 Gigabit Ethernet.

SONET/SDH data rates

In the above table, Payload bandwidth is the line rate less the bandwidth of the line and section overheads User throughput must also deduct path overhead from this, but path overhead bandwidth is variable based on the types of cross-connects built across the optical system.

Note that the typical data rate progression starts at OC-3 and increases by multiples of 4. As such, while OC-24 and OC-1536, along with other rates such as OC-9, OC-18, OC-36, and OC-96 may be defined in some standards documents, they are not available on a wide-range of equipment.

SONET physical layer

The physical layer in SONET actually comprises a large number of layers within it, only one of which is the optical/transmission layer (which includes bitrates, jitter specifications, optical signal specifications and so on). The SONET and SDH standards come with a host of features for isolating and identifying signal defects and their origins.

SONET/SDH Network Management Protocols

SONET equipment is often managed with the TL1 protocol. TL1 is a traditional telecom language for managing and reconfiguring SONET network elements. TL1 (or whatever command language a SONET Network Element utilizes) must be carried by other management protocols, including SNMP, CORBA and XML.

SONET Network Management is a large, difficult, and arcane subject, but there are some features that are fairly universal. First of all, most SONET NEs have a limited number of management interfaces defined. These are:

- Electrical Interface. The electrical interface (often 50 Ω) sends SONET TL1 commands from a local management network physically housed in the Central Officewhere the SONET NE is located. This is for "local management" of that NE and, possibly, remote management of other SONET NEs.

- Craft Interface. Local "craftspersons" can access a SONET NE on a "craft port" and issue commands through a dumb terminal or terminal emulation program running on a laptop. This interface can also be hooked-up to a console server, allowing for remote out-of-band management and logging.

- SONET and SDH have dedicated Data Communication Channels (DCC)s within the section and line overhead for management traffic. Generally, section overhead (regenerator section in SDH) is used. According to ITU-T G.7712, there are three modes used for management:

  - IP-only stack, using PPP as data-link
  - OSI-only stack, using LAP-Das data-link

- Dual (IP+OSI) stack using PPP or LAP-D with tunneling functions to communicate between stacks.

An interesting fact about modern SONET NEs is that, to handle all of the possible management channels and signals, most NEs actually contain a router for routing the network commands and underlying (data) protocols.

The main functions of SONET Network Management include:

- SONET Network and NE Provisioning. In order to allocate bandwidth throughout a SONET Network, each SONET NE must be configured. Although this can be done locally, through a craft interface, it is normally done through a Network Management System (sitting at a higher layer) that in turn operates through the SONET/SDH Network Management Network.

- Software Upgrade. SONET NE Software Upgrade is in modern NEs done mostly through the SONET/SDH Management network.

- Performance Management. SONET NEs have a very large set of standards for Performance Management. The PM criteria allow for monitoring not only the health of individual NEs, but for the isolation and identification of most network defects or outages. Higher-layer Network monitoring and management software allows for the proper filtering and troubleshooting of network-wide PM so that defects and outages can be quickly identified and responded to.

SONET equipment

With recent advances in SONET and SDH chipsets, the traditional categories of SONET NEs are breaking down. Nevertheless, as SONET Network architectures have remained relatively constant, even newer SONET Equipment (including "Multiservice Provisioning Platforms") can be examined in light of the architectures they will support. Thus, there is value in viewing new (as well as traditional) SONET Equipment in terms of the older categories.

SONET regenerator

Traditional SONET Regenerators terminate the SONET Section overhead, but not the line or path. SONET Regens extend long haul routes in a way similar to most regenerators, by converting an optical signal that has already traveled a long distance into electrical format and then retransmitting a regenerated high-power signal.

Since the late 1990s, SONET regenerators have been largely replaced by OpticAL Amplifiers. Also, some of the functionality of SONET Regens has been absorbed by the Transponders of Wavelength Division Multiplexingsystems.

SONET add-drop multiplexer (ADM)

SONET ADMs are the most common type of SONET Equipment. Traditional SONET ADMs were designed to support one of the SONET Network Architectures, though new generation SONET systems can often support several architectures, sometimes simultaneously. SONET ADMs traditionally have a "high speed side" (where the full line rate signal is supported), and a "low speed side", which can consist of electrical as well as optical interfaces. The low speed side takes in low speed signals which are multiplexed by the SONET NE and sent out from the high speed side, or vice versa.

SONET Digital Cross Connect system

Recent SONET Digital Cross Connect systems (DCSs or DXCs) support numerous high-speed signals, and allow for cross connection of DS1s, DS3s and even STS-3s/12c and so on, from any input to any output. Advanced SONET DCSs can support numerous subtending rings simultaneously.

SONET Network Architectures

Currently, SONET (and SDH) have a limited number of architectures defined. These architectures allow for efficient bandwidth usage as well as protection (i.e. the ability to transmit traffic even when part of the network has failed), and are key in understanding the almost worldwide usage of SONET and SDH for moving digital traffic. The three main architectures are:

- Linear APS (Automatic Protection Switching), also known as 1+1: This involves 4 fibers: 2 working fibers in each direction, and two protection fibers. Switching is based on the line state, and may be unidirectional, with each direction switching independently, or bidirectional, where the NEs at each end negotiate so that both directions are generally carried on the same pair of fibers.
- UPSR (Unidirectional Path Switched Ring): In a UPSR, two redundant (path-level) copies of protected traffic are sent in either direction around a ring. A selector at the egress node determines the higher-quality copy and decides to use the best copy, thus coping if deterioration in one copy occurs due to a broken fiber or other failure. UPSRs tend to sit nearer to the edge of a SONET network and, as such, are sometimes called "collector rings". Because the same data is sent around the ring in both directions, the total capacity of a UPSR is equal to the line rate N of the OC-N ring. For example if we had an OC-3 ring with 3 STS-1s used to transport 3 DS-3s from ingress node A to the egress node D, then 100% of the ring bandwidth (N=3) would be consumed by nodes A and D. Any other nodes on the ring, say B and C could only act as pass through nodes. The SDH analog of UPSR is Subnetwork Connection Protection (SNCP); however, SNCP does not impose a ring topology, but may also be used in mesh topologies.

- BLSR (Bidirectional Line Switched Ring): BLSR comes in two varieties, 2-fiber BLSR and 4-fiber BLSR. BLSRs switch at the line layer. Unlike UPSR, BLSR does not send redundant copies from ingress to egress. Rather, the ring nodes adjacent to the failure reroute the traffic "the long way" around the ring. BLSRs trade cost and complexity for bandwdith efficiency as well as the ability to support "extra traffic", which can be pre-empted when a protection switching event occurs. BLSRs can operate within a metropolitan region or, often, will move traffic between municipalities. Because a BLSR does not send redundant copies from ingress to egress the total bandwidth that a BLSR can support is not limited to the line rate N of the OC-N ring and can actually be larger than N depending upon the traffic pattern on the ring. The best case of this is that all traffic is between adjacent nodes. The worst case is when all traffic on the ring egresses from a single node, i.e. the BLSR is serving as a collector ring. In this case the bandwidth that the ring can support is equal to the line rate N of the OC-N ring. This is why BLSRs are seldom if ever deployed in collector rings but often deployed in inter-office rings. The SDH equivalent of BLSR is called Multiplex Section-Shared Protection Ring (MS-SPRING).

SONET synchronization

Synchronization of SONET and SDH networks is a difficult and arcane subject. Remember that a SONET NE will transport and/or multiplex traffic that has originated from a variety of different clock sources. In addition, a SONET NE may have a number of different synchronization options to choose from, which in some cases it will do so dynamically based on Synch Status Messages and other indicators.

As for Synchronization sources available to a SONET NE, these are:

- Local External Timing. This is generated by an atomic Cesium clock or a satellite-derived clock by a device located in the same central office as the SONET NE. The interface is often a DS1, with Sync Status Messages supplied by the clock and placed into the DS1 overhead.
- Line-derived timing. A SONET NE can choose (or be configured) to derive its timing from the line-level, by monitoring the S1 sync status bytes to ensure quality.
- Holdover. As a last resort, in the absence of higher quality timing, a SONET NE can go into "holdover" until higher quality external timing becomes available again. In this mode, a SONET NE uses its own timing circuits to time the SONET signal.

An interesting and hard-to-troubleshoot issue in SONET Networks is the existence of "timing loops". With a timing loop, SONET NEs in a network are each deriving their timing from another NE, and back again to initial NE, like a snake biting its own tail.

This network loop will eventually see its own timing "float away" from any external SONET networks, causing mysterious bit errors, the source of which can be hard to find (unless the presence of the timing loop is detected). In general, a SONET Network that has been properly configured will never find itself in a timing loop, but it is sometimes hard to avoid this without sophisticated network management tools.

Next-generation SONET/SDH

SONET/SDH development was originally driven by the need to transport multiple PDH signals like DS1, E1, DS3 and E3 along with other groups of multiplexed 64 kbit/s pulse-code modulated voice traffic. The ability to transport ATM traffic was another early application. In order to support large ATM bandwidths, the technique of concatenation was developed, whereby smaller SONET multiplexing containers (eg, STS-1) are inversely multiplexed to build up a larger container (eg, STS-3c) to support large data-oriented pipes. SONET/SDH is therefore able to transport both voice and data simultaneously.

One problem with traditional concatenation, however, is inflexibility. Depending on the data and voice traffic mix that must be carried, there can be a large amount of unused bandwidth left over, due to the fixed sizes of concatenated containers. For example, fitting a 100 Mbit/s Fast Ethernet connection inside a 155 Mbit/s STS-3c container leads to considerable waste.

Virtual Concatenation (VCAT) allows for a more arbitrary assembly of lower order multiplexing containers, building larger containers of fairly arbitrary size (e.g. 100 Mbit/s) without the need for intermediate SONET NEs to support this particular form of concatenation. Virtual Concatenation increasingly leverages X.86 or Generic Framing Procedure (GFP) protocols in order to map payloads of arbitrary bandwidth into the virtually concatenated container.

Link Capacity Adjustment Scheme (LCAS) allows for dynamically changing the bandwidth via dynamic virtual concatenation, multiplexing containers based on the short-term bandwidth needs in the network.

The set of next generation SONET/SDH protocols to enable Ethernet transport is referred to as Ethernet over SONET/SDH (EoS).

1.10 MULTIPLEXING

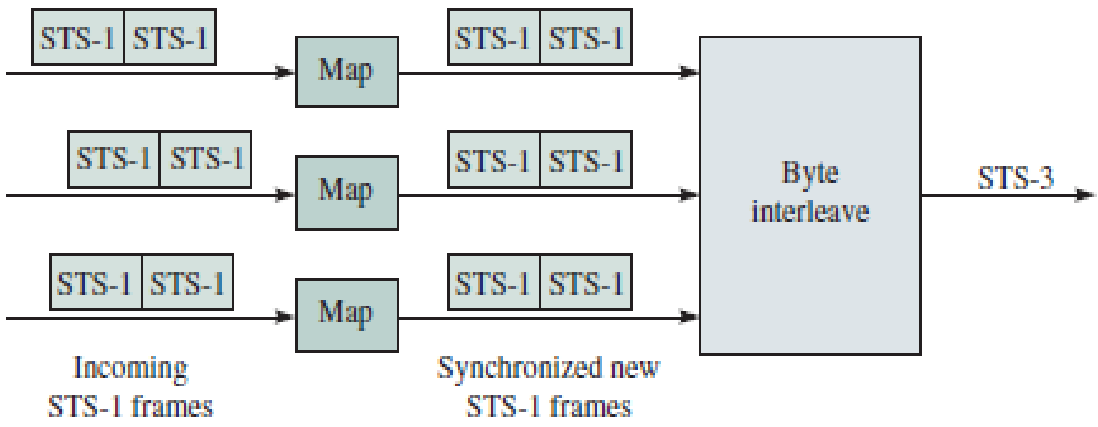
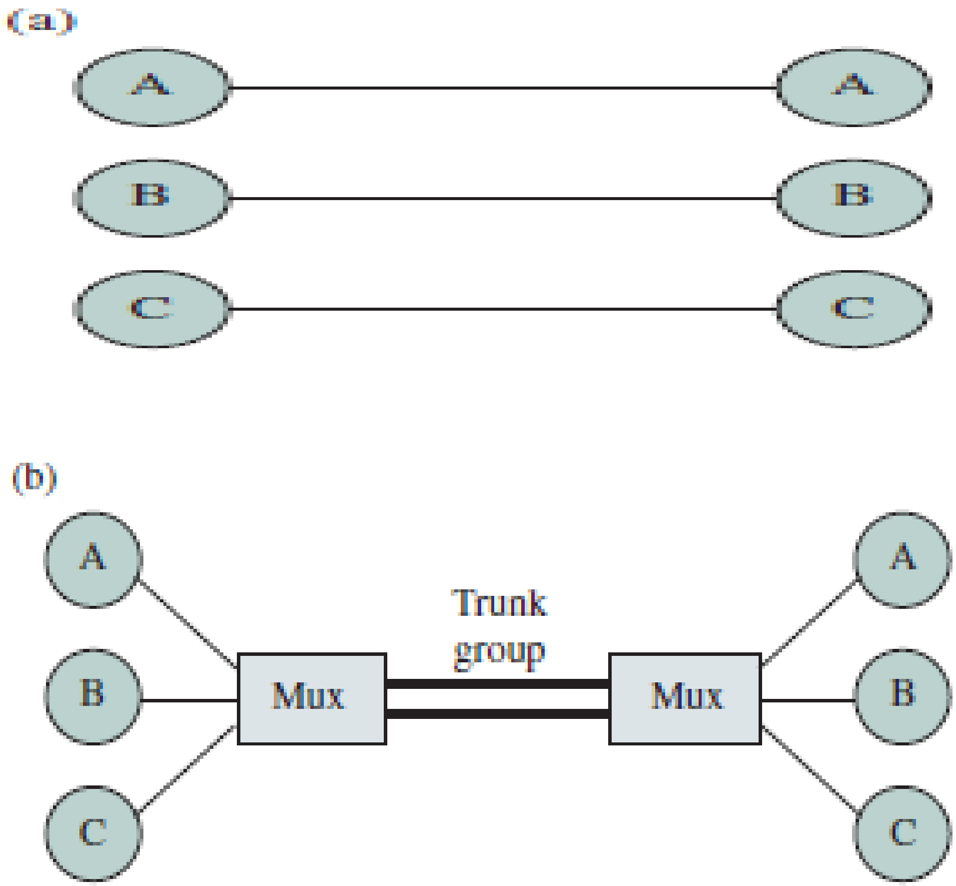
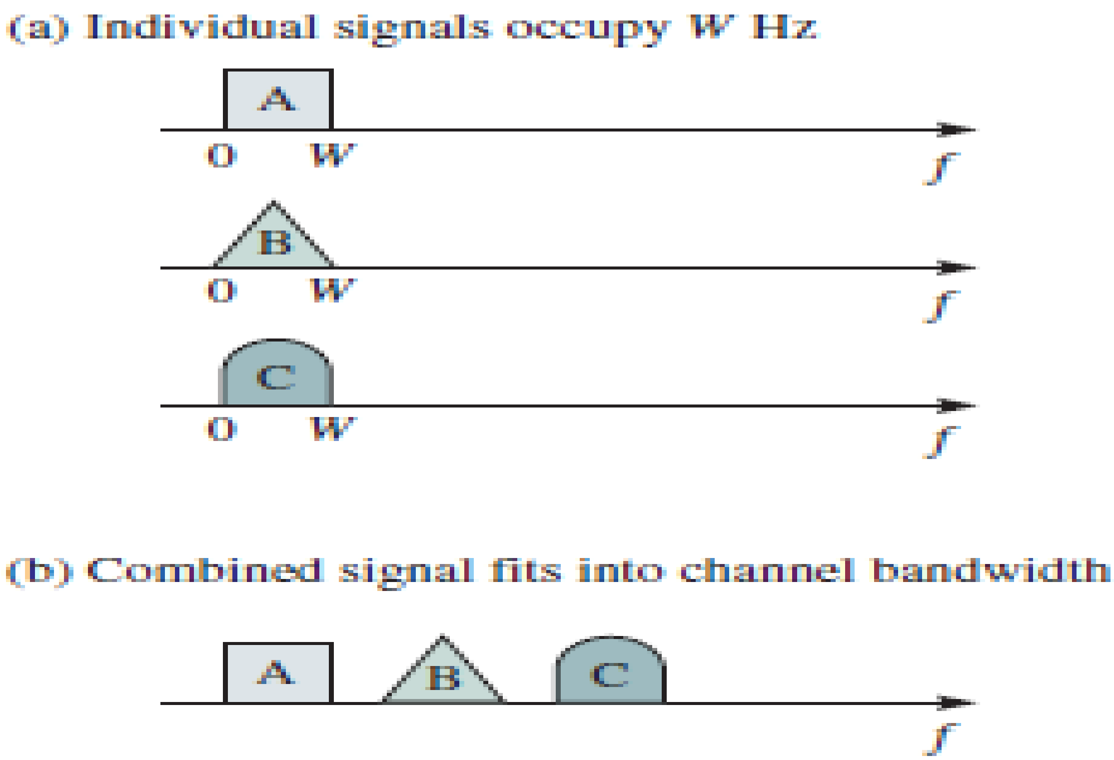


FIGURE 4.17 Synchronous multiplexing in SONET

- sharing of expensive network resources by several connections or information flows
- bandwidth, measured in Hertz for analog transmission systems and bits/second for digital transmission systems
- transmission lines connecting the two multiplexers are called trunks

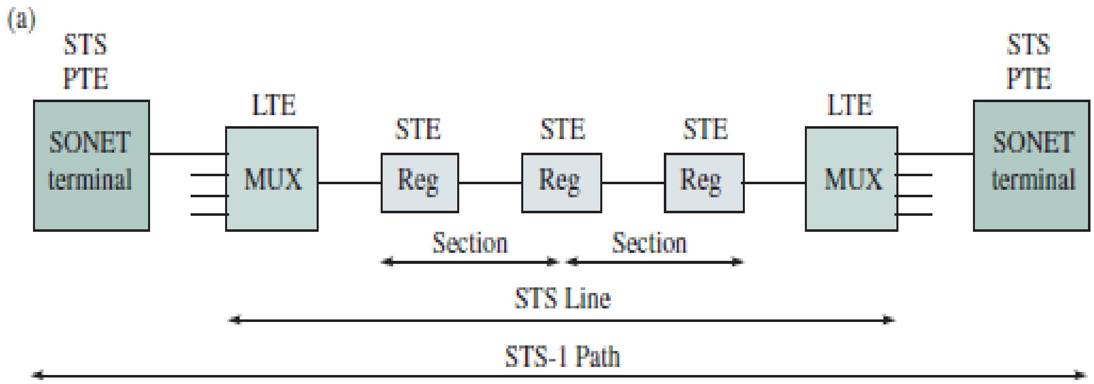


- **Frequency-Division Multiplexing**



10.11 SONET FRAME STRUCTURE

divided into three layers: sections, lines, and paths



STE: Section terminating equipment, for example, a repeater
LTE: Line terminating equipment, for example, an STS-1 to STS-3 multiplexer
PTE: Path terminating equipment, for example, an STS-1 multiplexer

SONET Frame Structure

- Section
  - refers to the span of fiber between two adjacent devices, such as two repeaters
  - section layer deals with the transmission of an STS-n signal across the physical medium
- Line
  - refers to the span between two adjacent multiplexers
  - therefore in general encompasses several sections
  - Lines deal with the transport of an aggregate multiplexed stream of user information and the associated overhead.
- Path
  - Refers to the span between the two SONET terminals at the endpoints of the system and in general encompasses one or more lines.

SONET system

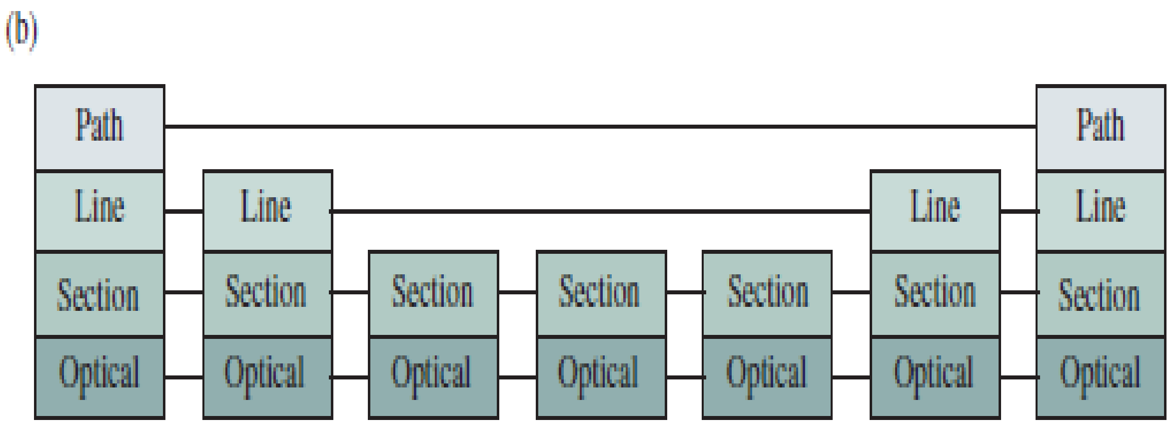


FIGURE 4.14 Section, line, and path layers of SONET

- every section has an associated optical layer
- section layer deals with the signals in their electrical form, and the optical layer deals with the transmission of optical pulses
- every regenerator involves converting the optical signal to electrical form to carry out the regeneration function and then back to optical form
- all of the equipment implements the optical and section functions
- Line functions are found in the multiplexers and end terminal equipment.
- Path function occurs only at the end terminal equipment

SONET frame structure

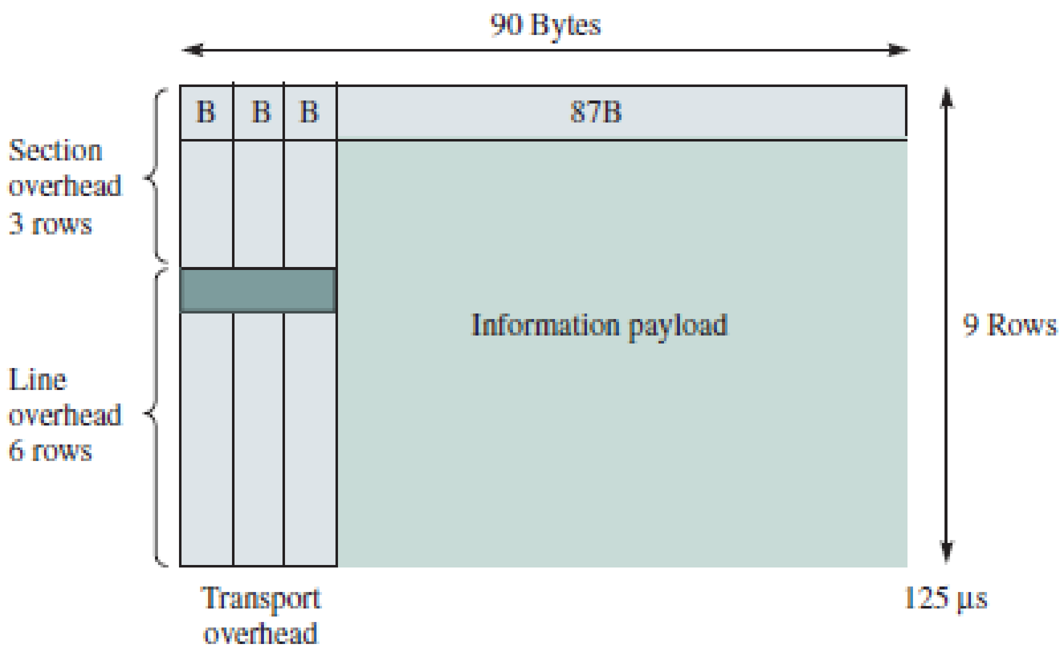


FIGURE 4.15 SONET STS-1 frame format

- SONET STS-1 frame that is defined at the line level.
- consisting of a rectangular array of bytes arranged in 9 rows by 90 bytes is repeated 8000 times a second
- each byte in the array corresponds to a bit rate of 64 kbps, and the overall bit rate of the STS-1 is 8 x 9 x 90 x 8000 = 51.84 Mbps
- first 3 columns of the array are allocated to section and line overhead.
- Section overhead is interpreted and modified at every section termination.

- used to provide framing, error monitoring, and other section-related management functions

- line overhead is interpreted and modified at every line termination
  - used to provide synchronization and multiplexing for the path layer, as well as protection-switching capability.
- remaining 87 columns of the frame constitute the **information payload** that carries the path layer information.
- bit rate of the information payload is
- information payload includes one column of path overhead information, but the column is not necessarily aligned to the frame
- user data and the path overhead are included in the synchronous payload envelope (SPE), which consists of a byte array of 87 columns by nine rows.
- path overhead constitutes the first column of this array
- This SPE is then inserted into the STS-1 frame.

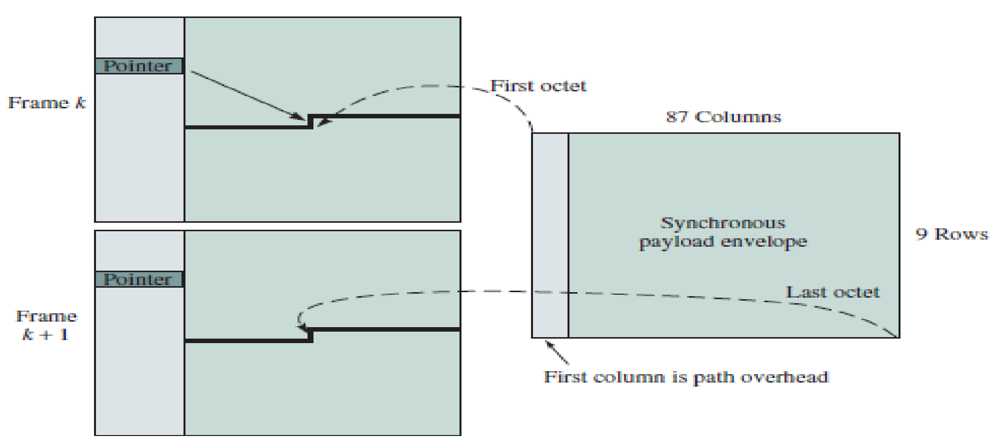


FIGURE 4.16 The synchronous payload envelope can span two consecutive frames

- SPE is not necessarily aligned to the information payload of an STS-1 frame

- Instead, the first 2 bytes of the line overhead are used as a pointer that indicates the byte within the information payload where the SPE begins.
- SPE can be spread over two consecutive frames
- use of the pointer makes it possible to extract a tributary signal from the multiplexed signal.
- add-drop capability.
- pointer structure maintains synchronization of frames and SPEs.
- If the payload stream is faster than the frame rate, then a buffer is required to hold payload bits as the frame stream falls behind the payload stream
- To allow the frame to catch up, an extra SPE byte is transmitted in a frame from time to time.
- This extra byte clears the backlog that has built up.

