UNIT-IV -ISDN & ATM

1. History of analog and digital network
2. Access to ISDN
3. ISDN layers
4. Broadband ISDN
5. Packet layer protocol
6. ATM
7. ATM architecture
8. ATM layers
9. Congestion control
10. Leaky bucket algorithm.

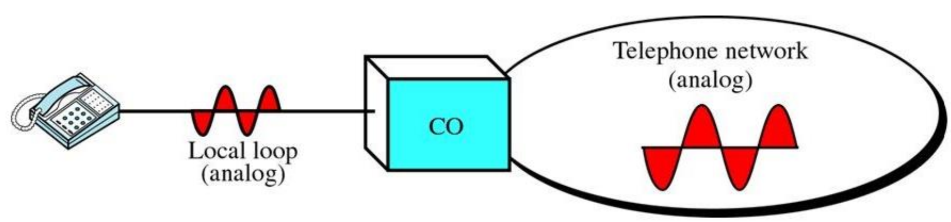
1 History:

1.1 Voice Communication over analog n/ws:

- Initially, telecommunication n/ws were entirely analog n/ws and were used for the transmission of analog information in the form of voice.
- Local loops connecting the subscriber's handset to the telephone company's central office were also analog.

Figure 16-2

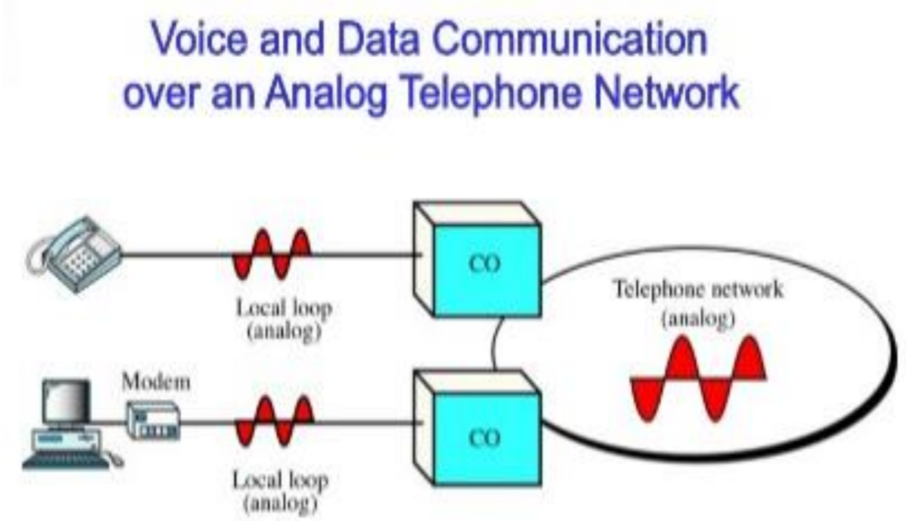
Voice Communication over an Analog Telephone Network



1.2 Voice and data communication over Analog n/ws:

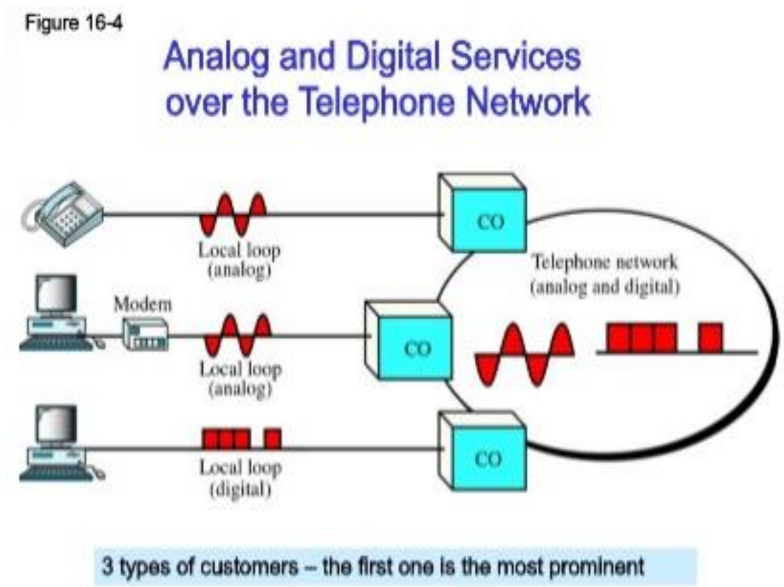
- With the advent of digital processing, subscribers needed to exchange data as well as voice.
- Modems, developed to allow digital exchanges over existing analog lines.