- Whenever this byte is inserted, the pointer is moved forward by one byte to indicate that the SPE starting point has been moved one byte forward
- When the payload stream is slower than the frame stream, the number of SPE bytes transmitted in a frame needs to be reduced by one byte from time to time.
- This is done by stuffing an SPE byte with dummy information and adjusting the pointer to indicate that the SPE now starts one byte later.

Module 2

ISDN - Definition - Protocol architecture - System architecture - Transmission channels - ISDN interface, B-ISDN.

1.1 ISDN

ISDN - Integrated Services Digital Network
Telephone services -> Telecommunication services
Used for voice, image and data

Current trends in telecommunication are toward integration of voice and data services. So far these services have been available separately, requiring separate subscription, communication links, and equipment. It has long been acknowledged that the integration of these services will result in significant flexibility and cost benefits to both service users and service providers. The Integrated Service Digital Network (ISDN) is a major attempt to realize these objectives.

PRINCIPLES OF ISDN

Support of voice and non-voice applications in the same network

– interfaces and data transmission facilities standardized by ITU-T

• Switched and non-switched connections

– packet & circuit switching, leased lines

• 64-kbps channel

  – chosen because at the time was the standard rate for digitized voice.

Layered protocol structure

– mapped into OSI model (advantages in utilizing existing standards as well as in developing new ones)

• Variety of configurations

– according to specific national situations &state of technology

Basic Concepts

ISDN provides a fully integrated digital network for voice and data communication. It supports both circuit and packet switching.

Each ISDN switch consists of an exchange termination part, which performs the necessary circuit switching functions, and a packet handler, which performs the necessary packet switching functions. The packet handlers implement X.25 and are connected to a public packet switched network via X.75. The exchange terminations are interconnected via tandem exchanges. STPs and SCPs provide network intelligence, and were described in the previous chapter. Subscriber access is provided via a network termination and/or terminal adapter (NT/TA). This provides the connectivity for a variety of user devices, including ISDN phones, Plain Old Telephone Sets (POTS), LANs, PBXs, and X.25 terminals.

ISDN Services

ISDN provides three types of services:

* Bearer services

* Teleservices

* Supplementary services

Tele and supplementary services represent the type of features and functions which are visible to end-users, while bearer services represent the parts of the network which remain hidden from end-users.

Bearer services facilitate the real-time communication of digital information between end-users. These services mainly relate to network functions and account for OSI layers 1-3. An example of a bearer services is the 64 kbps, 8 kHz structured, speech service. This service uses a data rate of 64 kbps together with 8 kHz timing information (which structures the data into octet intervals) for transmitting a Pulse Code Modulated (PCM) speech signal. The fact that the signal represents speech is known to the network, allowing it to employ transformations which may not preserve bit integrity but will result in good quality audio reproduction.

Teleservices provide a set of higher-level functions on top of bearer services. These services account for OSI layers 4-7. Examples of teleservices are:

* Telephony services which provide speech communication over a B channel with control signaling over the D channel.

* Facsimile services which facilitate the communication of bitmap images over a B channel with control signaling over the D channel.

* Teletex services which facilitate the interchange and communication of textual as well as formatted documents over a B channel with control signaling over the D channel.

Supplementary services enhance bearer and teleservices in an independent fashion. Examples of supplementary services are:

* The Centrex service emulates a private network and provides specialized features to a set of subscribers.
* The Call Transfer service allows a user to transfer an active call to a third-party.
* The Call Waiting service allows a user already engaged in a call to be informed of another incoming call.
* The Calling Line ID service provides the calling party's address information to the called party.

Although these services all appear geared toward circuit-switched telephone calls, they are equally applicable to packet-switched data calls.

2.2 DEFINITION

ISDN stands for Integrated Services Digital Network. It was first introduced by NEC in Japan. There basic purpose was integration of traditionally different computer and communication (C&C) services into a single one. The integration basically means incorporation of three types of services:

Voice (telephone)

Data (internet)

Entertainment (TV)

The integration should be most comfortably and efficiently done in digital domain, so the switching, multiplexing, signaling and transmission, everything should be digital. It was first named integrated digital network (IDN), which received lukewarm response as only the enterprises, not the general public, realized the potential behind that acronym. Later on it was named ISDN which more clearly states the idea (of integrating different services)behindit.

2.3 ISDN Protocol Architecture

Network	**Call control Q.931**	**X.25 Packet level**			
Data Link	**LAPD (Q.921)**		**V.120 or frame relay**		**LAPB**
Physical	**I.430 basic interface + I.431 primary interface**				
	Signal	**Packet**	**Circuit-switched**	**Semi-permanent**	**Packet-switched**
	D channel		**B channel**		

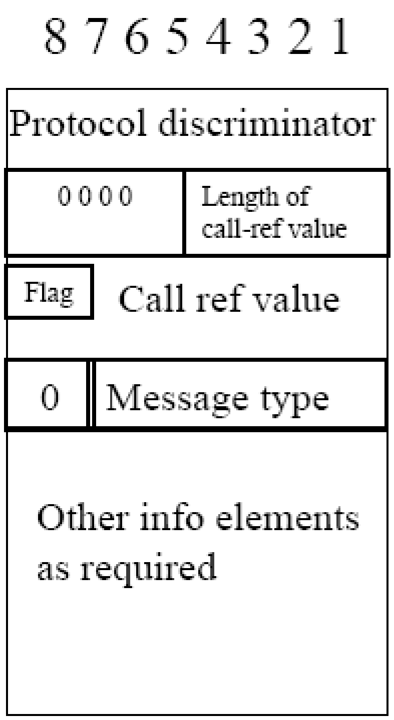
ISDN Call-Control Protocol

Protocol Descriminator
used to distinguish messages for user network call control.

call reference
identifies the B-channel call to which this message refers

Message Types

identifies which 1.45 1/Q.931 mes sage is being sent e.g. S E T UP, DISCONNECT .



Q.931 Messages

Circuit mode connection control➡
refers to the functions needed to set up, maintain, and deara circuit-switched connection on a B channel. This function corresponds to call control in exis ting circuit-switching telelecommunications networks .

➡

Packet mode connection control

refers to the functions needed to set up a circuit-switched connection (called an acces s connection in this context) to an IS DN packet-switching node; this connects the user to the packet-switching network furnished by the ISDN provider.

User to user signaling

allows two users to communicate without setting up a circuit-switched connection. A temporary signaling connection is es tablished and cleared in a manner s imilar to the control of a circuit-switched connection. Signaling takes place over the D channel and thus does not consume Bchannel resources.

Functions Of Q.931

1.used to set up a call initially.

2.sent between user and network once a call has been set up but prior to the disesablishment (termination) phas e. One of the mes sages in that group allows the network to relay, without modification, information between the two users of the call.

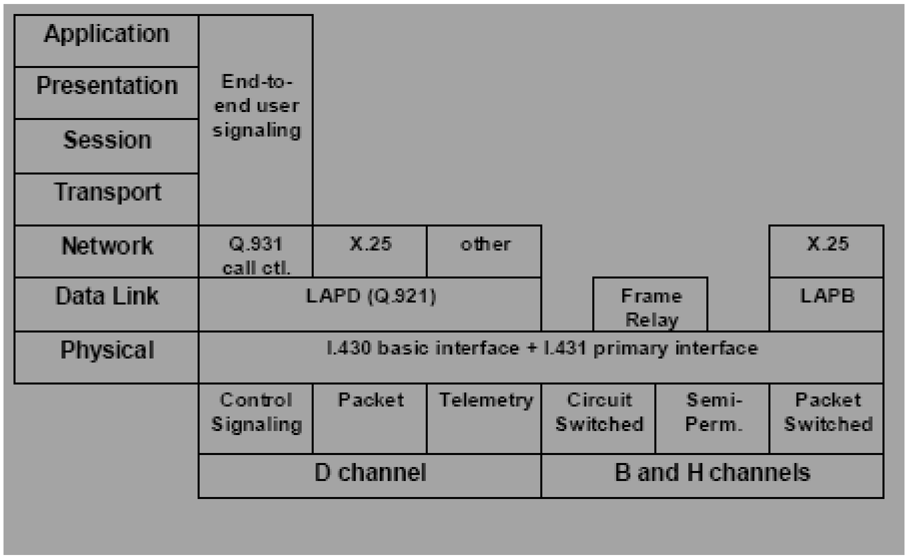
3. sent between user and network in order to terminate a call.

LAPD Protocol

Layer-3 information is transferred in unnumbered frames .

Error detection is used to discard damaged frames , but there is no error control or flow control.

Layer-3 information is trans ferred in frames that include sequence numbers and are acknowledged. E rror-control and flow-control procedures are included in theprotocol. This type is also referred to in the s tandard as multiple-frame operation.



Physical Layer

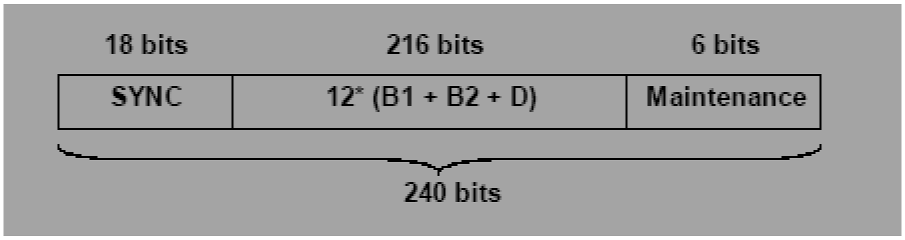
• 2B1Q (2 binary, 1quaternary) most common signaling method on U interface

• 2 bits per symbol

• 80 kbaud, 160 kbps

Bits	Quaternary Symbol	Voltage Level
00	-3	-2.5
01	-1	-0.833
10	+3	+2.5
11	+1	+0.833

Physical Layer Frame Format



Each frame 1.5 msec long

• SYNC field (9 quaternaries) : +3 +3 -3 -3 -3 +3 -3 +3 -3

• 12 * (8 bits from each B channel + 2 bits from D)

• Maintenance contains CRC, other operation info.

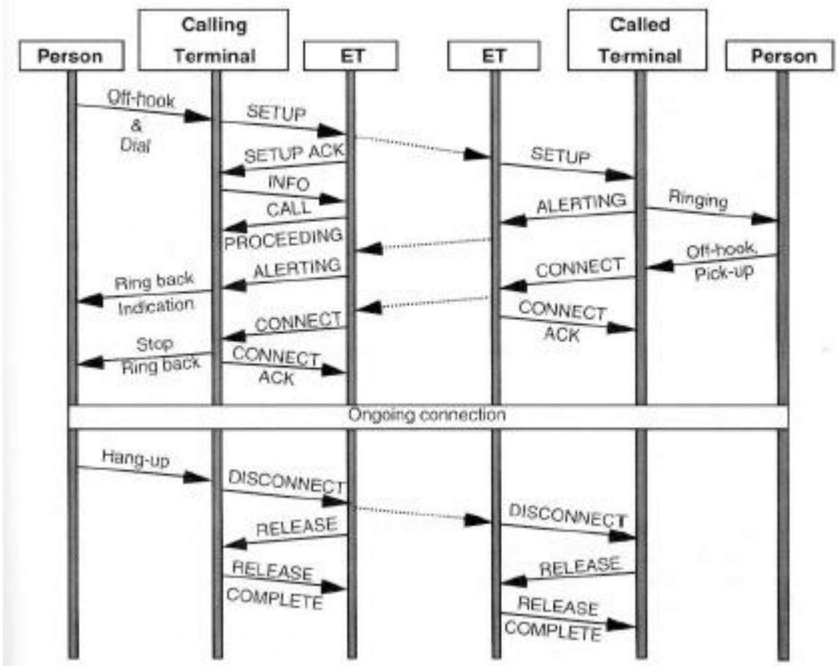
Link Access Protocol - D Channel(LAPD)

• Layer 2 protocol

• Almost identical to LAP-B used w/ X.25 (based on HDLC)

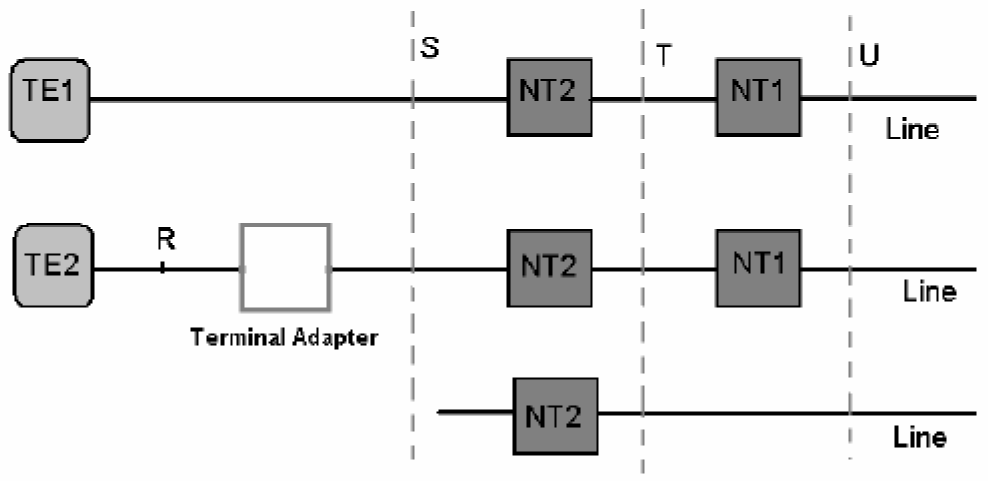
• Provides unacknowledged information-transfer service (unnumbered frames, error detection to discard frame but no error control or flow control) and acknowledged information transfer

ISDN Connection Establishment and Release



2.4 ISDN SYSTEM ARCHITECTURE

The key idea behind ISDN is the digital bit pipe, a conceptual pipe between the customer and the carrier through which bits flow. Whether the bits originated from a digital telephone, a digital terminal, a digital facsimile machine, or some other device is irrelevant. All that matters is that bits can flow through the pipe in both directions. The digital bit pipe can, and normally does, support multiple independent channels by time division multiplexing of the bit streams. The exact format of the bit stream and its multiplexing is a carefully defied part of the interface specifications for the digital bit pipe. Two principal standards for the bit pipe have been developed, a low bandwidth standard for home use and a higher bandwidth standard for business use that supports multiple channels that are identical to the home use channels. Furthermore, businesses may have multiple bit pipes if they need additional capacity beyond what the standard business pipe can provide. The carrier places a network terminating device (NT1), on the customer's premises and connects it to the ISDN exchange in the carrier's office, several kilometers away, using the twisted pair that was previously used to connect to the telephone. The NT1 box has a connector on it into which a passive bus cable can be inserted. Up to eight ISDN telephones, terminals, alarms, and other devices can be connected to the cable, similar to the way devices are connected to a LAN. From the customer's point of view, the network boundary is the connector on NT1.



For large businesses it is common to have more telephone conversations going on simultaneously than the bus can handle. Therefore, another device, NT2, called a PBX, connected to NT1 and providing the real interface for telephones, terminals and other equipment. An ISDN PBX is not very different from an ISDN switch.

The key idea behind ISDN is that of the digital bit pipe between the customer and the carrier through which bits flow in both directions. Whether the bits originate from a digital telephone, a digital terminal, a digital facsimile machine, or some other device is irrelevant.

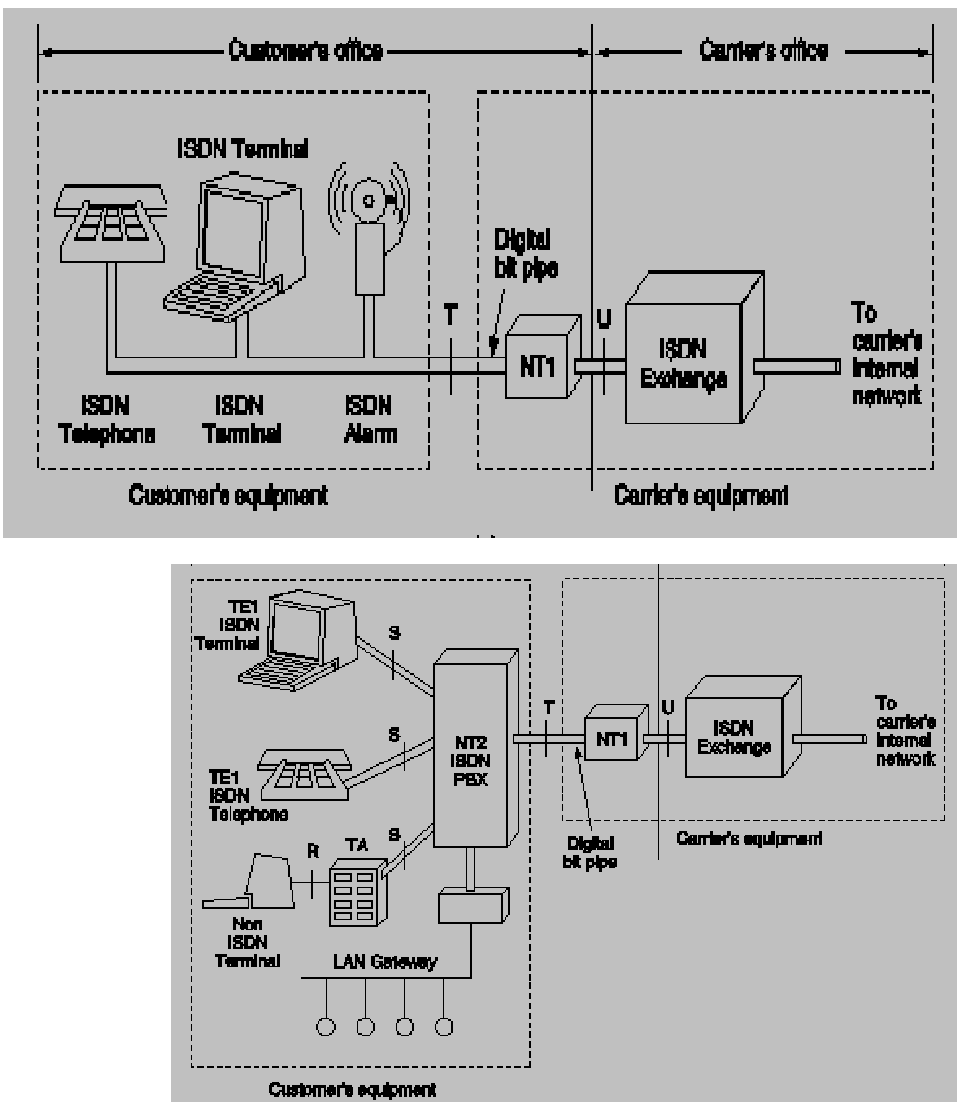
The digital bit pipe can support multiple independent channels by time division multiplexing of the bit stream. Two principal standards for the bit pipe have been developed:

- a low bandwidth standard for home use, and

- a higher bandwidth standard for business use that supports multiple channels identical to the home use channels.

Normal configuration for a home consists of a network terminating device NT1 (Fig. 2-41(a)) placed on the customer's premises and connected to the ISDN exchange in the carrier's office using the twisted pair previously used to connect the telephone. The NT1 box has a connector into which a bus cable can be inserted. Up to 8 ISDN telephones, terminals, alarms, and other devices can be connected to the cable. From the customer's point of view, the network boundary is the connector on NT1.

Fig*(a) Example ISDN system for home use. (b) Example ISDN system with a PBX for use in large businesses.*

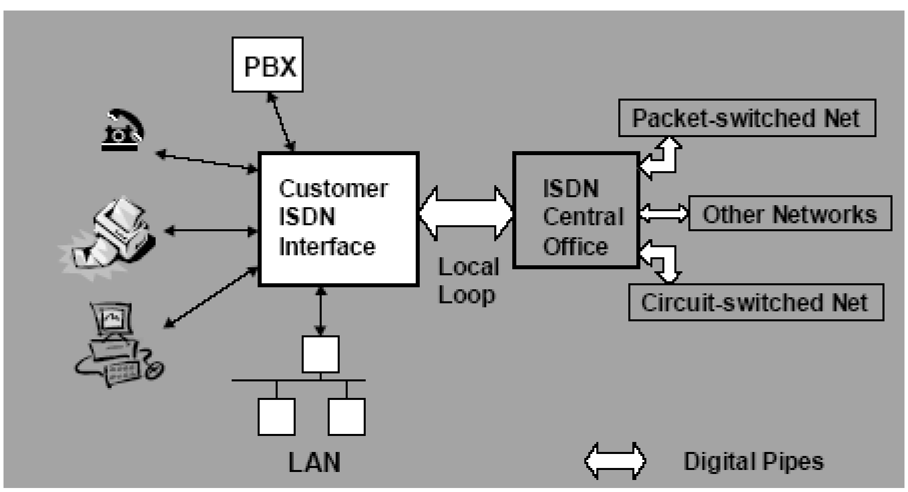


For large businesses, the model of Fig.(b) is used. There is a device NT2 called PBX (Private Branch eXchange - conceptually the same as an ISDN switch) there connected to NT1 and providing the interface for ISDN devices.

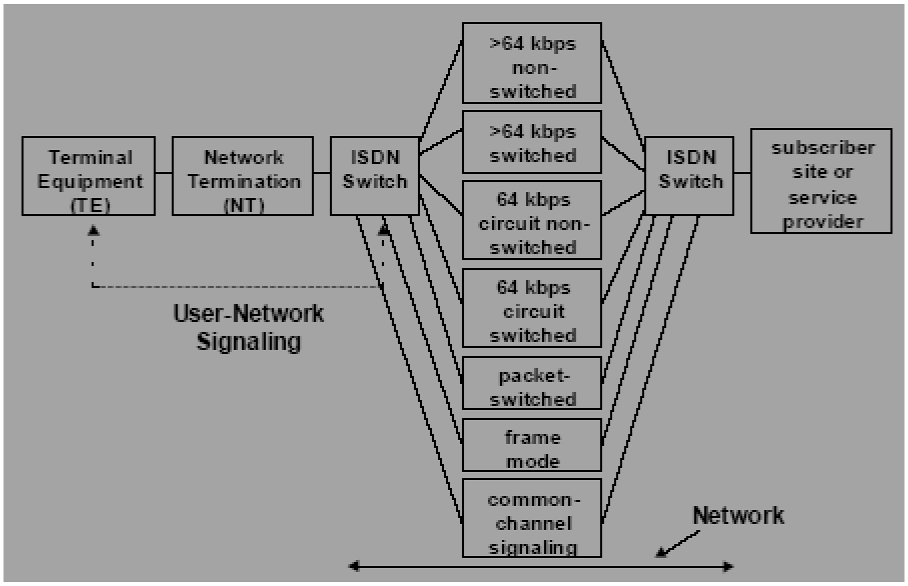
CCITT defined four reference points (Fig):

- U reference point = connection between the ISDN exchange and NT1,
- T reference point = connector on NT1 to the customer,
- S reference point = interface between the ISDN PBX and the ISDN terminal,
- R reference point = the connection between the terminal adapter and non-ISDN terminal.

ISDN Conceptual View



ISDN Architecture



Circuit-switched capabilities : 64-kbps

• Non-switched capabilities : 64 kbps dedicated link,higher data rate provided by BISDN

• Switched capabilities : > 64 kbps switched connections using ATM as part of BISDN

• Packet-switching capabilities : as provided by other data networks

• Frame-mode capabilities : supporting frame relay

• Common-channel signaling capabilities : used to control the network and provide call management.Internal to the network, SS7 is used.

2.5 ISDN Channels

Subscriber access to ISDN is via digital channels, of which there are three types:

*** B channels** are used for carrying user data (digitized voice or computergenerated data) at 64 kbps. This data rate is more than necessary in many situations (e.g., compressed digitized voice can be transmitted using less bandwidth). Consequently, a B channel is sometimes subdivided into smaller subchannel. Whether there is a subdivision or not, the network treats the whole thing as one channel. All subchannels therefore are between the same two endpoints and follow the same route.

*** D channels** are primarily used for common channel signaling purposes. They are typically associated with B channels and carry the control signals for B channel calls. D channels are also used for packet-switched data communication. A D channel may operate at 16 or 64 kbps.

*** H channels** are used in a high-speed trunk capacity. They are suitable for applications that require higher than 64 kbps data rates. Multi-media applications (e.g., audio, video, and graphics multiplexed over the same channel) are examples. H channels are divided into three categories depending on their speed:

* H0 operates at 384 kbps (= 6 B channels)
* H11 operates at 1536 kbps (= 23 B channels)
* H12 operates at 1920 kbps (= 30 B channels).

Only D channels can be used for carrying signaling information. B and H channels can only be used for carrying user data.In practice, channels are offered to users in a packaged form. Two such packages have been defined: basic access and primary access. The Basic Rate Access (BRA) package (also called 2B+D) is primarily intended for residential subscribers and consists of the following:

* Two B channels
* One 16 kbps D channel
* Overhead of 48 kbps for framing, synchronization, etc.

This produces a total bit rate of 192 kbps. The channels may be used for a variety of purposes. For example, the two B channels can be used for two independent voice services, or one of them can be used for voice and the other for a data service such as fax, teletex, or remote LAN access. Modest data communication requirements (e.g., remote banking transactions) may be met by the D channel alone. Other permitted combinations for basic access are: B+D or just D.

The Primary Rate Access (PRA) package is aimed at business users with greater bandwidth requirements. Primary access comes in two configurations:

At a bit rate of 1.544 mbps (North America and Japan) and consisting of:

* 23 B channels
* One 64 kbps D channel
* Overhead of 8 kbps

At a bit rate of 2.048 mbps (Europe) and consisting of:
* 30 B channels
* One 64 kbps D channel
* Overhead of 64 kbps

As with the basic access, lower configurations are also possible, depending on requirements. Primary access can also support H channels.

Standard bit rates:

– B-channel : 64 kbps

– D-channel : 16 or 64 kbps

– H-channel : 384 (H0), 1536 (H11), 1920 (H12) kbps

• B-channel is the basic user channel

– can carry digital data, PCM-encoded digital voice,or a mixture of lower-rate traffic

– with mixed traffic, all traffic must be destined for the same end-point (carried over the same circuit)

B-channel (continued)

– supports circuit-switched, packet-switched (exchange of data via X.25) and semipermanent connections

– in the case of circuit-switched connections, common channel signaling is used

• D-channel is dual-purpose

– carries signaling information to control circuit switched calls on B-channel

– may be used to carry low-speed data applications (e.g., videotex, telemetry)

H-channel is a high-speed channel

– can be used as a single trunk or subdivided by the user

– fast fax, video, high-speed data, high-quality audio and multiplexed information streams at lower data rates

• These channel types are grouped into transmission structures that are offered as a package to the user

2.6The ISDN Interface

The ISDN bit pipe supports multiple channels interleaved by time division multiplexing. Several channel types have been standardized:

- A - 4 kHz analog telephone channel
- B - 64 kbps digital PCM channel for voice or data
- C - 8 or 16 kbps digital channel for out-of-band signaling
- D - 16 kbps digital channel for out-of-band signaling
- E - 64 kbps digital channel for internal ISDN signaling
- H - 384, 1536, or 1920 kbps digital channel.

It is not allowed to make arbitrary combination of channels on the digital pipe. Three combinations have been standardized so far:

- Basic rate: 2B + 1D. It should be viewed as a replacement for POTS (Plain Old Telephone Service). Each of the 64 kbps B channels can handle a single PCM voice channel with 8 bits samples made 8000 times per second. D channel is for signaling (i.e., to inform the local ISDN exchange of the address of the destination). The separate channel for signaling results in a significantly faster setup time.

- Primary rate: 23B + 1D (US and Japan) or 30B + 1D (Europe). It is intended for use at the T reference point for businesses with a PBX.
- Hybrid: 1A + 1C

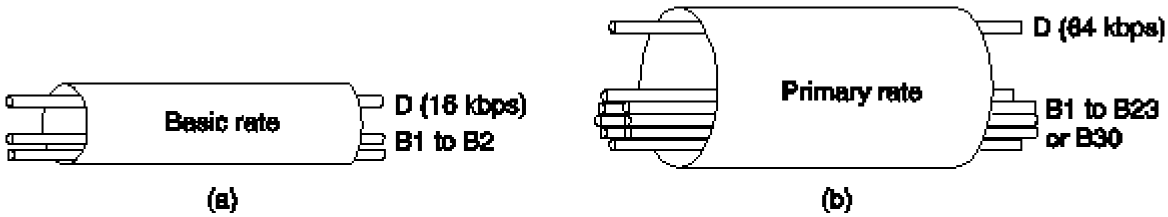
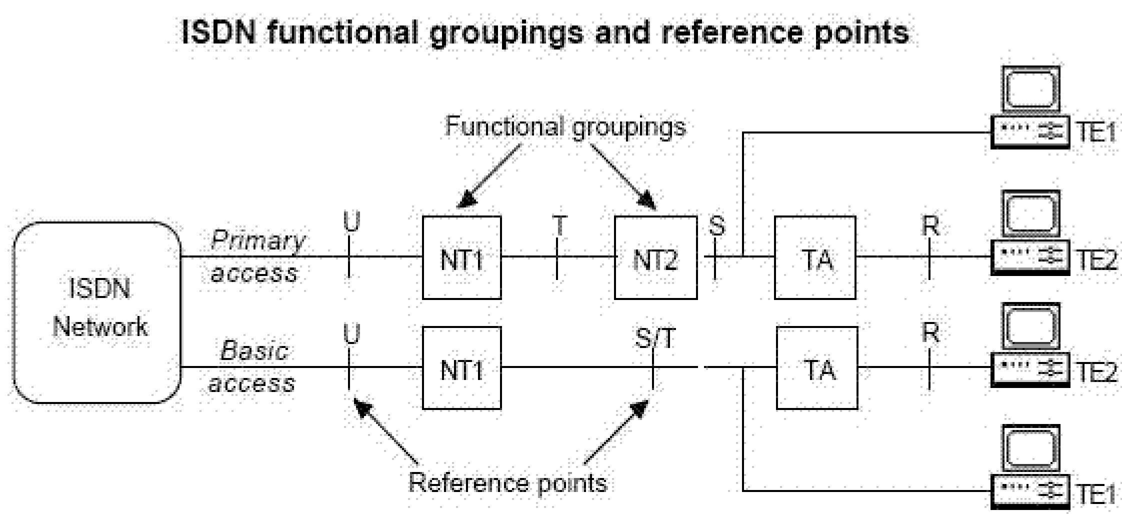


Fig. . (a) Basic rate digital pipe. (b) Primary rate digital pipe.

Because ISDN is so focused on 64 kbps channels, it is referred to as N-ISDN (Narrowband ISDN), in contrast to broadband ISDN (ATM).

Functional Groupings and Reference Points

User access to ISDN is provided at a number of different levels of abstraction. These levels are defined by functional groupings, which encompass functions equivalent to those denoted by one or more OSI layers. The interfaces between the functional groupings are called reference points.



The U (User) interface is a 2-wire physical interface to the network.

The Network Termination 1 (NT1) functional grouping provides OSI layer 1 capabilities and deals with signal transmission and physical connectors for interfacing Customer Premises Equipment (CPE) to ISDN. The NT1 transforms the U interface into a 4-wire subscriber S/T interface which supports 2B+D channels (in case of basic access) or T interface which supports 23B+D or 30B+D (in case of primary access). NT1 multiplexes these channels using TDM into a continuous bit stream for transmission over the U interface. NT1 also supports up to eight CPEs connected in a multidrop line arrangement to basic access. The NT1 device may be owned and operated by the service provider, baring the customer from direct access to the U interface, or it may be a CPE.

The Network Termination 2 (NT2) functional grouping provides additional OSI layer 2 and 3 capabilities on top of NT1. NT2 is a CPE which transforms the T (Terminal) interface into an S (System) interface. The S interface supports 2B+D channels. NT2 may perform switching and concentration functions. A typical NT2 device would be a digital PBX, serving a set of digital phones, or a LAN, serving a set of personal computers.

Two types of terminal equipment may be used for ISDN access.

Terminal Equipment 1 (TE1) denotes ISDN terminals which use a 4-wire physical link to the S or S/T interface. TE1 devices conform to ISDN standards and protocols and are especially designed for use with ISDN. A digital ISDN telephone and a PC with an ISDN card are examples.

Terminal Equipment 2 (TE2) denotes non-ISDN terminal equipment. Ordinary terminals and personal computers are examples. These devices can be connected to ISDN at the R (Rate) reference point. RS-232 and V.21 are examples of the type of standards that may be employed for the R reference point. The mapping between the R interface and the S or S/T interface is performed by a Terminal Adapter (TA), which performs the necessary protocol conversions and data rate adaptations between the two interfaces.

It is worth pointing out that although NT1, NT2, and TAs may be offered as separate devices, in practice this is not always the case. For example, some CPE manufacturers produce TAs that have NT1 and NT2 capabilities, as well as additional interfaces for other devices (e.g., analog telephones).

Reference Points

R (rate) non-IS DN equip complaying with X/V CCIT

T (terminal) minimal net config at CPE

S (system) ISDN terminals

2.7 BISDN

In the 1980s the telecommunications industry expected that digital services would follow much the same pattern as voice services did on the public switched telephone network, and conceived a grandiose vision of end-to-end circuit switched services, known as the Broadband Integrated Services Digital Network (B-ISDN). This was designed in the 1990s as a logical extension of the end-to-end circuit switched data service, ISDN.

The technology for B-ISDN was going to be Asynchronous Transfer Mode (ATM), which was intended to carry both synchronous voice and asynchronous data services on the same transport.

The B-ISDN vision has been overtaken by the disruptive technology of the Internet. The ATM technology survives as a low-level layer in most DSL technologies, and as a payload type in some wireless technologies such as WiMAX.

Broandband Integrated Services Digital Network (BISDN or Broadband ISDN) is designed to handle high-bandwidth applications. BISDN currently uses ATM technology over SONET-based transmission circuits to provide data rates from 155 to 622Mbps and beyond, contrast with the traditional narrowband ISDN (or N-ISDN), which is only 64 Kb ps basically and up to 2 Mbps.

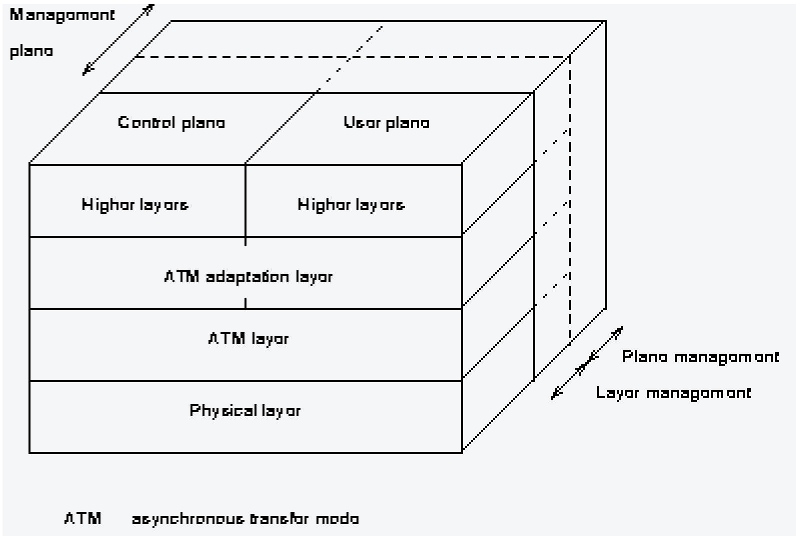
The designed Broadband ISDN (BISDN) services can be categori z ed as follows:

- Conversational services such astelephone-like services, which was also supported by N-ISDN. Also the additional bandwidth offered will allow such services as video telephony, video conferencing and high volume, high speed data transfer.
- Messaging services, which is mainly a store-and-forward type of service. Applications could include voice and video mail, as well as multi-media mail and traditional electronic mail.
- Retrieval services whichprovides access to (public) information stores, and information is sent to the user on demand only.
- No user control of presentation.This would be for instance, a TV broadcast, where the user can choose simply either to view or not.
- User controlled presentation.This would apply to broadcast information that the user can partially control.

The B-ISDN is designed to offer both connection oriented and connectionless services. The broadband information transfer is provided by the use of asynchronous transfer mode (ATM), in both cases, using end-to-end logical connections or virtual circuits. Broadband ISDN uses out-of-band signaling (as does N-ISDN). Instead of using a D Channel as in N-ISDN, a special virtual circuit channel can be used for signaling. However, B-ISDN was not widely deployed so far.

Protocol Structure - B ISDN: Broadband Integrated Services Digital Network (Broadband ISDN)

Broadband ISDN protocol reference model is based on the ATM reference model ATM adaptation layer (AAL). This layer is responsible for mapping the service offered by ATM to the service expected by the higher layers. It has two sublayers.



ATM Layer. This layer is independent of the physical medium over which transmission is to take place. It has those functions: Generic flow control (GFC) function, Cell header generation and extraction, Cell multiplex and demultiplex.

Physical layer. This consists of two sublayers: Transport Convergence (TC) and Physical medium (PM)

The management plane consists of two functions to perform layer management and plane management. The plane management is not layered as the other layers are. This is because it relies needs information on all aspects of the system to provide management facilities for the systems as a whole. The layer management provides information and control facilities for the protocol entities that exists in each individual layer. This includes operation and maintenance (OAM) functions for each layer.

The control plane is responsible for the supervision of connections, including call set-up, call release and maintenance.

The user plane provides for the transfer of user information. It also includes mechanisms to perform error recovery, flow control etc.

BISDN SERVICES

Interactive

Conversational

Messaging

Retrieval

Distribution

With User Control

Without User Control

Interactive BISDN Services

Conversational

End-to-end in both directions

User-to-user or user-to-machine

Machine-to-machine

Examples:

– *Video-telephony*

– *Video-conferencing*

– *Telemetry*

– *Internetworking*

– *Data transfer*

Messaging

• Non real-time dialog with more elaborated

features:

– Authenticated users

– Message processing, acknowledge receipt

– Digital signature

Examples:

– Electronic mail (eMail)

– Multimedia eMail: documents, sound, video.

– Electronic Document Transfer

Retrieval

• Access to information servers

Examples:

– Video-text

– Database access

– MM DB access

Distribution BISDN Services

With user control

• Examples:

– TV on demand

– Tele-text

Without user control

Examples:

– Broadcast TV channels

– Radio channels

Module 3

ATM – ATM Principles – BISDN reference model – ATM layers – ATM adaption Layer – AAL1, AAL2, AAL3/4, AAL5 – ATM addressing – UNI Signaling – PNNI Signalling

3.1ATM

Asynchronous transfer mode(ATM) is a method for multiplexing and switching that supports a broad range of services..ATM is a connection oriented packet-switching technique.

ATM combines several desirable features of packet switching and time-division multiplexing (TDM) circuit switching.

COMPARISION OF TDM AND PACKET MULTIPLEXING

	Variable bit rate	delay	Bursty traffic	Processing
TDM	Multirate only	Low, fixed	Inefficient	Minimal, very high speed
Packet	Easily handled	variable	Efficient	Header and packet Processing required

TDM versus packet multiplexing

The table compares four features of TDM and packet multiplexing.

The first comparison involves the capability to support services that generate information at a variable bit rate. Packet multiplexing easily handles variable bit rates. Because the information generated by the service is simply inserted into packets, the variable-bit-rate nature of the service translates into the generation of the corresponding packets.TDM systems transfer information at a constant bit rate. The bit rates that TDM can support are multiples of some basic rate.

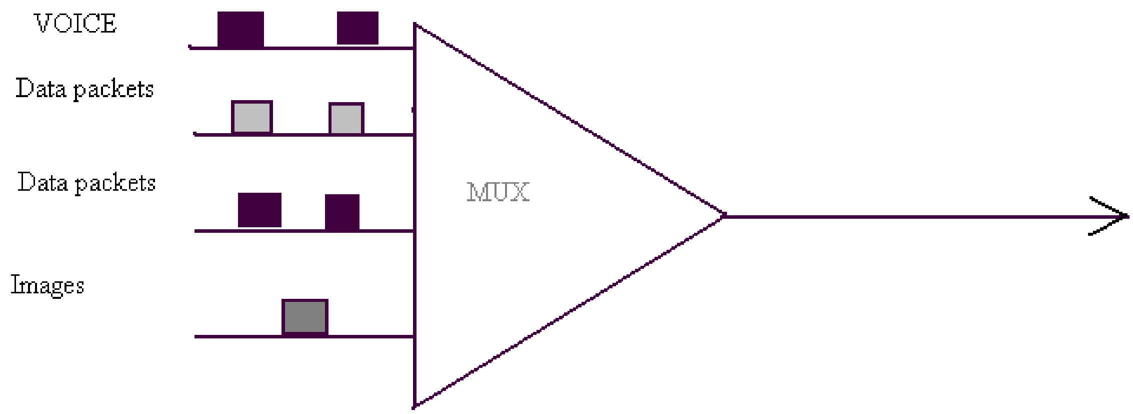
The second comparison involves the delay incurred in traversing the network. In case of TDM once a connection is set up the delays are small and nearly constant .Packet multiplexing, on the other hand, has inherently variable transfer delays because of the queuing that takes place in the multiplexers.

The third criterion for comparison is the capability to support burstly traffic.TDM dedicates the transmission resources, namely, slots, to a connection. If the connection is generating information in a bursty fashion, then many of the dedicated slots go unused. Packet multiplexing, on other hand, was developed specifically to handle burstly traffic and can do so in an efficient way.

Processing is the fourth comparison criterion. In TDM, hardware handles the transfer of slots, so the processing is minimal and can be done at very high speeds. Packet multiplexing, on the other hand, traditionally uses software to process the information in the packet headers.

ATM involves the conversion of all information flows into short fixed-length packets called cells. Cells contain abbreviated headers, or tables, which are essentially pointers to tables in the switches. ATM can easily handle services that generate information in burstly fashion or at variable bit rates. The abbreviated header of ATM and the fixed length facilitate hardware implementations that result in low delay and high speeds.

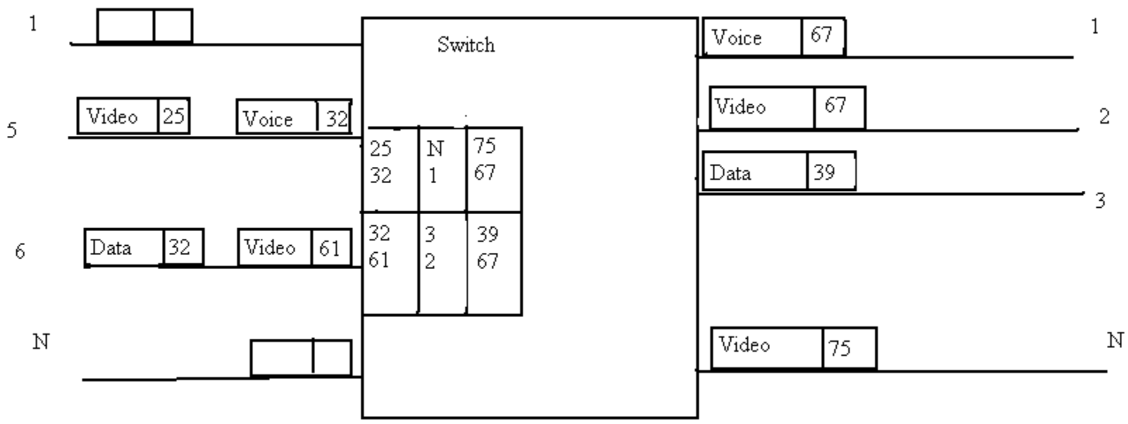
Figure shows the operation of an ATM multiplexer. The information flows generated by various users are converted into cells and sent to an ATM multiplexer. The multiplexer arranger the cells into one or more queues and implements some scheduling strategy that determines the order in which cells are transmitted. The purpose of the scheduling strategy is to provide for different qualities of service required by the different flows.



ATM multiplexing

ATM networks are connection-oriented and require a connection setup prior to the transfer of cells. The connection set up procedure requires the source to provide a traffic descriptor that describes the manner in which cells are produced for example, peak cell rate in cells/second, sustainable cell rate in cells/second, and maximum length of a burst of cells. The source also specifies a set of Quality-of-Service parameters that the connection must provide, for example cell delay and cell loss. The connection set up procedure involves identifying a path through the network that can meet these requirements. A connection admission control procedure is carried out at every multiplexer along the path. This path is called a virtual channel connection(VCC).

VCC is established by a chain of local identifiers that are defined during the connection setup at the input port to each switch between the source and the destination.



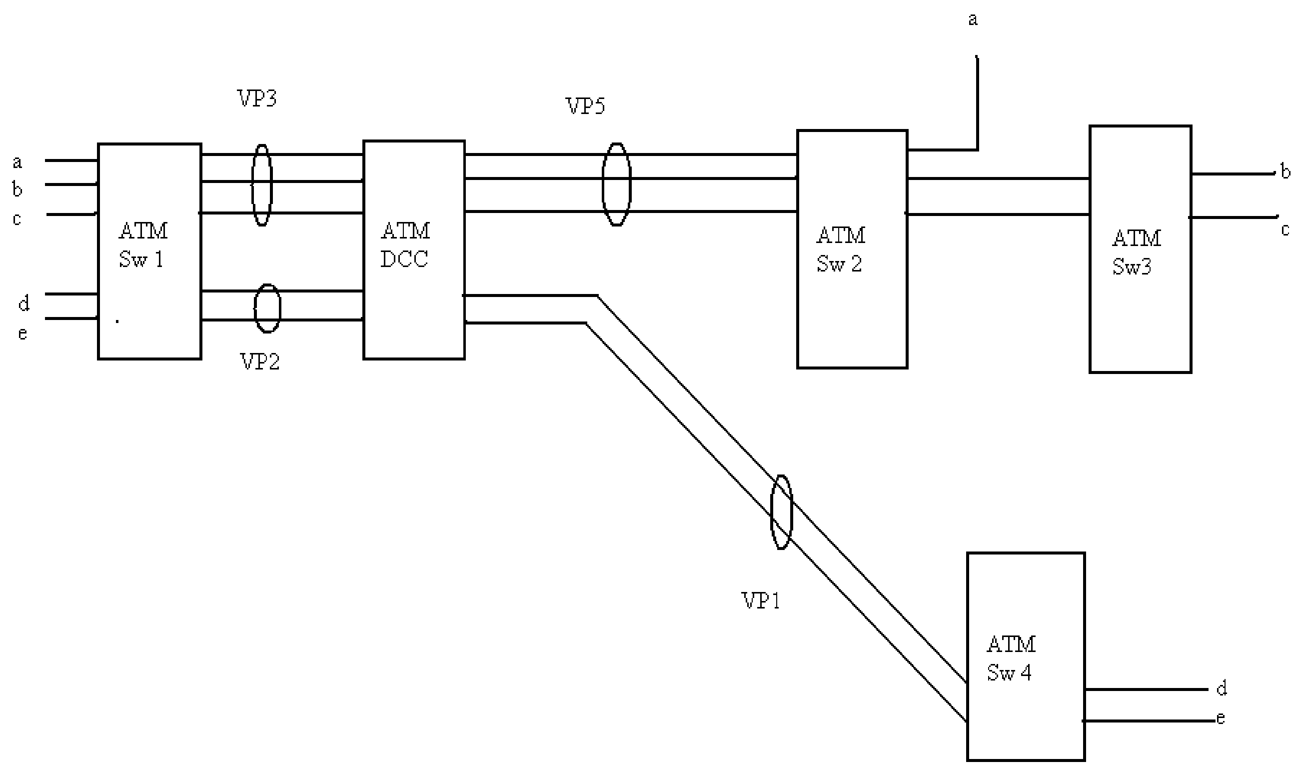
ATM switching.

Figure above show s the tables associated with two of the input ports to an ATM switch. Input port 5 we have cells from a voice stream arriving with identifier 32 in the header. We also have cells from a video stream arriving with identifier 25.When a cell with identifier 32 arrives at input port 5, the table look up for entry 32 indicates that the cell is to be switched to output port 1 and that the identifier in the header is to be changed to 67.Similarly, cells arriving at port 5 with identifier 25 are switched to output port N with new identifier 75.The identifier is locally defined for each input port. Thus input port 6 uses identifier 32 for a different VCC.

To understand how the local identifiers are defined in ATM, we first need to see how ATM incorporates some of the concepts used in SONET.SONET allows flows that have a common path through the network to be grouped together.ATM uses the concept of a virtual path to achieve this bundling.

The figure given below shows five VCCs in an ATM network.

The VCCs a, b and c enter the network at switch 1,share a common path up to switch 2, and are bundled together into a virtual path connection (VPC) that connects switch 1 to switch 2.This VPC happens to pass through an ATM cross-connect switch whose role in this example is to switch only virtual paths. The VPC that contains VCCs a,b and c has been given virtual path identifier (VPI) 3 between switch 1 and the cross-connect.The cross connect switches all cells with VP1 3 to the link connecting it to switch number 2 and changes the VP1 to 5 ,which identifies the virtual path between the cross-connect and ATM switch 2.This VPC terminates at switch 2 where the three VCCs are unbundled; cells from VCC a are switched out to a given output port, whereas cells from VCCs b and c proceed to switch 3.Figure also shows VCCs d and e entering at switch 1 with a common path to switch 4.These two channels are bundled together in a virtual path that is identified by VP2 between switch 1 and the cross-connect and by VPI 1 between the cross-connect and switch 4. . that is identified by VP2 between switch 1 and the cross-connect and by VPI 1 between the cross-connect and switch 4.



Role of virtual paths in an ATM network

The preceding discussion clearly shows that a virtual circuit in ATM requires two levels of identifiers: an identifier for the VPC, the VPI and a local identifier for the VCC the so called virtual channel identifier,VCI.

The figure below shows a cross section of the cell stream that arrives at a given input port of an ATM switch or a cross-connect. The cells of specific VCC are identified by a two –part identifier consisting of a VPI and a VCI.VCCs that have been bundled into a virtual path have the same VPI, and their cells are switched in the same manner over the entire length of the virtual path. At all switches along the virtual path, switching is based on the VPI only and the VCIs are unchanged. The VCIs are used and translated only at the end of the virtual path.

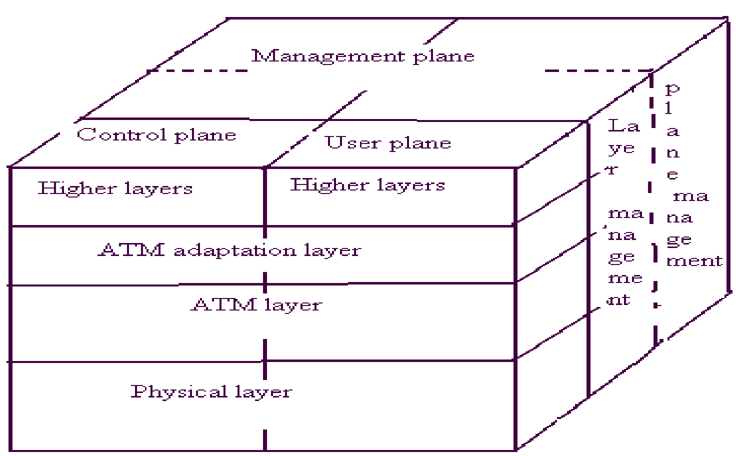
Advantages of ATM networks

1. The network infrastructure and its management is simplified by using a single transfer Mode for the network and extensive bandwidth management capabilities have been built into ATM architecture.
2. Unlike shared media networks, ATM is not limited by speed or distance; the switched nature of ATM allows it to operate over LANs as well as global backbone networks at speeds ranging from a few Mbps to several Gbps.
3. The QoS attributes of ATM allow it to carry voice, data and video, thus making ATM Suitable for an integrated services network.

3.2 ATM principles:

- small (48 byte payload, 5 byte header) fixed length *cells* (like packets)
    - fast switching
    - small size good for voice
- virtual-circuit network: switches maintain state for each "call"
- well-defined interface between "network" and "user" (think of telephone company)

3.3BISDN REFERENCE MODEL



The model contain three planes

- The user plane

- The control plane.
- The management plane.

The user plane is concerned with the transfer of user data including flow control and error recovery.