Figure 16-3



1.3 Analog and digital services to subscribers:

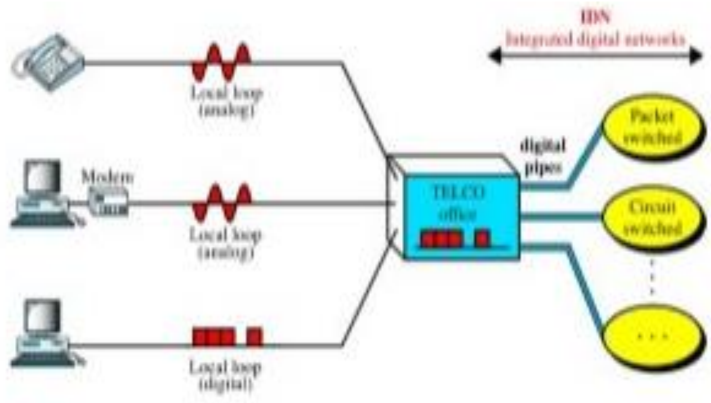
- To reduce cost and improve performance, telephone companies began to add digital technologies while continuing their analog services to customers.
- Three types of customers: Traditional Customers using their land loops only for analog purposes, customers suing analog facilities to transmit digital information via modem, customers using digital services to transmit data i.e. digital information.
- First group was prominent, so most services offered remained analog.



2. ISDN

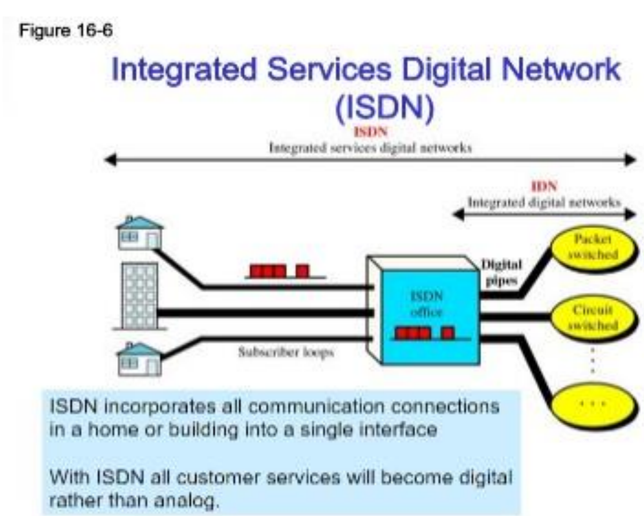
2.1 Integrated Digital N/w (1DN):

- Customers needed access to variety of N/ws, such as packet switched n/ws and circuit switched n/ws, so telephone companies created 1DNs.
- An IDN is a combination of n/ws available for different purposes.
- Access to these n/ws is by digital pipes, which are time-multiplexed channels sharing very high speed paths.



2.2 Integrated Services Digital N/w (ISDN):

- ISDN integrates customer services with the IDN.
- With ISDN all customer services will become digital rather than analog and the flexibility offered by the new technology will allow customers services to be made available on demand.
- ISDN incorporates all communications connections in a home or building into a single interface.



2.3 Subscriber Access to the ISDN;

ISDN standard defines 3 channel types, each with a different transmission rate: bearer channels, data channels and hybrid channels.