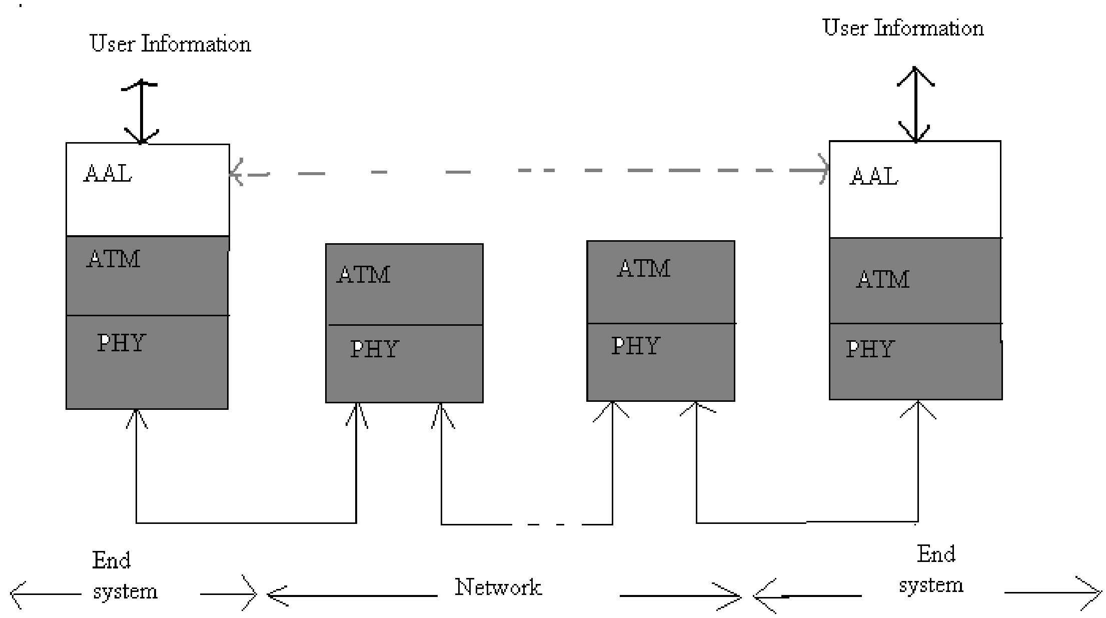
The control plane deals with the signaling required to set up, management and release connections.

The management plane is split into a layer management plane that is Concerned with the management of network resources and a plane management Plane that deals with the coordination of the other planes.

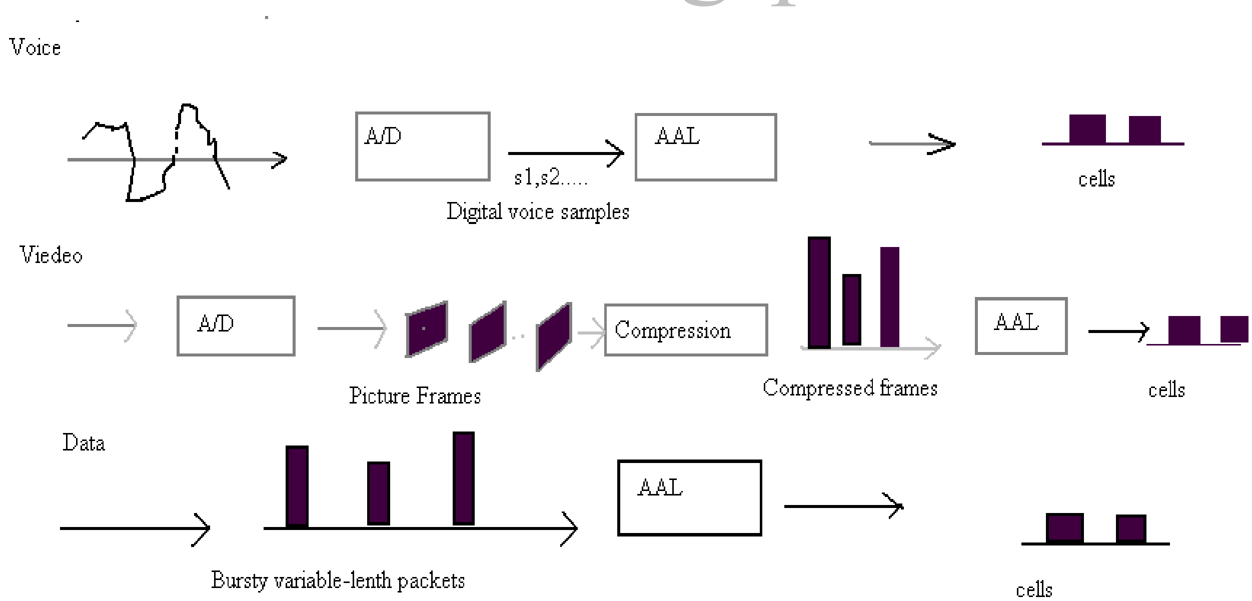
The user plane has three basic layers that together provide support for user applications:

- The ATM adaptation layer(AAL)
- The ATM layer.
- The physical layer.

The ATM adaptation layer is responsible for providing different applications with appropriate support. The AAL is also responsible for the conversion of the higher-layer service data units (SDUs) into 48 – byte blocks that can be carried inside ATM cells. The information generated by voice, data and video applications are taken by AALs and converted into sequences of cells that can be transported



User Plane layers



The AAL converts user information into cells by the ATM network The AAL entities reside in the terminal equipment , and hence they communicate on an end to-end basis across the ATM network.

The ATM layer is concerned with the sequenced transfer of ATM cells in connections set up across the network. The ATM layer accepts 48-byte blocks of information from the AAL and adds a 5-byte header to form the ATM cell.

In terms of number of users involved ATM supports two types of connections:

- Point to point connections—can be unidirectional or bidirectional
- Points to multi point connections are always unidirectional.

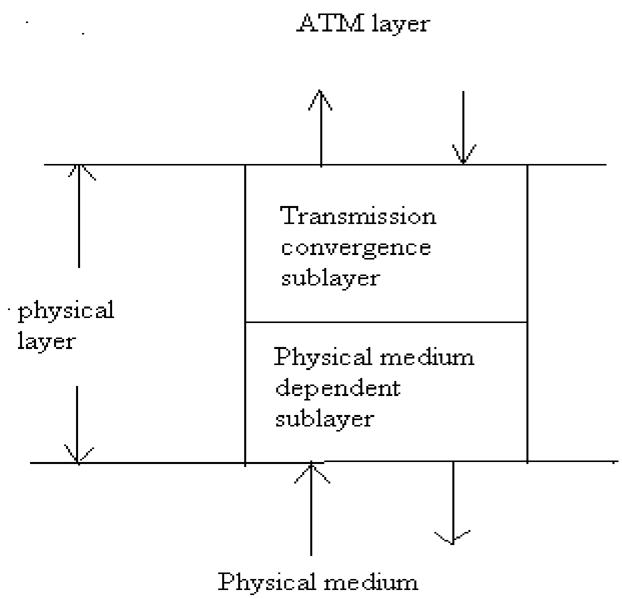
In terms of duration, ATM provides

- Permanent virtual connections (PVCs) –it act as permanent lines between user sites.
- Switched virtual connections (SVCs) —set up and released on demand of end user.

The role of control plane is to support signaling and network control applications. The control plane also has the same three basic layers as user plane signaling AAL has been defined for the control plane to provide for the reliable exchange of messages between ATM systems.

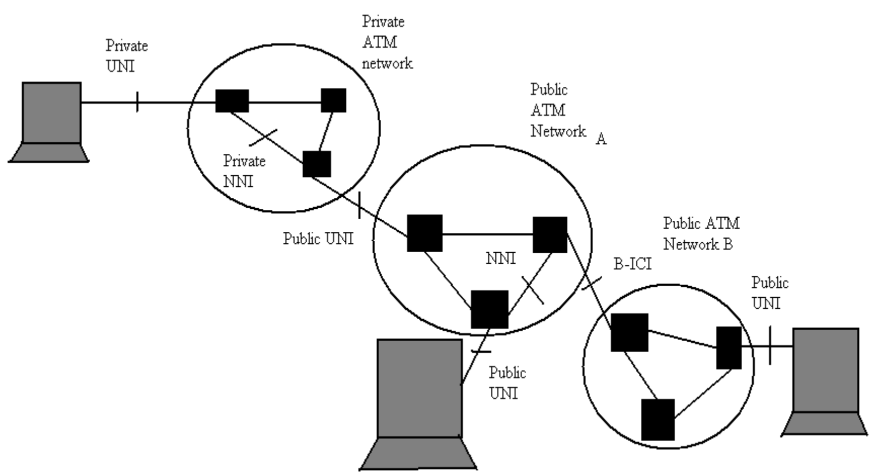
The physical layer is divided into two sub layers

- The physical medium dependent sub layer—concerned with details of the transmission of bits over the specific medium such as line coding, timing recovery pulse shape as well as connectors.
- The transmission convergence sub layer –establishes and maintains the boundaries of the ATM cells in the bit stream; generates and verifies header checksums, inserts and removes ATM cells.



ATM physical layer

ATM network interfaces.



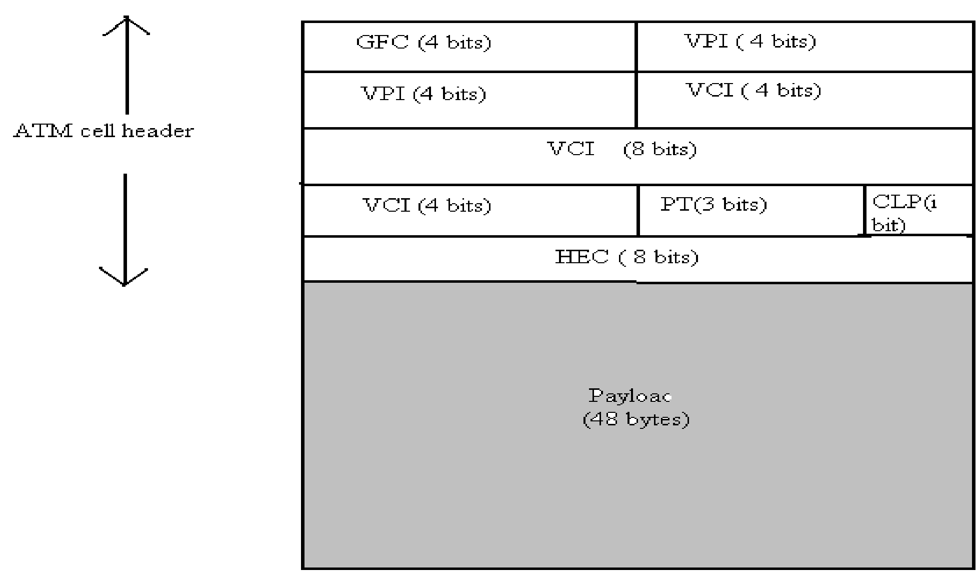
ATM network interfaces

Initially the source user must interact with the network through a user-network interface(UNI).The connection request must propagate across the network and eventually involve an interaction at the destination UNI.With in a network ,switches must interact across the network-network interface (NNI) to exchange information. Switches that belong to different public networks communicate across a broadband inter carrier interface (B-ICI).The source and destination end systems as well as all switches along the path across the network are eventually involved in the allocation of resources to meet the QoS requirements of a connection.

3.4 ATM LAYER

ATM layer is concerned with sequenced transfer of cells of information across connections established through the network.

ATM Cell Header



ATM cell header format

ATM involves converting all traffic that flows in the network into 53-byte blocks called cells. Each cell has 48 bytes of payload and a 5-byte header that allows the network to forwards each cell to its destination.

Generic flow control: The GFC field is 4 bits long and was intended to provide flow control and shared medium access to several terminals at the UNI.The GFC field has significance only at the UNI and is not carried end to end across the network.

Virtual path identifier: The VPI field is 8 bits long, so it allows the definition of up to 2^8=256 virtual paths in a given UNI link.

Virtual channel identifier: The VCI field is 16 bits long, so it allows the definition of up to 2^{16}=65,536 virtual channels per virtual path. The VIP/VCI field is the local identifier for a given connection in a given link, and the value of the field changes at every switch.

Payload type: The 3-bit payload type field allows eight types of ATM payloads .The most significant bit is used to distinguish between data cells (b_3=0)and operations, administration and maintenance (OAM)cells (b_3=1).

For data cells (b_3=0), the second bit serves as the explicit forward congestion indication, which is set by switches to indicate congestion and is used by the congestion control mechanism for the available bit rate(ABR.)

Cell loss priority: The CLP bit establishes two levels of priorities for ATM cells. A cell that has CLP =0 is to be treated with higher priority than a cell with CLP =1 during periods of congestion. In particular, CLP = 1 cells, should be discarded before CLP =0 cells.

Header error control: An 8-bit CRC checksum, using the generator polynomial is calculated over the first four bytes of the header. This code can correct all single errors and detect all double errors in the header. The checksum provides protection against misdelivery of cells from errors that may occur in transit.

Q o S Parameters

A central objective of ATM is to provide QoS guarantees in the transfer of cell streams across the network.

The following three QoS network performance parameters are defined in ATM standards.These are indicators of the intrinsic performance of a given network.

Cell error ratio: The CER of a connection is the ratio of the number of cells that are delivered with one or more bit errors during a transmission to the total number of transmitted cells.It is dependent on the underlying physical medium.

Cell misinsertion rate:The CMR is the average number of cells/second that are delivered mistakenly to a given connection destination.The CMR depends primarily on the rate at which undetected header errors result in misdelivered cells.The CMR calculation excludes blocks of cells that are severely errored.

Severely-errored cell block ratio:- A severely errored cell block event occurs when more than M cells are lost, in error, or misdelivered in a given received block of N cells, where M and N are defined by the network provider. The severely-errored cell block ratio (SECBR) is the ratio of severely-errored cell blocks to total number of transmitted cell blocks in a connection.

The following three QoS parameters may be negotiated between the user and the network during connection setup.

Cell loss ratio: The CLR for a connection is the ratio of the number of lost cells to total number of transmitted cells. It is specified as an order of 10^{-1} to 10^{-15} .The degree to which CLR can be negotiated depends on the sophistication of the buffer – allocation strategies that are available in a given network.

Cell transfer delay:- The CTD is the time that elapses from the instant when a cell enters the network at the source UNI to the instant when it exists at the destination UNI.The CTD includes propagation delay, processing delays, and queueing delays at multiplexers and switches.

Cell delay variation : The CDV measures the variability of total delay encountered by cells in a connection.The CDV excludes the fixed component D_0 of the CTD that is experienced by all cells in a connection, for example propagation delay and fixed processing delays.

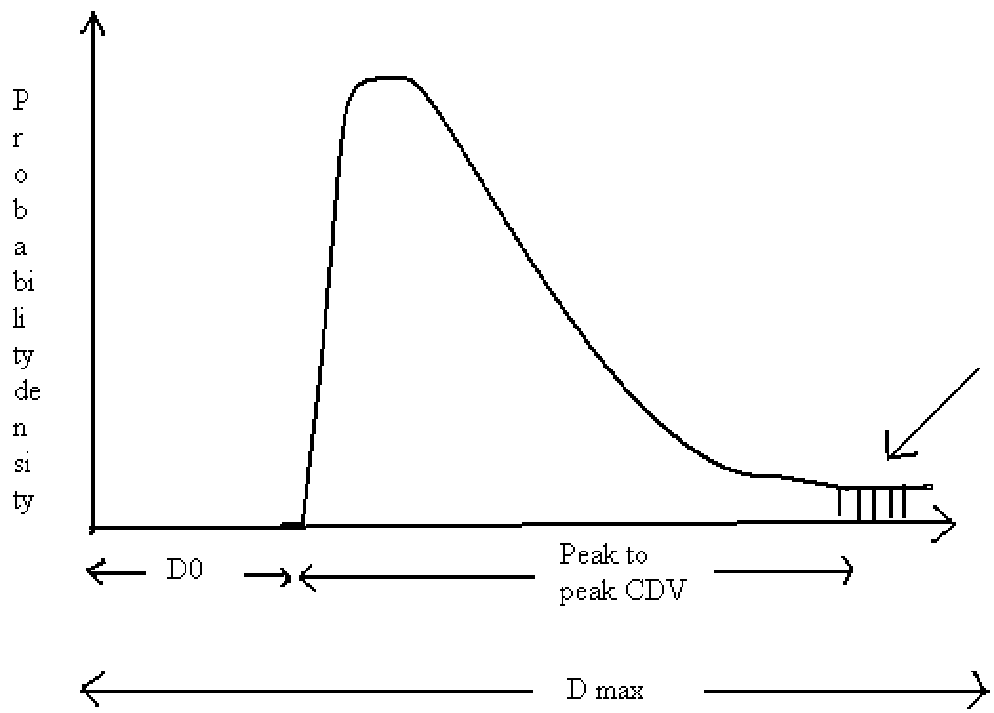
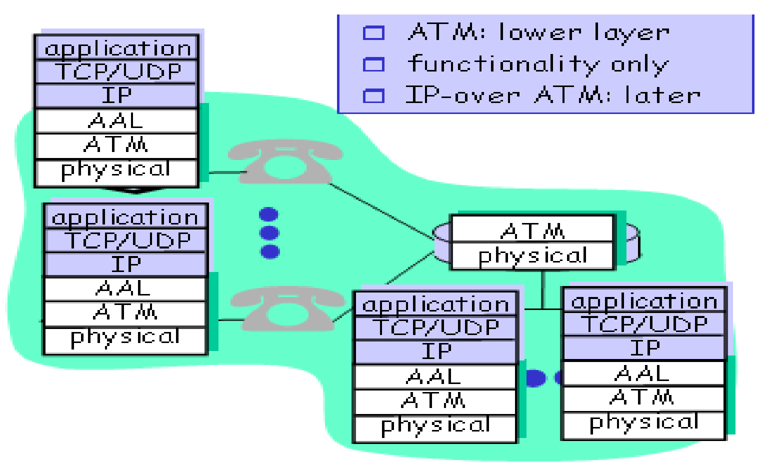


Fig 10.Probability density function of cell transfer delay

- ATM Adaptation Layer (AAL): interface to upper layers
  - end-system
  - segmentation/reassembly
- ATM Layer: cell switching
- Physical



3.5 ATM Adaptation Layer

The ATM Adaptation Layer (AAL) relays ATM cells between ATM Layer and higher layer. When relaying information received from the higher layers, it segments the data into ATM cells. When relaying information received from the ATM Layer, it must reassemble the payloads into a format the higher layers can understand. This operation, which is called Segmentation and Reassembly (SAR), is the main task of AAL. Different AALs were defined in supporting different traffic or service expected to be used. The service classes and the corresponding types of AALs were as follows:

Class A - Constant Bit Rate (CBR) service: AAL1 supports a connection-oriented service in which the bit rate is constant. Examples of this service include 64 Kbit/sec voice, fixed-rate uncompressed video and leased lines for private data networks.

Class B - Variable Bit Rate (VBR) service: AAL2 supports a connection-oriented service in which the bit rate is variable but requires a bounded delay for delivery. Examples of this service include compressed packetized voice or video. The requirement on bounded delay for delivery is necessary for the receiver to reconstruct the original uncompressed voice or video.

Class C - Connection-oriented data service: For connection-oriented file transfer and in general, data network applications where a connection is set up before data is transferred, this type of service has variable bit rate and does not require bounded delay for delivery. Two AAL protocols were defined to support this service class, and have been merged into a single type, called AAL3/4. But with its high complexity, the AAL5 protocol is often used to support this class of service.

Class D - Connectionless data service: Examples of this service include datagram traffic and in general, data network applications where no connection is set up before data is transferred. Either AAL3/4 or AAL5 can be used to support this class of service.

Operation Administration and Maintenance (OA&M) - OA&M is defined for supervision, testing, and performance monitoring. It uses loop-back for maintenance and ITU TS standard CMIP, with organization into 5 hierarchical levels: Virtual Channel (F5 - Between VC endpoints), Virtual Path (F4- Between VP endpoints), Transmission Path (F3- Between elements that perform assembling, disassembling of payload, header, or control), Digital Section (F2 Between section end-points, performs frame synchronization) and Regenerator Section (F1- Between regeneration sections).

The use of Asynchronous Transfer Mode (ATM) technology and services creates the need for an adaptation layer in order to support information transfer protocols, which are not based on ATM. This adaptation layer defines how to segment and reassemble higher-layer packets into ATM cells, and how to handle various transmission aspects in the ATM layer.

Examples of services that need adaptations are Gigabit Ethernet, IP, Frame Relay, SONET/SDH, UMTS/Wireless, etc.

The main services provided by AAL (ATM Adaptation Layer) are:

- Segmentation and reassembly
- Handling of transmission errors
- Handling of lost and misinserted cell conditions
- Timing and flow control

The following ATM Adaptation Layer protocols (AALs) have been defined by the ITU-T. It is meant that these AALs will meet a variety of needs. The classification is based on whether a timing relationship must be maintained between source and destination, whether the application requires a constant bit rate, and whether the transfer is connection oriented or connectionless.

- **AAL Type 1** supports constant bit rate (CBR), synchronous, connection oriented traffic. Examples include T1 (DS1), E1, and x64 kbit/s emulation.
- **AAL Type 2** supports time-dependent Variable Bit Rate (VBR-RT) of connection-oriented, synchronous traffic. Examples include Voice over ATM. AAL2 is also widely used in wireless applications due to the capability of multiplexing voice packets from different users on a single ATM connection.
- **AAL Type 3/4** supports VBR, data traffic, connection-oriented, asynchronous traffic (e.g. X.25 data) or connectionless packet data (e.g. SMDS traffic) with an additional 4-

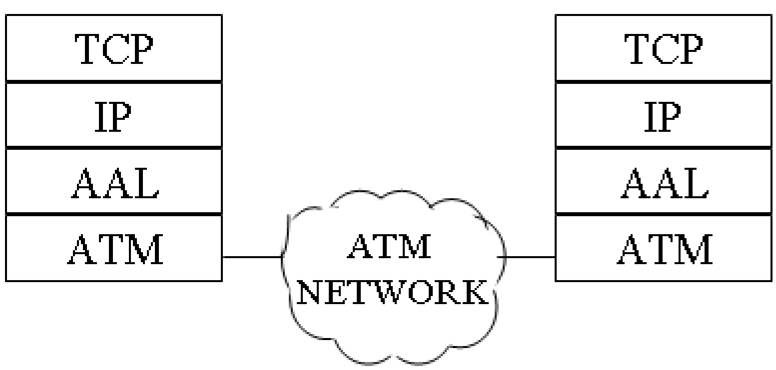
byte header in the information payload of the cell. Examples include Frame Relay and X.25.

- **AAL Type 5** is similar to AAL 3/4 with a simplified information header scheme. This AAL assumes that the data is sequential from the end user and uses the Payload Type Indicator (PTI) bit to indicate the last cell in a transmission. Examples of services that use AAL 5 are classic IP over ATM, Ethernet Over ATM, SMDS, and LAN Emulation (LANE). AAL 5 is a widely used ATM adaptation layer protocol. This protocol was intended to provide a streamlined transport facility for higher-layer protocols that are connection oriented.

AAL 5 was introduced to:

- reduce protocol processing overhead.
- reduce transmission overhead.
- ensure adaptability to existing transport protocols.

The AAL 5 was designed to accommodate the same variable bit rate, connection-oriented asynchronous traffic or connectionless packet data supported by AAL 3/4, but without the segment tracking and error correction requirements.



The following ATM Adaptation Layer protocols (AALs) have been defined for Asynchronous Transfer Mode. These protocols are loosely associated with the various classes of traffic expected to be carried:

AAL 1

Supports constant bit rate, connection-oriented, synchronous traffic (e.g., uncompressed voice).

AAL 2

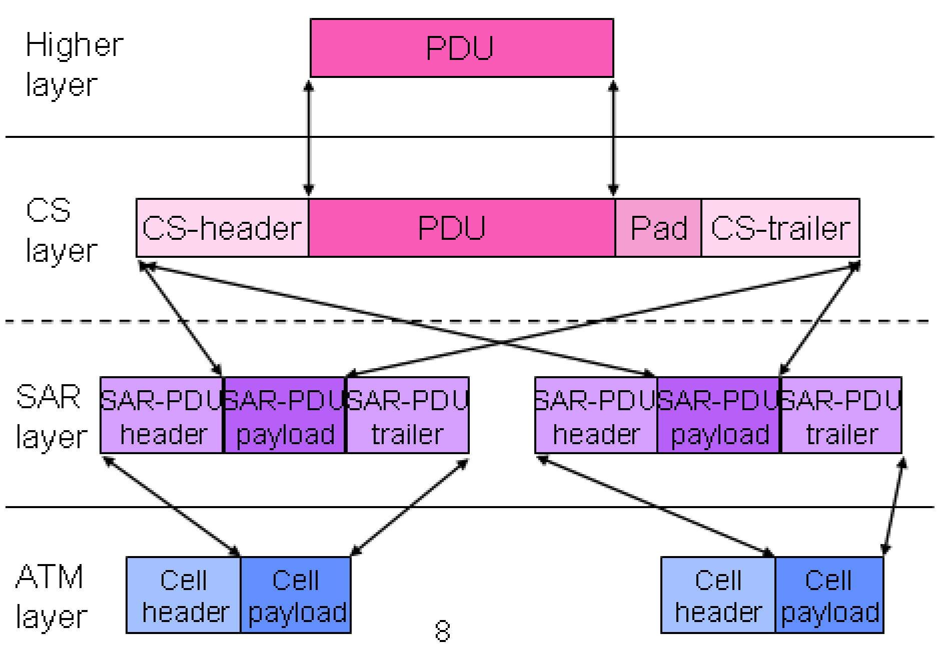
Definition never completed, but was envisioned to be assigned for variable bit rate, connection-oriented, synchronous traffic.

AAL 3/4

Supports variable bit rate, connection- oriented, asynchronous traffic (e.g., X.25 data) or connectionless packet data (e.g., SMDS traffic) with an additional 4-byte header in the information payload of the cell.

AAL 5

Similar to AAL 3/4 with a simplified information header scheme; this AAL assumes that the data is sequential from the end user and uses the PTI bit to indicate the last cell in a transmission Examples of services that use AAL 5 are Classic IP over ATM, and LAN Emulation (LANE). AAL 5 is the most widely used ATM Adaptation Layer Protocol.



AAL TYPE 1

This section addresses ATM Adaptation Layer protocol type 1 (AAL 1), which is designed to accommodate constant bit rate, connection-oriented, synchronous traffic. A good example of this type of traffic is uncompressed video.

Like all of the AAL protocols, AAL 1 is designed to accept the data from the upper-layers (i.e., USER layers) of the end user and prepare it for handoff in 48-byte segments to the ATM Layer of the ATM protocol stack. The ATM Layer then provides each segment with an appropriate header and hands the 53-byte ATM cell down to the Physical Layer for transmission over the ATM network.

The preparation offered by the AAL 1 protocol includes support for the following functions required by the upper layer:

- Timing recovery
- Timing indication
- Synchronization
- Indication of information loss

In order to provide these services, AAL 1 adds a one-byte (8 bit) header to each Protocol Data Unit segment generated at the Segmentation and Reassembly Sublayer. This information includes Convergence Sublayer specific information (when required) as well as an error checking mechanism. No trailer is added with this AAL protocol type.

AAL 1 Data Flow

```
                                                         LAYER
+-----------------------------------------------+
|               Service Data Unit               | ----- USER
+-----------------------------------------------+
                       V
+-----------------------------------------------+ ---+
|  Convergence Sublayer Protocol Data Unit      |    |
+-----------------------------------------------+    |
                       V                             |
+--------------++--------------++---------------+    |
```

```
    |   SAR-PDU    ||   SAR-PDU    ||   SAR-PDU    |    |- AAL
    +--------------++--------------++--------------+    |
     \  47 Bytes    \                \  47 Bytes    \   |
  +---+---------------+           +---+---------------+  |
  | H |   SAR-PDU     |           | H |   SAR-PDU     |  |
  +---+---------------+           +---+---------------+ -+
                                  V      48 Bytes    V
  H = SAR HEADER             +-----------------------+
 AH = ATM HEADER             | AH |    ATM PAYLOAD   | --- ATM
```

AAL 1 SAR Header Format

The 1 byte (8 bit) header added at the Segmentation and Reassembly Sublayer is divided into two 4-bit fields:

- Segment Number (SN)
- Sequence Number Protection (SNP)

The SN field is further subdivided into the 1-bit Convergence Sublayer Indication (CSI) subfield and the 3-bit Sequence Count (SC) field. The CSI subfield is used to convey Convergence Sublayer-specific information and is not used for all AAL implementations. The SC subfield carries the sequence number for the entire CS_PDU; this is generated by the Convergence Sublayer.

The other field (SNP) functions as the error check for the SAR header. The first three bits of this field are a Cyclical Redundancy Check (CRC) for the SN portion of the header. The polynomial used to generate this CRC is $x^3 + x + 1$

The remaining bit is an even parity check over the previous 7 bits (i.e., the SN field and the CRC subfield).

These fields are illustrated below:

```
          +-------------+-----------------------+
          | SAR HEADER  |        SAR-PDU        |
          +-------------+-----------------------+
          |             |
          +------+------+
          |  SN  |  SNP |        EPC = Even
          +------+------+              Parity
        /        |        \            Check
      +---+------+------+---+
```

```
          |CSI|  SC  | CRC  |EPC|
          +---+------+------+---+
            ^     ^     ^     ^
            |     |     |     |
   1 Bit ---+     |     |     +--- 1 Bit
         3 Bits --+     +-- 3 Bits
```

AAL TYPE 3/4

This section addresses ATM Adaptation Layer protocol type 3 and protocol type 4. These two types are merged into a single protocol, AAL 3/4, which is designed to accommodate both variable bit rate, asynchronous traffic (type 3) or connectionless packet data (type 4). Good examples of these types of traffic are X.25 Pacekt-Switched Data and SMDS respectively.

Like all of the AAL protocols, AAL 3/4 is designed to accept the data from the upper-layers (i.e., USER layers) of the end user and prepare it for handoff in 48-byte segments to the ATM Layer of the ATM protocol stack. The ATM Layer then provides each segment with an appropriate header and hands the 53-byte ATM cell down to the Physical Layer for transmission over the ATM network.

AAL 3/4 provides for full sequencing as well as error-control and error-recovery mechanisms. This is accomplished both at the Convergence Sublayer and the Segmentation and Reassembly Sublayer of the AAL Layer.

AAL 3/4 DATA FLOW

The first step in the data flow for AAL 3/4 is the Service Data Unit (SDU) being passed down from the Upper (User) Layer to the Convergence Sublayer of the AAL Layer.

AAL 3/4 subdivides the Convergence Sublayer into the Service Specific Convergence Sublayer (SSCS) and the Common Part Convergence Sublayer (CPCS). The Upper Layer SDU is passed through the SSCS Sublayer unchanged.

The CPCS Sublayer adds a 4-byte header and a 4-byte trailer; at this point, the User SDU is also padded, with 0-3 bytes as necessary, to assure that the entire CPCS Protocol Data Unit (PDU) length is divisible by 4.

The CPCS-PDU is then passed down to the Segmentation and Reassembly (SAR) sublayer of the AAL Layer. At this point, the entire CPCS-PDU, including the added header, trailer, and pad, is segmented in 44-byte pieces. The SAR then adds a 2-byte header and a 2-byte trailer to each segment; this 48-byte SAR-SDU segment is passed down to the ATM Layer for processing.

The ATM Layer adds the 5-byte ATM header to each segment received from the SAR and passes each segment to the Physical Layer for transport over the ATM network.

This process is illustrated below:

```
     +-----------------------------------------------+
     |              Service Data Unit                |
     +-----------------------------------------------+
     |                                               |
     V                                               V
+----+-----------------------------------------------+----+----+
| CH |  Convergence Sublayer Protocol Data Unit      | PAD| CT |
+----+-----------------------------------------------+----+----+
|                                                              |
V                                                              V
+--------------------++--------------------++------------------+
|    |               ||                    ||          |    |  |
+--------------------++--------------------++------------------+
|     Segment 1      ||     Segment 2      ||     Segment 3    |
V       (BOM)        VV       (COM)        VV       (EOM)      V

CH = CPCS Header  CT = CPCS Trailer   BOM = Beginning of Message
  COM = Continuation of Message  EOM = End of Message
++-------------------++--------------------+
||                   ||          |    |    |
++-------------------++--------------------+
||  Segment 2 (COM)  ||  Segment 3 (EOM)   |
VV                   VV                    V
                 +---+--------------------+---+
                 | H |          |    |    | T |
                 +---+--------------------+---+
                 |          SAR-SDU           |
                 V                            V
           +-----+----------------------------+
           |  AH |   |          |    |    |   |
           +-----+----------------------------+
                 |_______  ATM Payload ______|

           |_______  53-Byte ATM Cell _______|

H = SAR Header   T = SAR Trailer  AH = ATM Header
AAL 3/4 CPCS-PDU FORMAT
+----+-----------------------------------------------+----+----+
| CH |  Convergence Sublayer Protocol Data Unit      | PAD| CT |
+----+-----------------------------------------------+----+----+
```

```
|      \                                               /      |
|         \                                         /         |
|            \                                   /            |
+---+----+------+                              +--+----+------+
|CPI|Btag|BASize|                              |AL|Etag|Length|
+---+----+------+                              +--+----+------+
```

The Convergence Sublayer adds a 4-byte header, a 4-byte trailer, and extra bytes as necessary to maintain the size requirements for even ATM segmentation.

AAL 3/4 CPCS-PDU FORMAT

The CPCS header that is added to the data passed down from the Upper Layer (end user) consists of the following three fields:

CPI

Common Part Indicator. This 1-byte field is used to interpret the remainder of the fields in the header and trailer added for this sublayer.

Btag

Beginning Tag. This 1-byte field acts as an error check for this segment. The value in this field is also placed in the Etag (End Tag) field of the trailer, allowing a quick comparison after receipt to determine if the segment has been corrupted.

BASize

Buffer Allocation Size. This 2-byte field is encoded to indicate the length of the CPCS-PDU (segment) payload.

The CPCS trailer also is divided into three fields:

AL

Alignment. This 1-byte field is used to make the trailer size 4 bytes, in order to maintain the divisibility of the segment by 4. It is passed through the network transparently.

Etag

This 1-byte field contains the same information as the Btag field in the header. (See Btag above.)

Length

This 2-byte field indicates the length of the PDU payload. It is encoded to indicate the number of COUNTING UNITS in the length of the payload, with the counting unit size indicated in the CPI of the header.

AAL 3/4 SAR-SDU FORMAT

```
                 +---+-------------------+---+
                 | H |    SAR Payload    | T |
                 +---+-------------------+---+
               /     |               /       |
            /        |            /          |
         /           |         /             |
+----+----+------+          +-----+--------+
| ST | SN | MID  |          |  LI | CRC-10 |
+----+----+------+          +-----+--------+
```


The Segmentation and Reassembly header and trailer adds information concerning the segment order and provides a cyclical redundancy check (CRC) for the segment. Each is 2 bytes in length.

The SAR header is divided into three fields:

ST

Segment Type. This 2-bit field is used to indicate one of four segment types: Beginning of Message (BOM), Continuation of Message (COM), End of Message (EOM), or Single-Segment Message (SSM).

SN

Sequence Number. This 4-bit field allows the stream of SAR SDUs to be numbered using modulo 16. It is used to provide a "loss of segment" check for each full PDU that is segmented.

MID

Multiplexing Identification. This 10-bit field is used to multiplex CPCS connections on a single ATM Layer connection, when applicable.

The SAR trailer contains two parts:

LI

Length Indication. This 6-bit field is binary- encoded to indicate the number of bytes of the CPCS-PDU (received from the Convergence Layer above) that are contained in the payload portion of this segment. For BOM and COM segments, this value MUST be 44. For EOM segments, the value can range from 4 to 44, as appropriate. For SSM segments, permitted values range from 8 to 44.

CRC

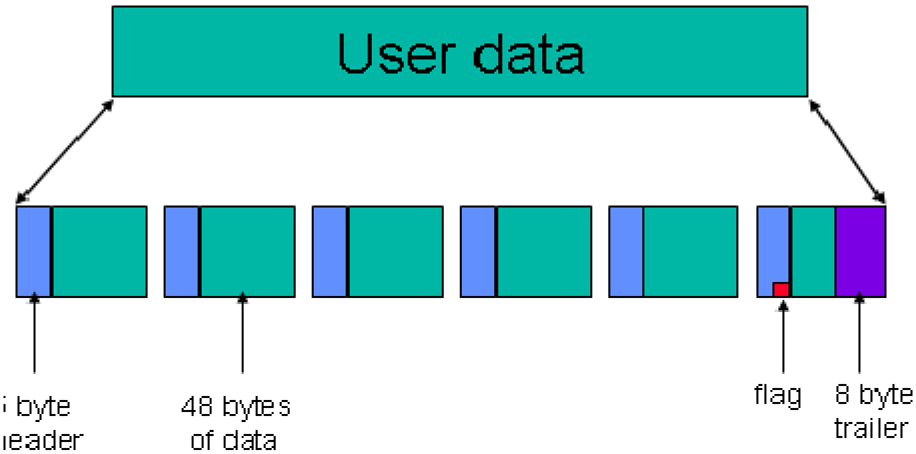
Cyclical Redundancy Check. This 10-bit sequence functions as an error check for the entire SAR-SDU, including the header, payload, and LI field of the trailer.

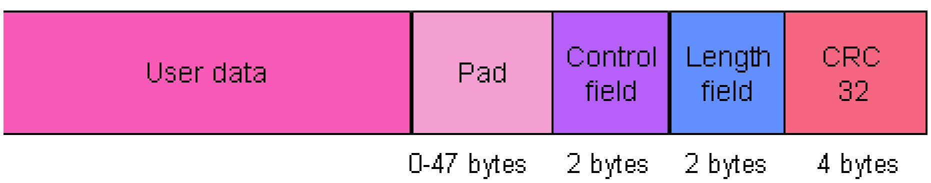
AAL TYPE 5

This section addresses ATM Adaptation Layer protocol type 5 (AAL 5), which is designed to accommodate the same variable bit rate, connection-oriented asynchronous traffic or connectionless packet data supported by AAL 3/4, but without the segment tracking and error correction requirements. It is used to transfer non-SMDS data, such as classical IP over ATM and LAN Emulation (LANE). It is also known as the Simple and Efficient Adaptation Layer (SEAL).

Like all of the AAL protocols, AAL 5 is designed to accept the data from the upper-layers (i.e., USER layers) of the end user and prepare it for handoff in 48-byte segments to the ATM Layer of the ATM protocol stack. The ATM Layer then provides each segment with an appropriate header and hands the 53-byte ATM cell down to the Physical Layer for transmission over the ATM network. For all cells except the last, the low order bit in the Payload Type Indicator (PTI) is set to zero to indicate the cell is not the last cell is a series that represents a single frame. For the last cell, the low order bit in the PTI field is set to one.

AAL 5 is geared for a streamlined transmission. It assumes that error recovery is performed by the higher layers, so that all 48 bytes of the payload may be allocated to carry data. It also assumes that only ONE message is transmitted over the UNI at one time. (If there is more than one user at a time, messages are queued for sequential transmission.)





AAL 5 DATA FLOW

The first step in the data flow for AAL 5 is the Service Data Unit (SDU) being passed down from the Upper (User) Layer to the Convergence Sublayer of the AAL Layer.

Like AAL 3/4, AAL 5 subdivides the Convergence Sublayer into the Service Specific Convergence Sublayer (SSCS) and the Common Part Convergence Sublayer (CPCS). The Upper Layer SDU is passed through the SSCS Sublayer unchanged.

The CPCS Sublayer adds an 8-byte trailer to the SDU and pads the SDU with 0-47 bytes of data in such a manner that the SDU may be divided evenly into 48-byte segments, with the trailer occupying the last 8 bytes of the last segment.

At this time, the CPCS-PDU is handed down to the SAR Sublayer, where it is divided into 48-byte segments. THE SAR SUBLAYER PERFORMS NO ADDITIONAL FUNCTIONS WITH AAL 5.

Each of the segments is then handed (unchanged) to the ATM Layer for incorporation into an appropriate ATM cell. The ATM Layer then sends the cell to the Physical Layer for transmission over the ATM network.

This process is outlined below:

```
+-----------------------------------------------+
|               Service Data Unit               |
+-----------------------------------------------+
|                                               |
V                                               V
+-----------------------------------------------+-------+------+
|  Convergence Sublayer Protocol Data Unit      |  PAD  |  CT  |
+-----------------------------------------------+-------+------+
|                                                              |
V                                                              V
+-------------------++-------------------++-------------------+
|                   ||                   ||     |       |     |
+-------------------++-------------------++-------------------+
|   First Segment   ||   Second Segment  ||   Last Segment    |
V                   VV                   VV                   V

CT = CPCS Trailer
-++--------------------++--------------------+
 ||                    ||     |       |      |
-++--------------------++--------------------+
 ||   Second Segment   ||     Last Segment   |
 VV                    VV                    V
                        +--------------------+
                        |     |       |      |
                        +--------------------+
                        |       SAR-SDU      |
                        V                    V
                  +-----+--------------------+
                  |  AH |     |       |      |
                  +-----+--------------------+
                        |___ ATM Payload ___|

                  |____ 53-Byte ATM Cell ___|

AH = ATM Header
```

AAL 5 CPCS-PDU FORMAT

```
+-----------------------------------------------+-------+------+
|  Convergence Sublayer Protocol Data Unit      |  PAD  |  CT  |
+-----------------------------------------------+-------+------+
                                                      /        |
                                                   /           |
                                              /                |
                                          /                    |
                                       +----+----+------+--------+
                                       |U-U |CPI |Length| CRC-32 |
                                       +----+----+------+--------+
```

The 8-byte CPCS trailer is divided into four fields:

CPCS-UU

CPCS User-to-User Indication. This 1-byte field is used to transparently transfer information from the origination user to the destination user.

CPI

Common Part Indicator. This is a 1-byte field used to align the CPCS-PDU trailer to the 32-bit boundary.

Length

This 2-byte field indicates the length of the CPCS payload, not including the PAD bytes.

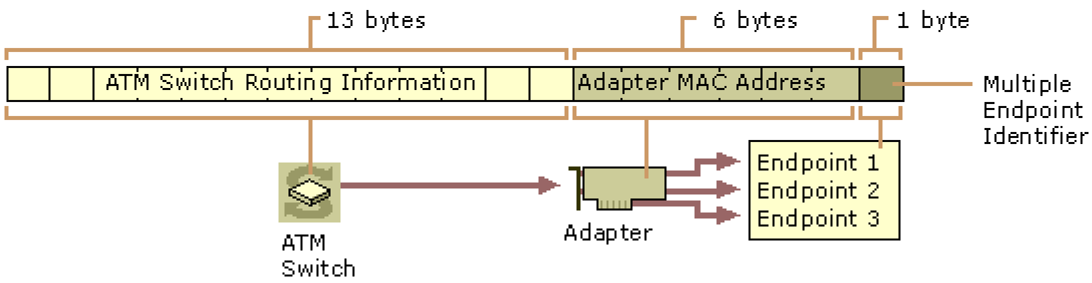
CRC-32

Cyclical Redundancy Check. The 4-byte CRC-32 field is a 32-bit error check for the entire contents of the CPCS-PDU, including the payload, PAD field, and first 4 bytes of the trailer. The CRC-32 generating polynomial is: $x^{32} + x^{26} + x^{23} + x^{22} + x^{16} + x^{12} + x^{11} + x^{10} + x^{8} + x^{7} + x^{5} + x^{4} + x^{2} + x + 1$.

<u>3.6 ATM Addresses</u>

ATM addresses are needed to support the use of virtual connections through an ATM network. At the simplest level, ATM addresses are 20 bytes in length and composed of three distinct parts. Figure shows the three parts of the 20-byte ATM address.

This ATM address breaks down into the following three basic parts:



ATM switch identifier The first 13 bytes identify a particular switch in the ATM network. The use of this portion of the address can vary considerably depending on which address format is in use. Each of the three major ATM addressing schemes in use provides information about ATM switch location differently. The three formats are the data country/region code (DCC) format, international code designator (ICD) format, and the E.164 format proposed by the ITU-T for international telephone numbering use in broadband ISDN networks.

Adapter MAC address The next 6 bytes identify a physical endpoint, such as a specific ATM adapter, using a media access control (MAC) layer address that is physically assigned to the ATM hardware by the manufacturer. The use and assignment of MAC addresses for ATM hardware is identical to MAC addressing for other Institute of Electrical and Electronic Engineers (IEEE) 802. *x* technologies, such as Ethernet and Token Ring.

Selector (SEL) The last byte is used to select a logical connection endpoint on the physical ATM adapter.

Although all ATM addresses fit this basic three-part structure, there are significant differences in the exact format of the first 13 bytes of any given address, depending on the addressing format that is being used or whether the ATM network is for public or private use.

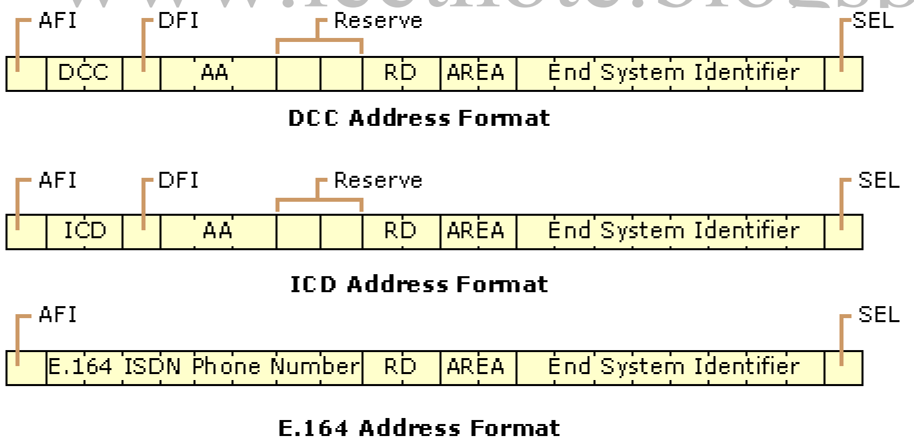
In summary, the 20-byte ATM address is in the hierarchical format starting with the switch at the highest level, down to the adapter, and then down to the logical endpoint.

All of the three ATM address formats that are currently in widespread use (DCC, ICD, and E.164) include the following characteristics:

- Compliance with the Network Service Access Point (NSAP) addressing plan as proposed by the Open Standards Interconnection (OSI) protocol suite of the International Standards Organization (ISO).
- Each can be used to establish and interconnect privately-built ATM networks that support switched virtual circuits (SVCs).

Addressing in Detail

The type of ATM address used depends on whether the addresses are for a public or private ATM network. ATM addresses are used to establish virtual circuit connections between ATM endpoints. Figure shows the three primary address formats.



The three formats are known as data country/region code (DCC), International Code Designator (ICD), and E.164 addresses. The E.164 address format is designed specifically for public ATM networks.

Addressing endpoints in an ATM network is part of the UNI signaling. While ATM switches route based on connections (identified by the VPI/VCI), once the cell arrives at the destination network, there has to be an address to get it to the proper node.

Endpoint addresses are carried in the payload of an ATM cell as part of the signaling message. There are three different address formats presently defined for use at the private UNI and an additional format for the public UNI:

- Data Country Code (DCC) (private)
- International Code Designator (ICD) (private)
- E.164 ATM private format (private)
- E.164 ATM public format (public)

Figure 9.6 shows the four address formats used in ATM. The Authority and Format Identifier (AFI) defines the authority responsible for address registration and the format used. The authority can be an ATM equipment manufacturer, service provider, telephone company, or administrator of a private network. The DCC is a 2-byte field that identifies the country in which the address is registered.

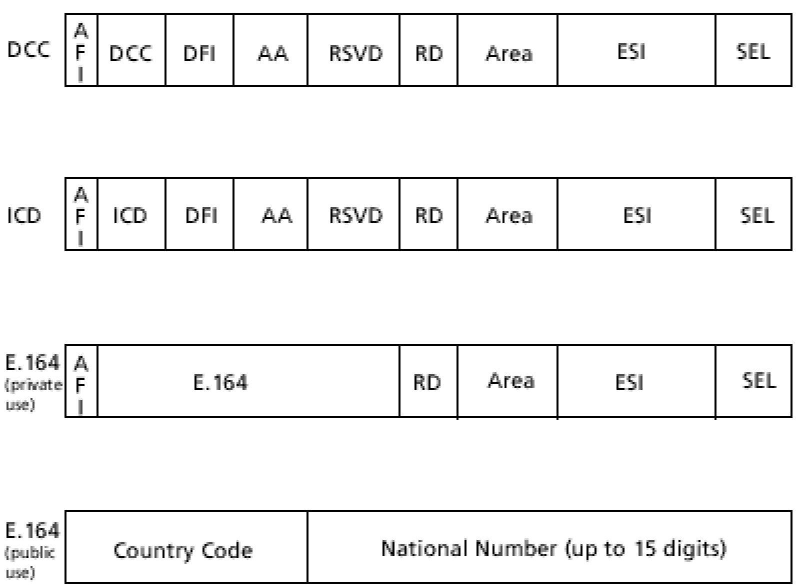


Figure 9.6
ATM Address Formats

The Domain Specific Identifier (ESI) is a 6-byte field that identifies an end system within an area. The selector field (SEL) is a 1-byte field used by the end system to select an endpoint within an end system. The ICD is a 2-byte field that identifies an international organization.

Codes are maintained by the British Standards Institute. The E.164 field is an 8-byte field that uses the same addressing as defined for ISDN; it is used to identify ISDN numbers.

When a cell is transmitted, the subscriber equipment provides the ESI and SEL values which identify the end system and endpoint originating the cell. The network then fills in the rest of the address information when the cell is passed to the network over the UNI. The addreses are registered with the network for future connections.

3.7 UNI Signaling

ATM UNI: ATM Signaling User-to-Network Interface

Signalling is the process by which ATM users and the network exchange the control of information, request the use of network resources, or negotiate for the use of circuit parameters. The VPI/VCI pair and requested bandwidth are allocated as a result of a successful signalling exchange. These messages are sent over the Signalling ATM Adaptation Layer (SAAL), which ensures their reliable delivery. The SAAL is divided into a Service Specific Part and a Common Part. The Service Specific Part is further divided into a Service Specific Coordination Function (SSCF), which interfaces with the SSCF user; and a Service Specific Connection-Oriented Protocol (SSCOP), which assures reliable delivery.

The UNI signalling protocols within the SAAL are responsible for ATM call and connection control, including call establishment, call clearing, status enquiry, and point-to-multipoint control. This signalling message uses the Q.931 message format, which is made up of a message header and a variable number of Information Elements (IEs).

The VPI/VCI pair and requested bandwidth are allocated as a result of a successful signalling exchange. Two levels of virtual connections can be supported at the UNI:

- A point-to-point or point-to-multipoint Virtual Channel Connection (VCC) which consists of a single connection established between two ATM VCC end-points.
- A point-to-point or point-to-multipoint Virtual Path Connection (VPC) which consist of a bundle of VCCs carried transparently between two ATM VPC end-points.

Note: For VPC at the Public UNI, traffic monitoring and throughput enforcement will be performed across all cells carried on the same VPI independently of the VCI values.

Protocol Structure - ATM Signaling UNI: Signaling User-to-Network Interface :AAL protocol stacks illustrated below support UNI connection control signaling:

	User-Network Signalling
SAAL	UNI SSCF
	SSCOP
	AAL Type 5 Common Part
	ATM Layer
	Physical Layer

UNI Signaling Message:

8	7	6	5	4	3	2	1	bit/Octet
Protocol discriminator								1
0	0	0	0	Length of call reference value				2
Flag	Call reference value							3
Call reference value (continued)								4
								5
Message type								6
Message type (continued)								7
Message length								8
Message length (continued)								9
Variable length Information Elements as required								etc.

- Protocol discriminator -Distinguishes Messages for user-network call control from other messages. (9 for Q.2931 messages)
- Call reference - Unique number for every ATM connection which serves to link all signalling messages relating to the same connection. It is comprised of the call reference value and the call reference flag. The call reference flag indicates who allocated the call reference value.