2.4 Channel rates

Channel	Data Rate (Kbps)
Bearer (B)	64
Date (D)	16, 64
Hybrid (H)	384, 1536 1920

2.4.1 B Channels:

- A bearer channel defined at a rate of 64 kbps.
- It is a basic user channel and can carry any type of digital information in full duplex mode.
- B – Channel carries transmissions end to end.

2.4.2 D Channels:

- A data channel can be either 16 or 64 kbps, depending on the needs of user.
- Primary function is to carry control signaling for the B channels, using a method called common channel (out of band) signaling.
- D channel acts like an operator between the user and the n/w at the n/w layer
- Less common uses for D channel include low rate data transfer and applications such as telemetry and alarm transmission.

2.4.3 H Channels:

- Hybrid channels are available with data rates of 384 Kbps (HO), 1536 K6ps (H11), or 1920 Kbps (H12).
- These rates suit H channels for high data rate applications such as video, tele-conferencing etc.

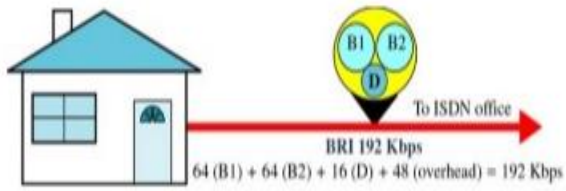
2.5 User Interfaces:

- Digital subscriber loops are of 2 types: basic rate interface (BRI) and primary rate interface (PRI).
- Both include one D channel and some number of either B or H channels.

2.5.1 BRI:

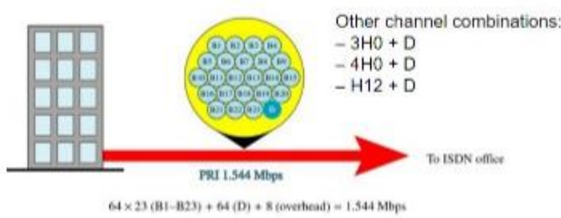
- It specifies a digital pipe consisting of 2 B channels and one 16 K6ps D channel.
- 2 B channels of 64 K6ps each, plus 1 D channel of 16 K6ps, equals 144 K6ps.
- Additionally, BRI itself requires 48 K6ps of operating overhead.
- Therefore, requires a digital pipe of 192 K6ps.
- BRI service is like a large pipe with 3 smaller pipes, remainder of space inside the large pipe carries the overhead bits required for its operation.
- BRI is designed to meet the needs of residential & small office customers.

- Same twisted pair local loop that delivers analog transmission can be used to handle digital transmission.



2.5.2 PRI:

- It specifies a digital pipe with 23 B channels & one 64 Kbps D channel.
- 23 B channels of 64 Kbps each, plus 1 D channel of 64 K6ps, equals 1.536 Mbps.
- Additionally, PRI itself uses 8 Kbps of overhead. digital pipe of 1.544 Mbps.
- So PRI is a large pipe with 24 smaller pipes, 23 for B & 1 for D channels.
- Rest of the pipe carries the overhead bits required for its operation.
- The individual transmissions are collected from their sources & multiplexed onto a single path for sending to the ISDN office.
- For more specialized transmission needs, other channel combinations are also supported by PRI standard. They are 3HO+D, 4HO+D & H12+D.



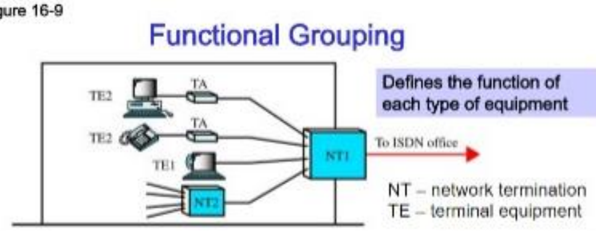
2.6 Functional Grouping:

- In ISND standard, the devices that enable users to access the services of the BRI or PRI are described by their functional duties and collected in functional grouping.
- Each functional grouping is a model that can be implemented using devices or equipment chosen by the subscriber.
- Functional groupings used are n/w terminations (types 1&2) terminal equipment (types 1 & 2) terminal adapters.