- Message type - The message may be of the following types:
    1. Call establishment messages: such as CALL PROCEEDING, sent by the called user to the network or by the network to the calling user to indicate initiation of the requested call. CONNECT, sent by the called user to the network and by the network to the calling user to indicate that the called user accepted the call. CONNECT ACKNOWLEDGE, sent by the network to the called user to indicate that the call was awarded and by the calling user to the network. And SETUP, sent by the calling user to the network and by the network to the calling user to initiate a call.
    2. Call clearing messages: such as RELEASE, sent by the user to request that the network clear the connection or sent by the network to indicate that the connection has cleared. RELEASE COMPLETE, sent by either the user or the network to indicate that the originator has released the call reference and virtual channel. RESTART, sent by the user or the network to restart the indicated virtual channel. RESTART ACKNOWLEDGE, sent to acknowledge the receipt of the RESTART message.
    3. Miscellaneous messages: such as STATUS, sent by the user or network in response to a STATUS ENQUIRY message. STATUS ENQUIRY, sent by the user or the network to solicit a STATUS message.
    4. Point-to-Multipoint messages: such as ADD PARTY, adds a party to an existing connection. ADD PARTY ACKNOWLEDGE, acknowledges a successful ADD PARTY. ADD PARTY REJECT, indicates an unsuccessful ADD PARTY. DROP PARTY, drops a party from an existing point-to-multipoint connection. DROP PARTY ACKNOWLEDGE, acknowledges a successful DROP PARTY.
- Message length - The length of the contents of a message.
- Information Elements - There are several types of information elements. Some may appear only once in the message; others may appear more than once. Depending on the message type, some information elements are mandatory and some are optional. The order of the information elements does not matter to the signalling protocol. The information elements in UNI 3.0 are listed in the following table:

IE	Description	Max. No.
Cause	Gives the reason for certain messages. For example, the Cause IE is part of the release message, indicating why the call was released.	2
Call state	Indicates the current state of the call.	1
Endpoint reference	Identifies individual endpoints in a point-to-multipoint call.	1
Endpoint state	Indicates the state of an endpoint in a point-to-multipoint call.	1
AAL parameters	Includes requested AAL type and other AAL parameters.	1
ATM user cell rate	Specifies traffic parameters.	1
Connection identifier	Identifies the ATM connection and gives the VPI and VCI values.	1
Quality of Service parameter	Indicates the required Quality of Service class for the connection.	1
Broadband high-layer information	Gives information about the high-layer protocols for compatibility purposes.	1
Broadband bearer capacity	Requests a service from the network (such as CBR or VBR link, point-to-point and point-to-multipoint link).	1
Broadband low-layer information	Checks compatibility with layer 2 and 3 protocols.	3
Broadband locking shift	Indicates a new active codeset.	-
Broadband non-locking shift	Indicates a temporary codeset shift.	-
Broadband sending complete	Indicates the competition of sending the called party number.	1
Broadband repeat indicator	Indicates how IEs which are repeated in the message should be handled.	1
Calling party number	Origin of the call.	1
Calling party subaddress	Subaddress of calling party.	1
Called party number	Destination of the call.	1
Called party subaddress	Subaddress of the called party.	1
Transit network	Identifies one requested transit network.	1

selection		
Restart indicator	Identifies which facilities should be restarted (e.g., one VC, all VCs).	1

3.8 PNNI Signaling

ATM PNNI: ATM Private Network-to-Network Interface

The ATM Private Network-Node Interface (PNNI), an ATM network-to-network signaling protocol, provides mechanisms to support scalable, QoS-based ATM routing and switch-to-switch switched virtual connection (SVC) interoperability.

The PNNI (Private Network-to-Network Interface), is a hierarchical, dynamic link-state routing protocol. It is designed to support large-scale ATM networks. The PNNI protocol uses VPI/VCI 0,18 for its messages. In addition, it uses signalling messages to support connection establishment across multiple networks. PNNI is based on UNI 4.0 and Q.2931. Specific information elements were added to UNI 4.0 in order to support the routing process of PNNI. PNNI Signalling contains the procedure to dynamically establish, maintain and clear ATM connections at the private network to network interface or network node interface between 2 ATM networks or 2 ATM network nodes. The PNNI signalling protocol is based on the ATM forum UNI specification and on Q.2931 .

PNNI Messages include:ALERTING, CALL PROCEEDING, CONNECT, SETUP, RELEASE, RELEASE COMPLETE, NOTIFY, STATUS, STATUS ENQUIRY, RESTART, RESTART ACKNOWLEDGE, STATUS, ADD PARTY, ADD PARTY ACKNOWLEDGE, PARTY ALERTING, ADD PARTY REJECT, DROP PARTY, DROP PARTY ACKNOWLED

Protocol Structure - ATM PNNI: ATM Private Network-to-Network Interface The structure of the PNNI header is shown in the following illustration:

2bytes	2bytes	1byte	1byte	1byte	1byte
Packet type	Packet length	Prot ver	Newest ver	Oldest ver	Reserved

- Packet type: The following packet types are defined:
    1. Hello - Sent by each node to identify neighbor nodes belonging to the same peer group.
    2. PTSP - PNNI Topology State Packet. Passes topology information between groups.
    3. PTSE - PNNI Topology State Element (Request and Ack). Conveys topology parameters such as active links, their available bandwidth, etc.
    4. Database Summary - Used during the original database exchange between two neighboring peers.
- Packet length- The length of the packet.
- Prot ver - Protocol Version. The version according to which this packet was formatted.
- Newest ver / Oldest ver - Newest version supported / oldest version supported. The newest version supported and the oldest version supported fields are included in order for nodes to negotiate the most recent protocol version that can be understood by both nodes exchanging a particular type of packet

Module 4

SATELLITE COMMUNICATION: Satellite communication principles - Geo stationary satellites - block schematic of satellite earth station - VSAT - VSAT networks - applications in personnel communication. (basic ideas only)

4.1 Satellite Communication Principles

A communications satellite (sometimes abbreviated to SATCOM) is an artificial satellite stationed in space for the purposes of telecommunications. Modern communications satellites use a variety of orbits including geostationary orbits, Molniya orbits, other elliptical orbits and low (polar and non-polar) Earth orbits.

For fixed (point-to-point) services, communications satellites provide a microwave radio relay technology complementary to that of submarine communication cables. They are also used for mobile applications such as communications to ships, vehicles, planes and hand-held terminals, and for TV and radio broadcasting, for which application of other technologies, such as cable, is impractical or impossible.

Mobile and Fixed Services is based on the premise that designers of future satellite systems must take account of the strong competition that satellites face from optical fibers. In future years, satellites will continue to be commercially viable media for telecommunications only if systems designers take account of the unique features that satellites have to offer. Accordingly, Satellite Communications places more emphasis on satellite mobile services and broadcasting, and less emphasis on fixed, point-to-point, high-capacity services than traditional textbooks in the field. Also, an emphasis is given in the book to design issues. Numerous illustative system design examples and numerical problems are provided. The particular attention given to methods of design of satellite mobile communications systems should make it an indispensable resource for workers in this field. The book also contains some recent results of propagation modelling and system design studies which should be of particular value to researchers and designers of satellite systems for mobile communications services.

Fixed Satellite Services: Ready for the Future?Fixed Satellite Services (FSS) have long been a steady and stable component of the telecommunications industry. Fixed Satellite Service operators brought in billions of dollars in revenues from carriers and service providers that leased satellite capacity to extend the reach of their services. However, the FSS operators have watched their market position erode due to the expansion of international fiber assets, leading to quickly falling satellite capacity prices and margins. Operators have responded by launching new, next-generation satellites to provide more managed and IP-based services to customers. To drive revenue growth, these companies have also expanded their potential customer base to include end users, particularly large enterprise customers with managed networking needs. FSS operators can be broken down into two categories, regional and global, with only a handful of companies that truly qualify as global satellite operators with significant capacity directed at every continent. This Report analyzes the global FSS operators along with their launch of new satellites and rollout of new services. Looking at revenues, as shown in Exhibit 1, it is easy to see that global FSS operators are large players in the international telecommunications market. GEA and SES have recently announced plans to merge, putting them at the same revenue level as Intelsat, PanAmSat, and Loral.

Advantages:

- Flexible (if transparent transponders)
- Easy to install new circuits
- Circuit costs independent of distance
- Broadcast possibilities
- Temporary applications (restoration)
- Niche applications
- Mobile applications (especially "fill-in")
- Terrestrial network "by-pass"
- Provision of service to remote or underdeveloped areas
- User has control over own network

- 1-for-N multipoint standby possibilities

Disadvantages

- Large up front capital costs (space segment and launch)
- Terrestrial break even distance expanding (now approx. size of Europe)
- Interference and propagation
- Congestion of frequencies and orbit

4.2Geostationary orbit

A satellite in a geostationary orbit appears to be in a fixed position to an earth-based observer. A geostationary satellite revolves around the earth at a constant speed once per day over the equator.

The geostationary orbit is useful for communications applications because ground based antennas, which must be directed toward the satellite, can operate effectively without the need for expensive equipment to track the satellite's motion. Especially for applications that require a large number of ground antennas (such as direct TV distribution), the savings in ground equipment can more than justify the extra cost and onboard complexity of lifting a satellite into the relatively high geostationary orbit.

A geostationary satellite is any satellite which is placed in a geostationary orbit. Satellites in geostationary orbit maintain a constant position relative to the surface of the earth.

Geostationary satellites do this by orbiting the earth approximately 22,300 miles above the equator. This orbital path is called the Clarke Belt, in honor of Arthur C. Clarke.

In other words, if a satellite in a geostationary orbit is in a certain place above the earth, it will stay in that same spot above the earth. Its latitude stays at zero and its longitude remains constant.

In contrast to geostationary orbits, Medium Earth Orbit and Low Earth Orbit satellites constantly change their positions in relation to the surface of the earth.

A single geostationary satellite will provide coverage over about 40 percent of the planet.

Geostationary satellites are commonly used for communications and weather-observation.

The typical service life expectancy of a geostationary satellite is ten to fifteen years.

Because geostationary satellites circle the earth at the equator, they are not able to provide coverage at the Northernmost and Southernmost latitudes.

Geostationary orbits are often referred to as geosynchronous or just GEO.

Medium Earth Orbit represents a series of tradeoffs between geostationary orbit (GEO) and Low Earth Orbit (LEO).

Medium Earth Orbit enables a satellite provider to cover the earth with fewer satellites than Low Earth Orbit, but requires more satellites to do so that geostationary orbit.

Medium Earth Orbit terrestrial terminals can be of lower power and use smaller antennas than the terrestrial terminals of geostationary orbit satellite systems. However, they cannot be as low power or have as small antennas as Low Earth Orbit terrestrial terminals.

Medium Earth Orbit satellite systems offer better Round Trip Time (RTT) than geosynchronous orbit sysMedium Earth Orbit (MEO) refers to a satellite which orbits the earth at an altitude below 22,300 miles (geostationary orbit) and above the altitude of Low Earth Orbit (LEO) satellites.tems, but not as low as Low Earth Orbit systems.

Low Earth Orbit (LEO) refers to a satellite which orbits the earth at altitudes between (very roughly) 200 miles and 930 miles.

Low Earth Orbit satellites must travel very quickly to resist the pull of gravity -- approximately 17,000 miles per hour. Because of this, Lowe Earth Orbit satellies can orbit the planet in as little as 90 minutes.

Low Earth Orbit satellite systems require several dozen satellites to provide coverage of the entire planet.

Low Earth Orbit satellites typically operate in polar orbits.

Low Earth Orbit satellites are used for applications where a short Round Trip Time (RTT) is very important, such as Mobile Satellite Services (MSS).

Low Earth Orbit satellites have a typical service life expectancy of five to seven years.

Perigee is the point at which a satellite in an elliptical orbit is closest to the Earth.

At its perigee, the satellite travels faster than at any other point in its orbit.

The opposite of perigee is apogee.

In a satellite network, Round Trip Time (RTT) is the time required for a signal to travel from a terrestrial system up to the satellite and back, or for a signal to travel from a satellite down to a terrestrial system and back up to the satellite again.
Round Trip Time is limited by the speed of light. Round Trip Time will be shortest for a Low Earth Orbit (LEO) satellite and longest for a geostationary satellite.

4.3Block Schematic of satellite Earth Station

Satellite Transponder Communications Link

A transponder is a broadband RF channel used to amplify one or more carriers on the downlink side of a geostationary communications satellite. It is part of the microwave repeater and antenna system that is housed onboard the operating satellite. Examples of these satellites include AMC 4 and Telstar 5, located at 101 and 97 degrees west longitude, respectively. These satellites and most of their cohorts in the geostationary orbit have bent-pipe repeaters using C and Ku bands; a bent pipe repeater is simply one that receives all signals in the uplink beam, block translates them to the downlink band, and separates them into individual transponders of a fixed bandwidth. Figure 1 shows the basic concept. Each transponder is amplified by either a traveling wave tube amplifier (TWTA) or a solid state power amplifier (SSPA). Satellites of this type are very popular for transmitting TV channels to broadcast stations, cable TV systems, and directly to the home. Other applications include very small aperture terminal (VSAT) data communications networks, international high bit rate pipes, and rural telephony. Integration of these information types is becoming popular as satellite transponders can deliver data rates in the range of 50 to 150 Mbps. Achieving these high data rates requires careful consideration of the design and performance of the repeater.

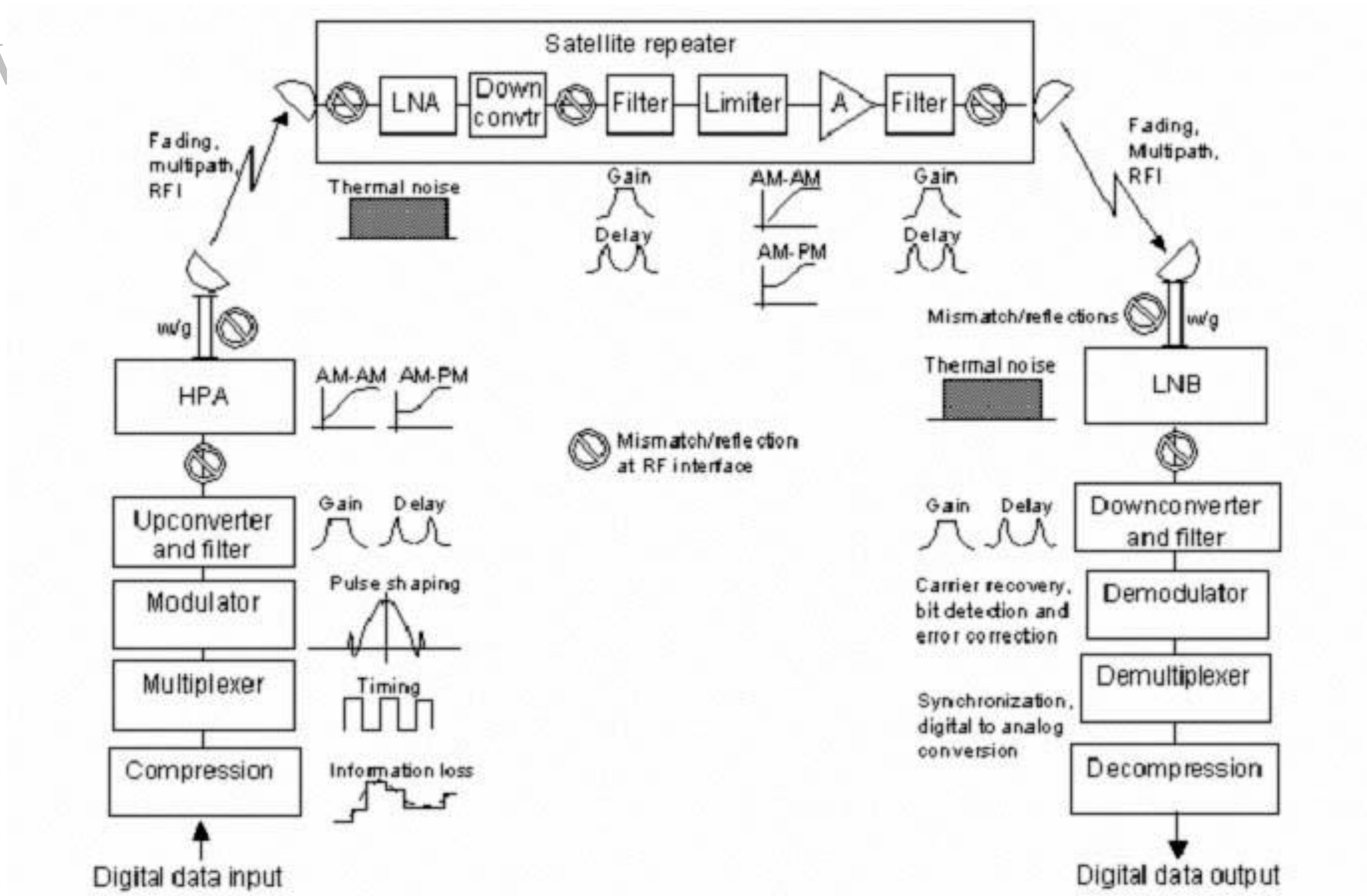


Figure 1. Elements of a basic Satellite link showing ground stations and the satellite transponder. The nature and location of the various system impairments are also shown.

The most significant impairments to digital transmission come about in the filtering, which constrains bandwidth and introduces delay distortion, and the power amplification, which produces AM/AM and AM/PM conversion. These effects will be discussed in detail later in this article. For maximum power output with the highest efficiency (e.g., to minimize solar panel DC supply), this amplifier should be operated at its saturation point. However, many services are sensitive and susceptible to AM/AM and AM/PM conversion, for which backoff is necessary. With such an operating point, intermodulation distortion can be held to an acceptable level; however, backoff also reduces downlink power.

The transponder itself is simply a repeater. It takes in the signal from the uplink at a frequency f_1, amplifies it and sends it back on a second frequency f_2. Figure 2 shows a typical frequency plan with 24-channel transponder. The uplink frequency is at 6 GHz, and the downlink frequency is at 4 GHz. The 24 channels are separated by 40 MHz and have a 36 MHz useful bandwidth. The guard band of 4 MHz assures that the transponders do not interact with each other.

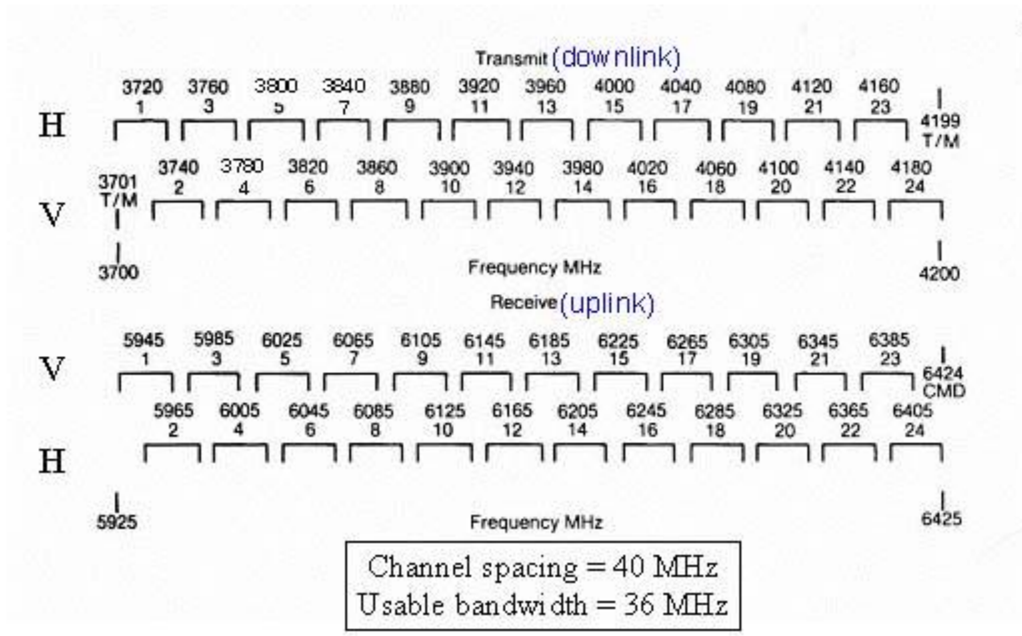


Figure 2. Basic 24 channel C-band transponder frequency plan

A transponder (also TPDR, TR, XPNDR, XPDR) is an electronic device used in wireless communications, the word itself is shorthand for transmitter-responder.

This device is primarily used as a re-transmitter due to the fact that it receives a particular signal from a particular source, then it amplifies (strengthens) the signal before sending it to a predefined location. Transponders have an abnormally large number of applications in our daily lives. Some of the most common uses are: satellite television, satellite telephony, air traffic control and in automobiles. They are also embedded in cars to open gates automatically. We shall look at some of these applications later. First of all it is important to mention that transponders are of two general varieties which are active transponders and passive transponders.

Active transponder: These devices as the name implies, continually emit radio signals which are tracked and monitored. These can also be automatic devices which strengthen the received signals and relay them to another location.

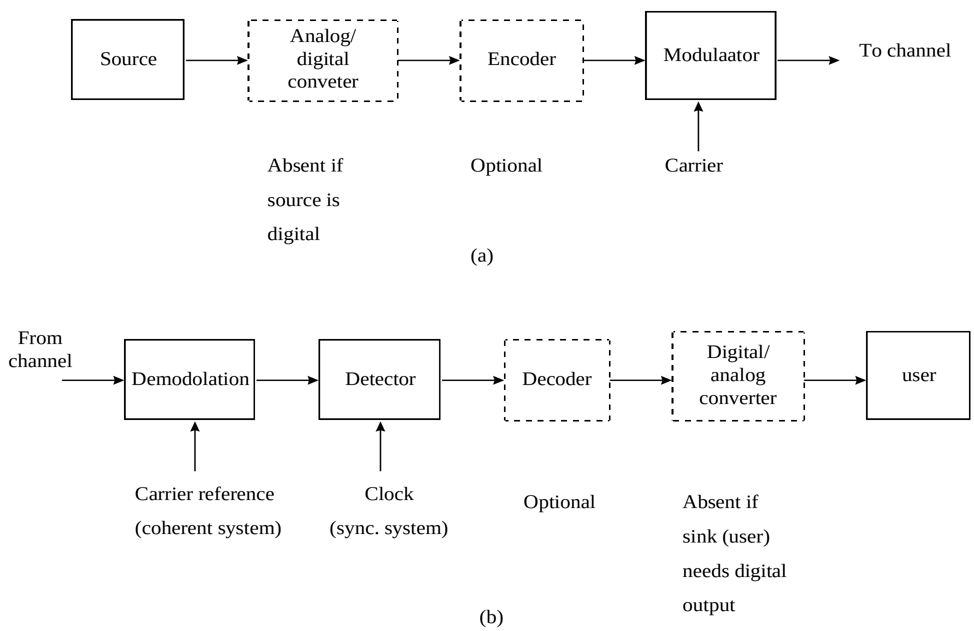
These devices are so frequently used that we often fail to recognize them. For example, how do you think lap times of NASCAR and formula one cars are monitored so accurately? Well the answer lies in the transponders which cars have embedded in them. Each car has a unique ID code which is transmitted as the car moves. A special cable loop is dug into the ground at the start-finish lines. So when the cars zoom by the finish line, their IDs are recorded along with their lap times. These times are automatically displayed on the position board along with split times, laps remaining and so on.

Another important use of active transponders is in satellite communications. Normally there are hundreds of thousands of tiny transponders embedded in one satellite. These receive an incoming signal over a range of frequencies (band), measured in hertz and megahertz and retransmit these signals on a different band simultaneously. The incoming signal originating from a point on the earth (e.g. A broadcaster), is called the uplink and the outgoing signal back to the earth is called the downlink. The logic behind using satellites for this purpose is simple - as radio signals cannot curve along the curvature of the earth, they are sent in a straight line up and received down in a straight line. This reduces time of signal delivery and increases range.

Now we come to the passive transponder which although not as active as their counterparts still play a very important role. These transponders contain information which is used to identify particular objects. For example passive transponders are sometimes embedded in our credit cards and on magnetic labels in large stores. These are paired with active transponders which amplify and transcribe the information.

Uplink is the signal path from an earth station to a satellite.

The opposite of uplink is downlink. Downlink is the signal path from the satellite toward the



earth.

Block diagram of a general Communication System.

Main Body : Transmitter + Channel + Receiver

Input transducer : Input information → V or I → Signal (Analog or digital)

Transmitter : To couple the message to the channel via modulation, filtering & amplifier

Purposes :

(1) Ease of radiation

(2) Reduce the noise and interference

(3) For channel assignment

(4) For multiplication signals

(5) Overcome equipment limitations.

Channel : Channel effect

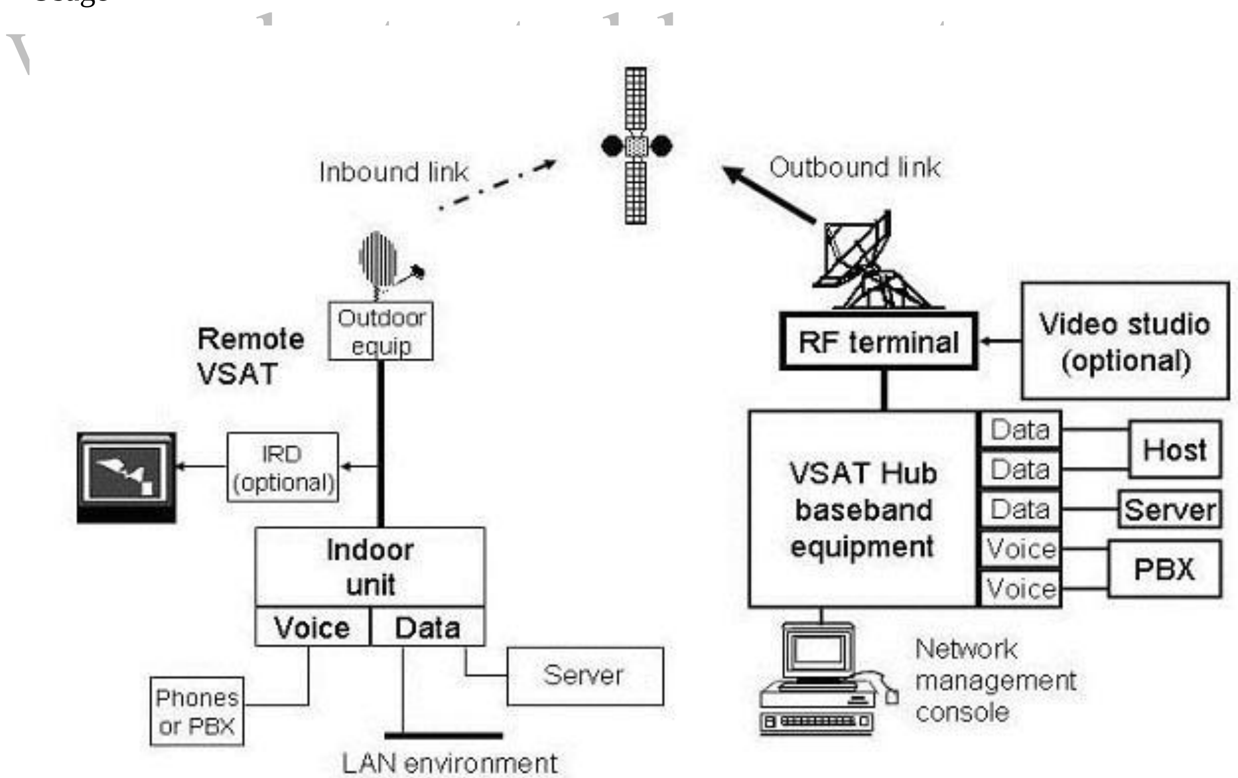
Receiver : Extract desired signal, convert signal for input transducer (demodulator)

Output transducer

4.4Very small aperture terminal

A Very Small Aperture Terminal (VSAT), is a two-way satellite ground station with a dish antenna that is smaller than 3 meters. Most VSAT antennas range from 75 cm to 1.2 m. Data rates typically range from 56 Kbit/s up to 4 Mbit/s. VSATs access satellites in geosynchronous orbit to relay data from small remote earth stations (terminals) to other terminals (in mesh configurations) or master earth station "hubs" (in star configurations). VSATs are most commonly used to transmit narrowband data (point of sale transactions such as credit card, polling or RFID data; or SCADA), or broadband data (for the provision of Satellite Internet access to remote locations, VoIP or video). VSATs are also used for transportable, on-the-move (utilising phased array antennas) or mobile maritime communications.

Usage



The first commercial VSATs were C band receive-only systems by Equatorial Communications using spread spectrum technology. More than 30,000 60 cm antenna systems were sold in the early 1980s. Equatorial later developed a C band (4/6 GHz) 2 way system using 1 m x 0.5 m antennas and sold about 10,000 units in 1984-85.

In 1985, Schlumberger Oilfield Research co-developed the world's first Ku band (12-14 GHz) VSATs with Hughes Aerospace to provide portable network connectivity for oil field drilling and exploration units. Ku Band VSATs make up the vast majority of sites in use today for data or telephony applications.

The largest VSAT network (more than 12,000 sites) was deployed by Spacenet and MCI for the US Postal Service. Other large VSAT network users include Walgreens Pharmacy, Dollar General, Wal-Mart, CVS, Riteaid, Yum! Brands (Taco Bell, Pizza Hut, Long John Silver's and other Quick Service Restaurant chains),Intralot, GTECH and SGI for lottery terminals. VSATs are used by car dealerships affiliated with manufacturers such as Ford and General Motors for transmitting and receiving sales figures and orders, as well as for receiving internal communications, service bulletins, and interactive distance learning courses from manufacturers. The FordStar network, used by Ford and its local dealers, is an example of this.

VSAT technology is also used for two-way satellite Internet providers such as HughesNet, StarBand and WildBlue in the United States; and ASTRA2Connect across Europe. These services are used across the world as a means of delivering broadband Internet access to locations which cannot get less expensive broadband connections such as ADSL or cable internet access; usually remote or rural locations.

Configurations

Most VSAT networks are configured in one of these topologies:

- A star topology, using a central uplink site, such as a network operations center (NOC), to transport data back and forth to each VSAT terminal via satellite,

- A mesh topology, where each VSAT terminal relays data via satellite to another terminal by acting as a hub, minimizing the need for a centralized uplink site,
- A combination of both star and mesh topologies. Some VSAT networks are configured by having several centralized uplink sites (and VSAT terminals stemming from it) connected in a multi-star topology with each star (and each terminal in each star) connected to each other in a mesh topology. Others configured in only a single star topology sometimes will have each terminal connected to each other as well, resulting in each terminal acting as a central hub. These configurations are utilized to minimize the overall cost of the network, and to alleviate the amount of data that has to be relayed through a central uplink site (or sites) of a star or multi-star network.
- SAT was originally intended for sporadic store-and-forward data communications but has evolved into real-time internet services. VSAT uses existing satellite broadcasting technology with higher powered components and antennas manufactured with higher precision than conventional satellite television systems. The satellite antenna at the customer's location includes, in addition to the receiver, a relatively high-powered transmitter that sends a signal back to the originating satellite. A very small portion of a transponder is used for each VSAT return path channel. Each VSAT terminal is assigned a frequency for the return path which it shares with other VSAT terminals using a shared transmission scheme such as time division multiple access.[1]

- An innovative feature of VSAT is that the technology has evolved to the point that something that previously could only be done with large, high-powered transmitting satellite dishes can now be done with a much smaller and vastly lower-powered antenna at the customer's premises. In addition, several return-path channels can co-exist on a single satellite transponder, and each of these return-path channels is further subdivided to serve multiple customers.
- In the system used by WildBlue, 31 different spot beams are used to serve the continental United States instead of the one beam used by conventional satellites.[2] Thus, the same Ka-band transponders and frequencies are used for different regions throughout the United States, effectively re-using the same bandwidth in different regions.
- The return path is transmitted from the customer's receiver in the L-band to a device called a low-noise block upconverter. There it is converted into the much higher frequency

satellite transmission frequency, such as Ku-band and Ka-band, and amplified. Finally the signal is emitted to the dish antenna which focuses the signal into a beam that approximately covers the satellite with its beam. Because the transmission cannot be precise in these smaller dishes there is some effort to use frequencies for the uplink that are not used by adjacent satellites otherwise interference can occur to those other satellites.

- Another satellite communications innovation, also used by satellite trucks for video transmission, is that only a small portion of a single satellite transponder is used by each VSAT channel. Previously a single transponder was required for a single customer but now several customers can use one transponder for the return path. This is in addition to time-based subdivision.
- dvances in technology have dramatically improved the price/performance equation of FSS (Fixed Service Satellite) over the past five years. New VSAT systems are coming online using Ka band technology that promise higher bandwidth rates for lower costs.
- FSS satellite systems currently in orbit have a huge capacity with a relatively low price structure. FSS satellite systems provide various applications for subscribers, including: telephony, fax, television, high speed data communication services, Internet access, Satellite News Gathering (SNG), Digital Audio Broadcasting (DAB) and others. These efficient communication systems, both for residential and business users.

VSAT is an abbreviation for a Very Small Aperture Terminal. It is basically a two-way satellite ground station with a less than 3 meters tall (most of them are about 0.75 m to 1.2 m tall) dish antenna stationed. The transmission rates of VSATs are usually from very low and up to 4 Mbit/s. These VSATs' primary job is accessing the satellites in the geosynchronous orbit and relaying data from terminals in earth to other terminals and hubs. They will often transmit narrowband data, such as the transactions of credit cards, polling, RFID (radio frequency identification) data, and SCADA (Supervisory Control and Data Acquisition), or broadband data, such as satellite Internet, VoIP, and videos. However, the VSAT technology is also used for various types of communications.

Equatorial Communications first used the spread spectrum technology to commercialize the VSATs, which were at the time C band (6 GHz) receive only systems. This commercialization led to over 30,000 sales of the 60 cm antenna systems in the early

1980s. Equatorial Communications sold about 10,000 more units from 1984 to 1985 by developing a C band (4 and 6 GHz) two way system with 1 m x 0.5 m dimensions.

In 1985, the current world's most used VSATs, the Ku band (12 to 14 GHz) was co-developed by Schlumberger Oilfield Research and Hughes Aerospace. It is primarily used to provide portable network connection for exploration units, particularly doing oil field drilling.

Implementations of VSAT

Currently, the largest VSAT network consists of over 12,000 sites and is administered by Spacenet and MCI for the US Postal Service (USPS). Walgreens Pharmacy, Dollar General, CVS, Riteaid, Wal-Mart, Yum! Brands (such as Taco Bell, Pizza Hut, Long John Silver's, and other fast food chains), GTEC, SGI, and Intralot also utilizes large VSAT networks. Many huge car corporations such as Ford and General Motors also utilizes the VSAT technology, such as transmitting and receiving sales figures and orders, along with announcing international communications, service bulletins, and for distance learning courses. An example of this is the "FordStar Network."

Two way satellite Internet providers also use the VSAT technology. Companies like StarBand, WildBlue, and HughesNet in the United States and SatLynx, Bluestream, and Technologie Satelitarne in Europe, and many other broadband services around the world in rural areas where high speed Internet connections cannot be provided use it too. A statistic from December 2004 showed that over a million VSATs were in place.

VSAT Configurations

Most of the current VSAT networks use a topology:

- Star topology: This topology uses a central uplink site (eg. Network operations center (NOC)), which transports the data to and from each of the VSAT terminals using satellites
- Mesh topology: In this configuration, each VSAT terminal will relay data over to another terminal through the satellite, acting as a hub, which also minimizes the need for an uplink site
- Star + Mesh topology: This combination can be achieved (as some VSAT networks do) by having multiple centralized uplink sites connected together in a multi-star topology which is in a bigger mesh topology. This topology does not cost so much in maintaining the network while also lessening the amount of data that needs to be relayed through one or more central uplink sites in the network.

VSAT's Strengths

VSAT technology has many advantages, which is the reason why it is used so widely today. One is availability. The service can basically be deployed anywhere around the world. Also, the VSAT is diverse in that it offers a completely independent wireless link from the local infrastructure, which is a good backup for potential disasters. Its deployability is also quite amazing as the VSAT services can be setup in a matter of minutes. The strength and the speed of the VSAT connection being homogenous anywhere within the boundaries is also a big plus. Not to forget, the connection is quite secure as they are private layer-2 networks over the air. The pricing is also affordable, as the networks themselves do not have to pay a lot, as the broadcast download scheme (eg. DVB-S) allows them to serve the same content to thousands of locations at once without any additional costs. Last but not least, most of the VSAT systems today use onboard acceleration of protocols (eg. TCP, HTTP), which allows them to delivery high quality connections regardless of the latency.

VSAT Drawbacks

As with everything, VSAT also has its downsides. Firstly, because the VSAT technology utilizes the satellites in geosynchronous orbit, it takes a minimum latency of about 500 milliseconds every trip around. Therefore, it is not the ideal technology to use with protocols that require a constant back and forth transmission, such as online games. Also, surprisingly, the environment can play a role in slowing down the VSATs. Although not as bad as one way TV systems like DirecTV and DISH Network, the VSAT still can have a dim signal, as it still relies on the antenna size, the transmitter's power, and the frequency band. Last but not least, although not that big of a concern, installation can be a problem as VSAT services require an outdoor antenna that has a clear view of the sky. An awkward roof, such as with skyscraper designs, can become problematic.

Typical customer applications of VSATs include:

Supermarket shops (tills, ATM machines, stock sale updates and stock ordering).
Chemist shops - Shoppers Drug Mart - Pharmaprix.
Broadband direct to the home. e.g. Downloading MP3 audio to audio players.

Broadband direct small business, office etc, sharing local use with many PCs.

Internet access from on board ship Cruise ships with internet cafes, commercial shipping communications.

Garages / vehicle sales / petrol stations / motor spares (tills, ATM machines, stock sale updates and stock ordering).

Hotel chains, hotel internet cafes.

Insurance offices, quotations access to head office computers, VPN.

Car rental offices, ATM machines.

Airlines, travel agents, booking systems.

Airport air traffic control, flight data.

Financial institutions - Banks, ATM machines.

Lottery terminals.

Manufacturers - sales offices, service divisions, plants.

Job centres.

Customs and tax offices / border passport control checkpoints.

Internet Service Providers. POP, VoIP, Cafe.

Phone booths, VoIP, SCPC.

Data file and software distributors.

Pipeline monitoring, well heads, oil rigs.

Rural telephony, data, videophone.

Schools.

Military, data transfer, voice, temporary fixed and mobile VSAT.

Environmental monitoring, weather stations, seismic monitoring.

Build, restore and grow a data network in and around Europe.

Mobile phone base station in remote locations or on board ship.

Module 5

Wireless Lan – Infrared Vs Radio transmission – Infrastructure & ad hoc n/w – IEEE 802.11 – Hiper Lan – Bluetooth – Physical Layer – MAC layer – Networking – Security

5.1WIRELESS LAN

wireless LAN or WLAN is a wireless local area network, which is the linking of two or more computers without using wires. WLAN utilizes spread-spectrum or OFDM modulation technology based on radio waves to enable communication between devices in a limited area, also known as the basic service set. This gives users the mobility to move around within a broad coverage area and still be connected to the network.

For the home user, wireless has become popular due to ease of installation, and location freedom with the gaining popularity of laptops. Public businesses such as coffee shops or malls have begun to offer wireless access to their customers; some are even provided as a free service. Large wireless network projects are being put up in many major cities..

Benefits

The popularity of wireless LANs is a testament primarily to their convenience, cost efficiency, and ease of integration with other networks and network components. The majority of computers sold to consumers today come pre-equipped with all necessary wireless LAN technology.

The benefits of wireless LANs include:

- Convenience: The wireless nature of such networks allows users to access network resources from nearly any convenient location within their primary networking environment (home or office). With the increasing saturation of laptop-style computers, this is particularly relevant.

- Mobility: With the emergence of public wireless networks, users can access the internet even outside their normal work environment. Most chain coffee shops, for example, offer their customers a wireless connection to the internet at little or no cost.

- Productivity: Users connected to a wireless network can maintain a nearly constant affiliation with their desired network as they move from place to place. For a business, this implies that an employee can potentially be more productive as his or her work can be accomplished from any convenient location.

- Deployment: Initial setup of an infrastructure-based wireless network requires little more than a single access point. Wired networks, on the other hand, have the additional cost and complexity of actual physical cables being run to numerous locations (which can even be impossible for hard-to-reach locations within a building).

- Expandability: Wireless networks can serve a suddenly-increased number of clients with the existing equipment. In a wired network, additional clients would require additional wiring.

- Cost: Wireless networking hardware is at worst a modest increase from wired counterparts. This potentially increased cost is almost always more than outweighed by the savings in cost and labor associated to running physical cables.
- Flexibility: With in radio coverage ,nodes can communicate without further restriction.Radio waves can penetrate walls,senders and receivers can be placed anywhere .
- Design. Wireless network allow for the design of small ,independent devices which can for example be put into a packet.
- Robustness: Wireless networks can survive disasters Eg.earthquakes or user pulling a plug

Disadvantages

Wireless LAN technology, while replete with the conveniences and advantages described above, has its share of downfalls. For a given networking situation, wireless LANs may not be desirable for a number of reasons. Most of these have to do with the inherent limitations of the technology.

- Security: Wireless LAN transceivers are designed to serve computers throughout a structure with uninterrupted service using radio frequencies. Because of space and cost, the antennas typically present on wireless networking cards in the end computers are generally relatively poor. In order to properly receive signals using such limited antennas throughout even a modest area, the wireless LAN transceiver utilizes a fairly considerable amount of power. What this means is that not only can the wireless packets be intercepted by a nearby adversary's poorly-equipped computer, but more importantly, a user willing to spend a small amount of money on a good quality antenna can pick up packets at a remarkable distance; perhaps hundreds of times the radius as the typical user. In fact, there are even computer users dedicated to locating and sometimes even cracking into wireless networks, known as wardrivers. On a wired network, any adversary would first have to overcome the physical limitation of tapping into the actual wires, but this is not an issue with wireless packets. To combat this consideration, wireless networks users usually choose to utilize various encryption technologies available such as Wi-Fi Protected Access (WPA). Some of the older encryption methods, such as WEP are known to have weaknesses that a dedicated adversary can compromise. (See main article: Wireless security.)

- Range: The typical range of a common 802.11g network with standard equipment is on the order of tens of meters. While sufficient for a typical home, it will be insufficient in a larger structure. To obtain additional range, repeaters or additional access points will have to be purchased. Costs for these items can add up quickly. Other technologies are in the development phase, however, which feature increased range, hoping to render this disadvantage irrelevant. (See WiMAX)

- Reliability: Like any radio frequency transmission, wireless networking signals are subject to a wide variety of interference, as well as complex propagation effects (such as multipath, or especially in this case Rician fading) that are beyond the control of the network administrator. In the case of typical networks, modulation is achieved by complicated forms of phase-shift keying (PSK) or quadrature amplitude modulation (QAM), making interference and propagation effects all the more disturbing. As a result, important network resources such as servers are rarely connected wirelessly.

- Speed /quality of service: The speed on most wireless networks (typically 1-108 Mbit/s) is reasonably slow compared to the slowest common wired networks (100 Mbit/s up to several Gbit/s)..

5.2Comparison: infrared vs. radio transmission

Infrared

uses IR diodes, diffuse light,multiple reflections (walls,furniture etc.)

Advantages

simple, cheap, available in many mobile devices

no licenses needed

simple shielding possible

Disadvantages

interference by sunlight, heat sources etc.

many things shield or absorb IR light

low bandwidth.

Example

IrDA (Infrared Data Association)

interface available everywhere

Radio

typically using the license free

ISM band at 2.4 GHz

Advantages

experience from wireless WAN and mobile phones can be used

coverage of larger areas

possible (radio can penetrate walls, furniture etc.)

Disadvantages

very limited license free frequency bands

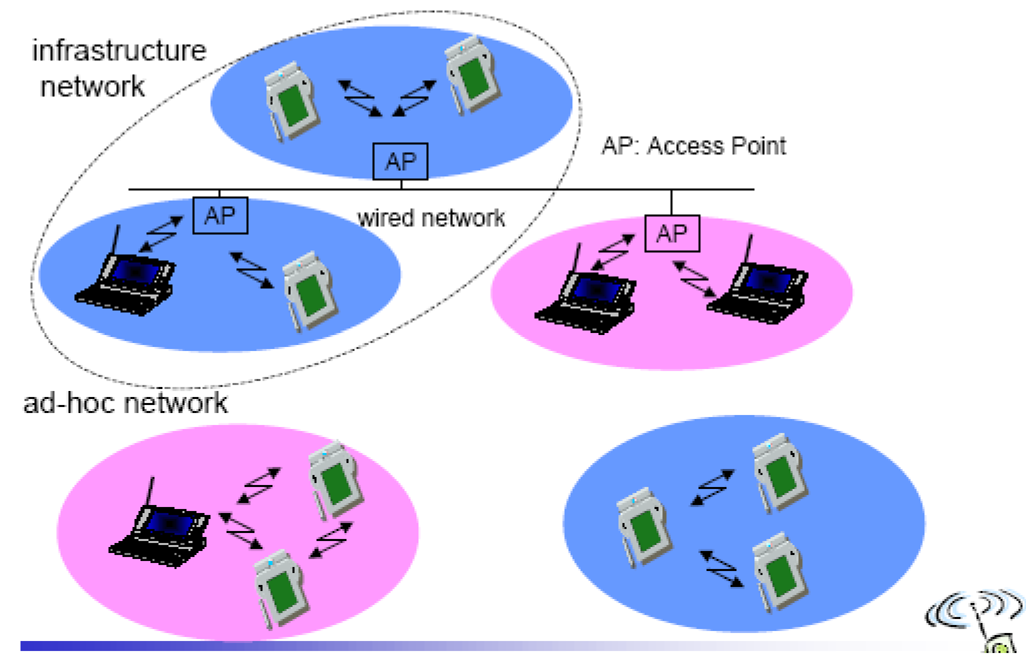
shielding more difficult, interference with other electrical devices

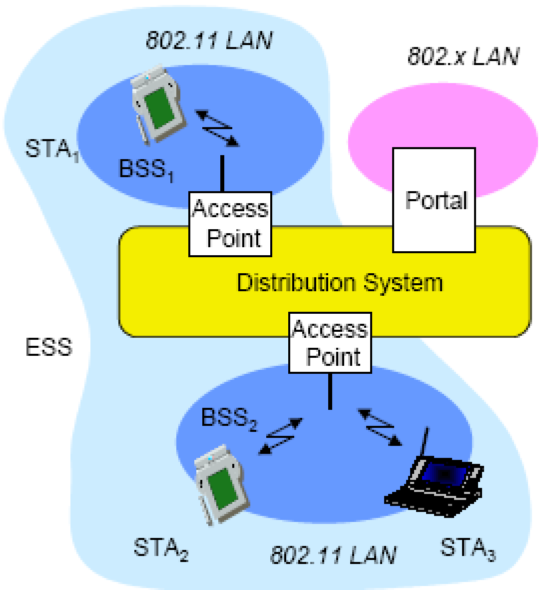
Example

WaveLAN, HIPERLAN,

Bluetooth

5.3 Infrastructure and Ad hoc Networks





Station (STA)

terminal with access mechanisms to the wireless medium and radio contact to the access point

Basic Service Set (BSS)

group of stations using the same radio frequency

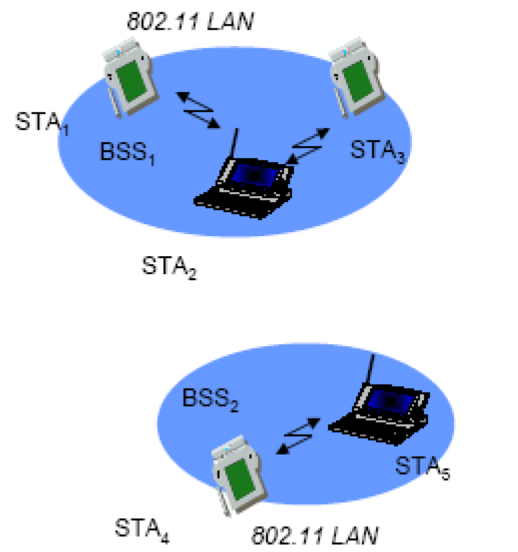
Access Point

station integrated into the wireless LAN and the distribution system

Portal

bridge to other (wired) networks Distribution System

interconnection network to form one logical network (EES:Extended Service Set) based on several BSS

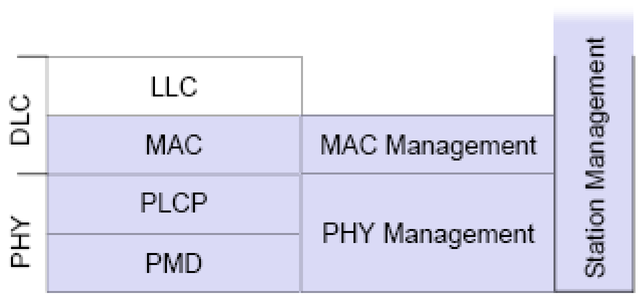


Direct communication within a limited range

Station (STA):terminal with access mechanisms to the wireless medium

Basic Service Set (BSS):group of stations using the same radio frequency

5.4IEEE 802.11



The IEEE 802.11 covers the physical layer PHY and medium access layer MAC like the other 802.X LANs do.

The **physical layer** is subdivided into the **physical layer convergence protocol (PLCP**) and the **physical medium dependent sub layer PMD.**

The basic tasks of the **MAC layer** are

- medium access
- fragmentation of user data
- Encryption.

The **PLCP sub layer** provides

- A carrier sense signal, called clear channel assessment (CCA).
- A common PHY service access point (SAP).

The **PMD sub layer** handles

- Modulation
- Encoding / decoding of signals.

The **MAC management**

- Supports the association and re-association of a station to an access point and roaming between different access points.
- It also controls authentication mechanism and encryption.
- synchronization of a station with regard to an access point and
- Power management to save battery power.
- Maintains the MAC management information base (MIB).

The main tasks of **PHY management** include

- Channel tuning
- PHY MIB maintenance.

The **Station Management** interacts with both management layers and is responsible for additional higher layer functions.

Physical layer

IEEE 802.11 supports three different physical layers: one layer based on infra red and two layer based on radio transmission. All PHY variants include the provision of the clear channel assessment signal (CCA).This is needed for the MAC mechanisms controlling medium access and indicates if the medium is currently idle.

Medium Access control layer

It has to complete several tasks

- Control medium Access
- Offer support for roaming

- Authentication
- Power conservation.

The basic services provided by the MAC layer are the

- mandatory asynchronous data service
- An optional time-bounded service.

Asynchronous service supports broadcast and multi-cast packets and packet exchange is based on a 'best effort' model,

The following three basic access mechanisms have been defined for IEEE 802.11::

- The mandatory basic method based on a version of CSMA/CA
- An optional method avoiding the hidden terminal problem.
- Contention-free polling method for time-bounded service.

The first two methods are summarized as distributed coordination function(DCF).The third method is called point coordination function(PCF).

MAC mechanisms are also called **distributed foundation wireless medium access control (DFWMAC).**

For all access methods, several parameters for controlling the waiting time before medium access are important.

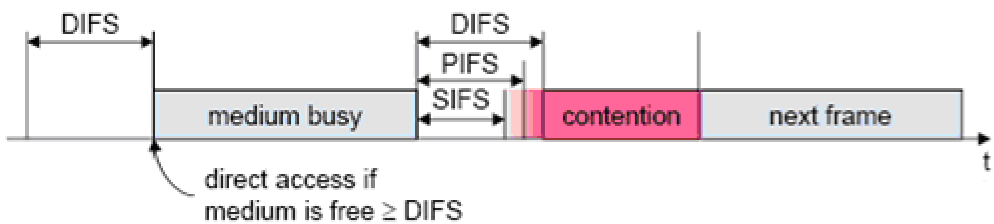


Fig : medium access and inter-frame spacing

Fig shows three different parameters that define the priorities of medium access. The values of the parameters depend on the PHY and are defined in relation to a **slot time**. Slot time is derived from the medium propagation delay, transmitter delay, and other PHY dependent parameters.

The medium can be busy or idle. If the medium is busy this can be due to data frames or other control frames, During contention phase several nodes try to access the medium.

- Short inter-frame spacing (SIFS):The shortest waiting time for medium access is defined for short control messages, such as acknowledgements of data packets or polling responses. For DSSS SIFS is 10μS and for FHSS is 28 μS.
- PCF inter-frame spacing (PIFS):A waiting time between DIFS and SIFS is used for a time – bounded service .An access point polling other nodes only has to wait PIFS for medium access.PIFS is defined as SIFS plus one slot time.
- DCF inter-frame spacing (DIFS): This parameter denotes the longest waiting time and has the lowest priority for medium access. This waiting time is used for asynchronous data service within a contention period.DIFS is defined as SIFS plus two slot times.

Basic DFWMAC-DCF using CSMA/CA.

This is a random access scheme with carrier sense and collision avoidance through random backoff.

If the medium is busy, nodes have to wait for the duration of DIFS,entering a contention phase afterwards. Each node now chooses a random backoff time within a contention window and delays medium access for this random amount of time. The continues to sense the medium As soon as a node senses the channel is busy , it has lost this cycle and has to wait for the next, chance i. e., until the medium is idle again for at least DIFS. But if the randomized additional waiting time for a node is over and the medium is still idle, the node can access the medium immediately. The additional waiting time is measured in multiples of the waiting time. The additional randomly distributed delay helps to avoid collisions-otherwise all stations would try to transmit data after waiting for the medium becoming idle again plus DIFS.

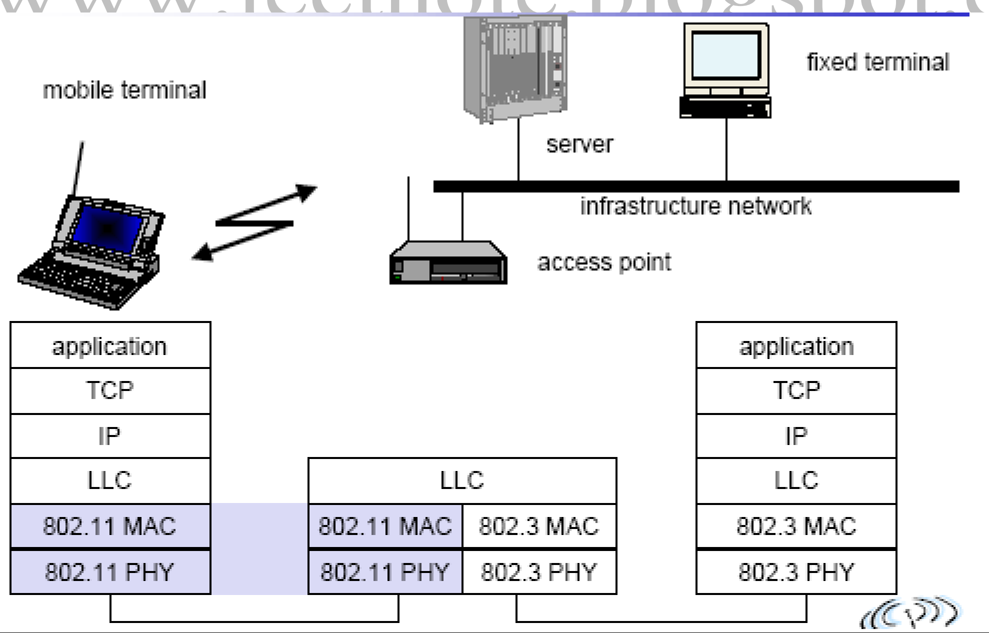
The basic CSMACA mechanism is not fair.Independent of the overall time a node has already waited for transmission; each node has the same chances for transmitting data in the next cyle.IEEE 802.11 adds a backoff timer .Again each node selects a random waiting time within the range of the contention window.If a certain station does not get access to the medium in the first cycle, it stops its backoff timer, waits for the channel to be idle again for DIFS and starts the counter again As soon as the counter expires, the node accesses the medium. This means that deferred stations do not choose a randomized backoff time again, but continue to count down. Station that have waited longer have the advantage over stations that have just entered, in that they only have to wait for the remainder of their backoff timer from the previous cycles.

DFWMAC-DCF with RTS/CTS extension.

Hidden terminal problem occurs if one station can receive two others,but those stations cannot receive each other.The two stations may sense the channes is idle ,send a frame, and cause a collision at the receiver in the middle.To deal with this problem, the standard defines an additional mechanism using two control packets, RTS and CTS.

DFWMAC-PCF with polling.

The **point co-ordinator** in the access point splits the access time into **super frame** periods. **A super frame** comprises a **contention free period** and a **contention period**.The contention period can be used for the two access mechanisms.



MAC

access mechanisms,fragmentation, encryption

MAC Management

synchronization, roaming, MIB,power management

PLCP Physical Layer Convergence Protocol

clear channel assessment signal (carrier sense)

PMD Physical Medium Dependent

modulation, coding PHY Management

qchannel selection, MIB Station Management

coordination of all management functions

802.11 - Physical layer

3 versions: 2 radio (typ. 2.4 GHz), 1 IR data rates 1 or 2 Mbit/s FHSS (Frequency Hopping Spread Spectrum) spreading, despreading, signal strength, typ. 1 Mbit/s

min. 2.5 frequency hops/s (USA), two-level GFSK modulation

DSSS (Direct Sequence Spread Spectrum)

DBPSK modulation for 1 Mbit/s (Differential Binary Phase Shift Keying), DQPSK for 2 Mbit/s (Differential Quadrature PSK) preamble and header of a frame is always transmitted with 1 Mbit/s, rest of transmission 1 or 2 Mbit/s chipping sequence: +1, -1, +1, +1, -1, +1, +1, +1, -1, -1, -1 (Barker code) max. radiated power 1 W (USA), 100 mW (EU), min. 1mW Infrared 850-950 nm, diffuse light, typ. 10 m range carrier detection, energy detection, synchronization.

FHSS PHY packet format

Synchronization

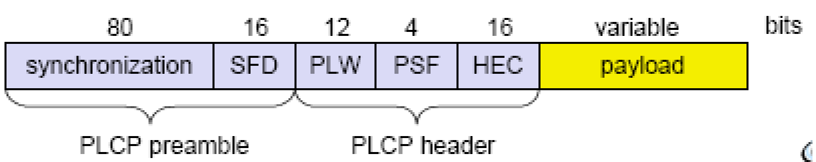
synch with 010101... pattern SFD (Start Frame Delimiter)

0000110010111101 start pattern PLW (PLCP_PDU Length Word)

length of payload incl. 32 bit CRC of payload, PLW < 4096 PSF (PLCP Signaling Field)

data of payload (1 or 2 Mbit/s) HEC (Header Error Check)

CRC with x16+x12+x5+1



5.5HiperLAN

HiperLAN (High Performance Radio LAN) is a Wireless LAN standard.[1] It is a European alternative for the IEEE 802.11 standards (the IEEE is an international organization). It is defined by the European Telecommunications Standards Institute (ETSI). In ETSI the standards are defined by the BRAN project (Broadband Radio Access Networks).
HiperLAN is a set of wireless local area network (WLAN) communication standards primarily used in European countries. There are two specifications: HiperLAN/1 and HiperLAN/2. Both have been adopted by the European Telecommunications Standards Institute (ETSI).

The HiperLAN standards provide features and capabilities similar to those of the IEEE 802.11 wireless local area network (LAN) standards, used in the U.S. and other adopting countries. HiperLAN/1 provides communications at up to 20 Mbps in the 5-GHz range of the radio frequency (RF) spectrum. HiperLAN/2 operates at up to 54 Mbps in the same RF band. HiperLAN/2 is compatible with 3G (third-generation) WLAN systems for sending and receiving data, images, and voice communications. HiperLAN/2 has the potential, and is intended, for implementation worldwide in conjunction with similar systems in the 5-GHz RF band.

HiperLAN/1

Planning for the first version of the standard, called HiperLAN/1, started 1991, when planning of 802.11 was already going on. The goal of the HiperLAN was the high data rate, higher than 802.11. The standard was approved in 1996. The functional specification is EN300652, the rest is in ETS300836.

The standard covers the Physical layer and the Media Access Control part of the Data link layer like 802.11. There is a new sublayer called Channel Access and Control sublayer (CAC). This sublayer deals with the access requests to the channels. The accomplishing of the request is dependent on the usage of the channel and the priority of the request.

CAC layer provides hierarchical independence with Elimination-Yield Non-Preemptive Multiple Access mechanism (EY-NPMA). EY-NPMA codes priority choices and other functions into one variable length radio pulse preceding the packet data. EY-NPMA enables the network to function with few collisions even though there would be a large number of users. Multimedia applications work in HiperLAN because of EY-NPMA priority mechanism. MAC layer defines protocols for routing, security and power saving and provides naturally data transfer to the upper layers.

On the physical layer FSK and GMSK modulations are used in HiperLAN/1.

HiperLAN features:

- range 50 m
- slow mobility (1.4 m/s)
- supports asynchronous and synchronous traffic
- sound 32 kbit/s, 10 ns latency
- video 2 Mbit/s, 100 ns latency
- data 10 Mbit/s

HiperLAN does not conflict with microwave and other kitchen appliances, which are on 2.4GHz.

HiperLAN/2

HiperLAN/2 functional specification was accomplished February 2000. Version 2 is designed as a fast wireless connection for many kinds of networks. Those are UMTS back bone network, ATM and IP networks. Also it works as a network at home like HiperLAN/1. HiperLAN/2 uses the 5 GHz band and up to 54 Mbit/s data rate.[1]

The physical layer of HiperLAN/2 is very similar to IEEE 802.11a wireless local area networks. However, the media access control (the multiple access protocol) is Dynamic TDMA in HiperLAN/2, while CSMA/CA is used in 802.11a.

Basic services in HiperLAN/2 are data, sound, and video transmission. The emphasis is in the quality of these services (QoS).[1]

The standard covers Physical, Data Link Control and Convergence layers. Convergence layer takes care of service dependent functionality between DLC and Network layer (OSI 3). Convergence sublayers can be used also on the physical layer to connect IP, ATM or UMTS networks. This feature makes HiperLAN/2 suitable for the wireless connection of various networks.

On the physical layer BPSK, QPSK, 16QAM or 64QAM modulations are used.

5.6Bluetooth

Bluetooth is an open wireless protocol for exchanging data over short distances from fixed and mobile devices, creating personal area networks (PANs). It was originally conceived as a wireless alternative to RS232 data cables. It can connect several devices, overcoming problems of synchronization.

Bluetooth uses a radio technology called frequency-hopping spread spectrum, which chops up the data being sent and transmits chunks of it on up to 79 frequencies. In its basic mode, the modulation is Gaussian frequency-shift keying (GFSK). It can achieve a gross data rate of 1 Mb/s. Bluetooth provides a way to connect and exchange information between devices such as mobile phones, telephones, laptops, personal computers, printers, Global Positioning System (GPS) receivers, digital cameras, and video game consoles through a secure, globally unlicensed Industrial, Scientific and Medical (ISM) 2.4 GHz short-range radio frequency bandwidth. The Bluetooth specifications are developed and licensed by the

Bluetooth Special Interest Group (SIG). The Bluetooth SIG consists of companies in the areas of telecommunication, computing, networking, and consumer electronics.

Uses

Bluetooth is a standard and communications protocol primarily designed for low power consumption, with a short range (power-class-dependent: 1 meter, 10 meters, 100 meters) based on low-cost transceiver microchips in each device. Bluetooth makes it possible for these devices to communicate with each other when they are in range. Because the devices use a radio (broadcast) communications system, they do not have to be in line of sight of each other.

Bluetooth profiles

In order to use Bluetooth, a device must be compatible with certain Bluetooth profiles. These define the possible applications and uses of the technology.

More prevalent applications of Bluetooth include:

- Wireless control of and communication between a mobile phone and a hands-free headset. This was one of the earliest applications to become popular.
- Wireless networking between PCs in a confined space and where little bandwidth is required.
- Wireless communication with PC input and output devices, the most common being the mouse, keyboard and printer.
- Transfer of files, contact details, calendar appointments, and reminders between devices with OBEX.
- Replacement of traditional wired serial communications in test equipment, GPS receivers, medical equipment, bar code scanners, and traffic control devices.
- For controls where infrared was traditionally used.
- For low bandwidth applications where higher [USB] bandwidth is not required and cable-free connection desired.

- Sending small advertisements from Bluetooth-enabled advertising hoardings to other, discoverable, Bluetooth devices[citation needed].
- Wireless bridge between two Industrial Ethernet (e.g. PROFINET) networks.
- Two seventh-generation game consoles, Nintendo's Wii[6] and Sony's PlayStation 3, use Bluetooth for their respective wireless controllers.
- Dial-up internet access on personal computers or PDAs using a data-capable mobile phone as a modem.

Bluetooth protocol stack

Bluetooth stack and Bluetooth protocols

Bluetooth is defined as a layer protocol architecture consisting of core protocols, cable replacement protocols, telephony control protocols, and adopted protocols

Mandatory protocols for all Bluetooth stacks are: LMP, L2CAP and SDP

Additionally, these protocols are almost universally supported: HCI and RFCOMM

LMP (Link Management Protocol)

Used for control of the radio link between two devices. Implemented on the controller.

L2CAP (Logical Link Control & Adaptation Protocol)

Used to multiplex multiple logical connections between two devices using different higher level protocols. Provides segmentation and reassembly of on-air packets.

In Basic mode, L2CAP provides packets with a payload configurable up to 64kB, with 672 bytes as the default MTU, and 48 bytes as the minimum mandatory supported MTU.

In Retransmission & Flow Control modes, L2CAP can be configured for reliable or isochronous data per channel by performing retransmissions and CRC checks.

Bluetooth Core Specification Addendum 1 adds two additional L2CAP modes to the core specification. These modes effectively deprecate original Retransmission and Flow Control modes:

- **Enhanced Retransmission Mode** (ERTM): This mode is an improved version of the original retransmission mode. This mode provides a reliable L2CAP channel.
- **Streaming Mode** (SM): This is a very simple mode, with no retransmission or flow control. This mode provides an unreliable L2CAP channel.

Reliability in any of these modes is optionally and/or additionally guaranteed by the lower layer Bluetooth BDR/EDR air interface by configuring the number of retransmissions and flush timeout (time after which the radio will flush packets). In-order sequencing is guaranteed by the lower layer.

Only L2CAP channels configured in ERTM or SM may be operated over AMP logical links.

L2CAP

The **(L2CAP)** is a data link control protocol on top of the base band layer offering logical channels between Bluetooth devices with QoS properties. L2CAP is available for ACL5 only. Audio applications using SCOs hive tc use the baseband layer directly (see Figure 7.44). L2CAP provides thre different types of logical channels that are transported via the ACL between = **master** and slave:

• Connectionless

• Connection-oriented:

• Signaling:

SDP (Service Discovery Protocol)

Used to allow devices to discover what services each other support, and what parameters to use to connect to them. For example, when connecting a mobile phone to a Bluetooth headset, SDP will be used to determine which Bluetooth profiles are supported by the headset (Headset Profile, Hands Free Profile, Advanced Audio Distribution Profile etc) and the protocol multiplexer settings needed to connect to each of them. Each service is identified by a Universally Unique Identifier (UUID), with official services (Bluetooth profiles) assigned a short form UUID (16 bits rather than the full 128)

HCI (Host/Controller Interface)

Standardised communication between the host stack (e.g. a PC or mobile phone OS) and the controller (the Bluetooth I.C.) This standard allows the host stack or controller I.C. to be swapped with minimal adaptation.

There are several HCI transport layer standards, each using a different hardware interface to transfer the same command, event and data packets. The most commonly used are USB (in PCs) and UART (in mobile phones and PDAs).

In Bluetooth devices with simple functionality, e.g. headsets, the host stack and controller can be implemented on the same microprocessor. In this case the HCI is optional, although often implemented as an internal software interface.

Communication and connection

A master Bluetooth device can communicate with up to seven devices in a Wireless User Group. This network group of up to eight devices is called a piconet.

A piconet is an ad-hoc computer network, using Bluetooth technology protocols to allow one master device to interconnect with up to seven active devices. Up to 255 further devices can be inactive, or parked, which the master device can bring into active status at any time.

At any given time, data can be transferred between the master and one other device, however, the devices can switch roles and the slave can become the master at any time. The master switches rapidly from one device to another in a round-robin fashion. (Simultaneous transmission from the master to multiple other devices is possible, but not used much.)

The Bluetooth specification allows connecting two or more piconets together to form a scatternet, with some devices acting as a bridge by simultaneously playing the master role in one piconet and the slave role in another.

Many USB Bluetooth adapters are available, some of which also include an IrDA adapter. Older (pre-2003) Bluetooth adapters, however, have limited services, offering only the Bluetooth Enumerator and a less-powerful Bluetooth Radio incarnation. Such devices can link computers with Bluetooth, but they do not offer much in the way of services that modern adapters do.

Baseband Error Correction

Three types of error correction are implemented in Bluetooth systems,

- 1/3 rate (Forward Error Correction) (FEC)
- 2/3 rate FEC
- **A**utomatic **R**epeat Re**q**uest (ARQ)

Setting up connections

Any Bluetooth device will transmit the following information on demand:

- Device name.
- Device class.
- List of services.
- Technical information, for example, device features, manufacturer, Bluetooth specification used, clock offset.

Any device may perform an inquiry to find other devices to connect to, and any device can be configured to respond to such inquiries. However, if the device trying to connect knows the address of the device, it always responds to direct connection requests and transmits the information shown in the list above if requested. Use of a device's services may require pairing or acceptance by its owner, but the connection itself can be initiated by any device and held until it goes out of range. Some devices can be connected to only one device at a time, and connecting to them prevents them from connecting to other devices and appearing in inquiries until they disconnect from the other device.

Every device has a unique 48-bit address. However these addresses are generally not shown in inquiries. Instead, friendly Bluetooth names are used, which can be set by the user. This name appears when another user scans for devices and in lists of paired devices.

Most phones have the Bluetooth name set to the manufacturer and model of the phone by default. Most phones and laptops show only the Bluetooth names and special programs are required to get additional information about remote devices. This can be confusing as, for example, there could be several phones in range named T610 (see Bluejacking).

Pairing

Pairs of devices may establish a relationship by creating a shared secret known as a *link key*, this process is known as *pairing*. If a link key is stored by both devices they are said to be *bonded*. A device that wants to communicate only with a bonded device can cryptographically authenticate the identity of the other device, and so be sure that it is the same device it previously paired with. Once a link key has been generated, an authenticated ACL link between the devices may be encrypted so that the data that they exchange over the airwaves is protected against eavesdropping. Link keys can be deleted at any time by either device, if done by either device this will implicitly remove the bonding between the devices; so it is possible one of the device to have a link key stored but not be aware that it is no longer bonded to the device associated with the given link key.

Bluetooth services generally require either encryption or authentication, as such require pairing before they allow a remote device to use the given service. Some services, such as the Object Push Profile, elect not to explicitly require authentication or encryption so that pairing does not interfere with the user experience associated with the service use-cases.

Pairing mechanisms have changed significantly with the introduction of Secure Simple Pairing in Bluetooth 2.1. The following summarizes the pairing mechanisms:

- **Legacy pairing**: This is the only method available before Bluetooth 2.1. Each device must enter a PIN code, pairing is only successful if both devices enter the same PIN code. Any 16-digit ACSII string may be used as a PIN code, however not all devices may be capable of entering all possible PIN codes.
    - **Limited Input Devices**: The obvious example of this class of device is a Bluetooth Hands-free headset, which generally have few inputs. These devices usually have a *fixed PIN*, for example "0000" or "1234", that are hard-coded into the device.
    - **Numeric Input Devices**: Mobile phones are classic examples of these devices. They allow a user to enter a numeric value up to 16 digits in length.
    - **Alpha-numeric Input Devices**: PCs and smartphones are examples of these devices. They allow a user to enter full ASCII text as a PIN code. If pairing with a less capable device the user needs to be aware of the input limitations on the other device, there is no mechanism available for a capable device to determine how it should limit the available input a user may use.
- **Secure Simple Pairing**: This is required by Bluetooth 2.1. A Bluetooth 2.1 device may only use legacy pairing to interoperable with a 2.0 or older device. Secure Simple Pairing uses a type of public key cryptography, and has the following modes of operation:
    - **Just Works**: As implied by the name, this method just works. No user interaction is required; however, a device may prompt the user to confirm the pairing process. This method is typically used by headsets with very limited IO capabilities, and is more secure than the fixed PIN mechanism which is typical for this set of limited devices. This method provides no man in the middle (MITM) protection.
    - **Numeric Comparison**: If both devices have a display and at least one can accept a binary Yes/No user input, they may use Numeric Comparison. This method displays a 6-digit numeric code on each device. The user should compare the numbers to insure they are identical. If the comparison succeeds, the user(s) should confirm pairing on the device(s) that can accept an input. This method provides MITM protection, assuming the user confirms on both devices and actually performs the comparison properly.

- **Passkey Entry**: This method may be used between a device with a display and a device with numeric keypad entry (such as a keyboard), or two devices with numeric keypad entry. In the first case, the display is used to show a 6-digit numeric code to the user, who then enters the code on the keypad. In the second case, the user of each device enters the same 6-digit number. Both cases provide MITM protection.
- **Out of Band** (OOB): This method uses an external means of communication (such as NFC) to exchange some information used in the pairing process. Pairing is completed using the Bluetooth radio, but requires information from the OOB mechanism. This method provides some level of MITM protection, assuming the OOB method used provides MITM.

Security

A radio interface is by nature easy to access. Bluetooth devices can transmit private data, e.g., schedules between a PDA and a mobile phone. A user clearly does not want another person to eavesdrop the data transfer. Just imagine a scenario where two Bluetooth enabled PDAs in suitcases 'meet' on the conveyor belt of an airport exchanging personal information! Bluetooth offers mechanisms for authentication and encryption on the MAC layer, which must be Implemented in the same way within each device.

The main security features offered by Bluetooth include a challenge- response routine for authentication, a stream cipher for encryption, and a session key generation. Each connection may require a one-way, two-way, or no authentication using the challenge-response routine. All these schemes have to be implemented in silicon, and higher layers should offer stronger encryption if needed. The security features included in Bluetooth only help to set up a local domain of trust between devices.

The security algorithms use the public identity of a device, a secret private User key, and an internally generated random key as input parameters. For each transaction, a new random number is generated on the Bluetooth chip. Key management is left to higher layer software.

QUESTION BANK

PART A

1) What are the basic elements of an Ethernet system?
2) How is CSMA/CD related to ALOHA?
3) What are the services available in an ISDN?
4) Define B-ISDN and its features?
5) What is adaptation layer?
6) Explain statistical multiplexing of ATM networks and why it is needed?
7) What are the unique features of Geosynchronous satellites?
8) State Sampling (Nyquist) theorem?
9) Differentiate FDM and TDM?
10) What is the function of Bluetooth?
11) What is DIX standard?
12) Differentiate persistent and nonpersistent CSMA?
13) What are the four fields in the ATM cell header?
14) Write any two advantages of ISDN?
15) Explain a SONET frame?
16) What are the two main types of ISDN services based on units of bandwidth?
17) What is footprint?
18) What are VSATS? Give their advantages and disadvantages?
19) Compare and contrast IR and Radio transmission technology?
20) Compare IEEE 802.11a and HIPERLAN 2?
21) What is SONET?
22) Write a short note on 10mbps Ethernet?
23) Write any two advantages of ISDN?
24) Give the format of ATM header?
25) Compare AAL 1 and AAL 2?
26) What is UNI signaling?
27) Give advantage of VSAT?
28) Write some application of VSAT?

29) What is IR transmission technology?
30) Compare infrastructure and adhoc networks?
31) How does Fast Ethernet differ from Ethernet?
32) Compare a connection oriented network service with a connectionless network service?
33) Write the various principles of ISDN?
34) What do these terms denote
    a. NT1 b) NT2 c) TE1 d) TA with respect to ISDN user-network interface?
35) Compare synchronous transfer mode and asynchronous transfer mode?
36) Write the frame format of SONET?
37) What are the frequency bands used in satellites? Write their ranges?
38) What is the principal advantage of Geostationary satellite? Why is it called so?
39) What are the software and hardware requirements to connect to the internet?
40) Compare a shell account with PPP account? What is DNS?
41) Write the advantages of using optic fibre over other medium?
42) What is UTP? Compare UTP and STP?
43) What are the types of channels available in ISDN? What are the functions of these channels?

44) Write the ISDN address structure?
45) How is a cell established in ATM using virtual paths?
46) What is SONET? What are the features of SONET?
47) Draw the functional block diagram of a digital earth station. Explain the purpose of various blocks in it?
48) Compare Wireless LANs with Wired LANs?
49) What is Internet? How has Internet evolved over the years?
50) What are the different classes of IP address? How they differ?

PART B

1) 1.Draw Ethernet switching hub schematic and switching hub derivative and explain its working?
2) What happens in a token bus if a station accepts the token and then crashes immediately? If the token ring is of 4mbps and has a token holding timer value of 10ms. What is the longest frame that can be sent in this ring?
3) Explain the ISDN architecture using suitable diagram?
4) What is the structure of the frame format of ISDN and explain protocols?
5) If 1024-byte message is sent with AAL 3/4, what is the efficiency obtained? What fraction of the transmitted bits are useful data bits? If AAL 5 is used. What are the answers?
6) Explain with architecture ATM adaptation layer and the B-ISDN reference model?
7) Explain the basic principle of satellite communication system by drawing the earth station diagram?
8) How do the satellite communications become the back-bone of the communication/transmission system?

9) Explain MAC sublayer and HYPERLAN?
10) Explain the extend of security in WLANS with respect to various parameters?
11) Give the frame format of Ethernet?
12) Write a note on Gigabit Ethernet?
13) Explain ISDN protocol architecture and transmission channels
14) What is UNI signaling and PNNI signaling?
15) Write the principle characteristics of ATM and explain the three layers?
16) Write in detail about BISDN reference model?
17) Give block schematic of satellite earth station?
18) Explain the satellite communication principles?
19) Explain IEEE 802.11 standard with its system protocol and architecture?
20) Explain the Bluetooth protocol stack in detail?
21) Compare and contrast Fast Ethernet, Gigabit Ethernet and Wireless Ethernet?
22) Explain SONET multiplexing and SONET frame structure?
23) Write a note on ISDN protocol architecture?

24) Explain the BISDN reference model?
25) Describe the three layers of ATM?
26) Explain VSAT networks?
27) Write in detail about Geostationary satellites?
28) Compare IR and Radio transmission?
29) Write the advantages and disadvantages of Bluetooth technology?
30) Describe the architecture of IEEE 802.11 standard?
31) What does the term topology refer to? What are the common topologies used for LANs? Explain them?
32) What is the relationship between a medium and a topology? What is FDDI and explain its uses?
33) What are the protocols used in ISDN at the various layers? Explain their purpose?
34) What is a B-ISDN and explain its requirements? How does B-ISDN differ from ISDN?
35) Explain the concept of Virtual channel and Virtual path switching?
36) What is VSAT? Explain VSAT star network with carrier access configurations?

37) What is WLAN? Explain components, working and advantages of WLANS?
38) What is packet and packet switching? Explain how networks accessed through packet switching?
39) What is dedicated line? Write the advantage of dedicated and leased lines? How can Internet be accessed using satellites?
40) Explain the various transmission media used in LAN. When will you go for a practical media and based on what reasons?
41) .What is circuit switching? How are networks connected using circiuit switching?
42) Explain the functional diagram of B-ISDN layers with the protocols used?
43) What are the various ATM services? What is the role of ATM in the Internet?

44) What are the different error control strategies followed in VSAT network? Write on Wireless LAN standards? Give some examples for wireless LAN receiving devices. How do they operate? What are the applications of wireless LANs? Are there any disadvantages of wireless LANs? If yes, what are they?
45) What are the requirements for wireless LANs? Explain the working of microwave LANs?