2.6.1 N/w Termination 1 (NT10:

- An NT1 device controls the physical & electrical termination of the ISDN at the users premises and connects the users internal system to the digital subscriber loop.
- An NT1 organizes the data streams from a connected subscriber into frames that can be sent over the digital pipe, and translates the frames received from the n/w into a format usable by the subscriber's devices.
- An NT1 synchronizes the data stream with the frame – building process in such a way that multiplexing occurs automatically.

Figure 16-9



2.6.2 N/W Termination 2 (NT2)

- An NT2 device performs functions at the physical, data link & n/w layers of the OSI model (layers 1,2& 3).
- NT2s provide multiplexing (layer1), flow control (layer 2) & packetizing (layer 3).

- An NT2 provides intermediate signal processing between the data-generating devices and an NT1.
- NT2s can be implemented by a variety of equipment types. eg. a private branch exchange (digital PBX) can be an NT2 – it coordinates transmissions from a number of incoming links (user phone lines) and multiplexes them to make them transmittable by an NT1.
- A LAN also can function as an NT2.

2.6.3 Terminal Equipment 1 (TE1):

- The term terminal equipment is used by ISDN std. to mean the same thing as DTE in other protocols.
- It refers to digital subscriber equipment.
- TE1 is any device that supports the ISDN stds.
- Egs. Digital telephones, integrated voice / data terminals & digital facsimiles.

2.6.4 Terminal Equipment 2 (TE2):

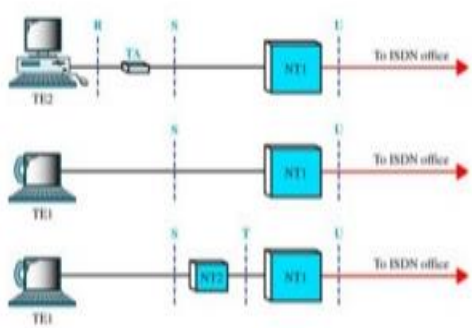
- To provide backward compatibility with a customer's existing equipment, the ISDN std. defines a second level of terminal equipment called TE2.
- TE2 is any non – ISDN device, such as terminal, workstation, host computer or regular telephone.
- TE2 devices are not immediately compatible with an ISDN n/w but can be used with the help of another device called a terminal adapter (TA).

2.6.5 <u>Terminal Adapter (TA):</u>

- TA converts information received in non- ISDN format from a TE2 into a format capable of being carried by the ISDN.

2.7 <u>Reference points:</u>

- Here, reference points refer to the label used to identify individual interfaces between 2 elements of an ISDN installation.
- Functional grouping defines the function of each types of equipment used in ISDN, reference points defines the functions of the connections between them.
- Specifically, a reference point defines how 2 n/w elements must be connected and the format of the traffic between them.
- Reference points R, S,T and U can be seen in the figure.

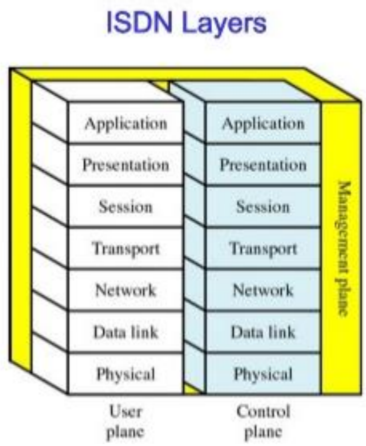


- Reference point R defines the connection between a TE2 & a TA; Reference point S defines the connection between a TE1 or TA & an NT1 or NT2 (if present); Reference point T defines the interface between an NT2 & an NT1; Reference point U defines the interface between an NT1 and the ISDN office.

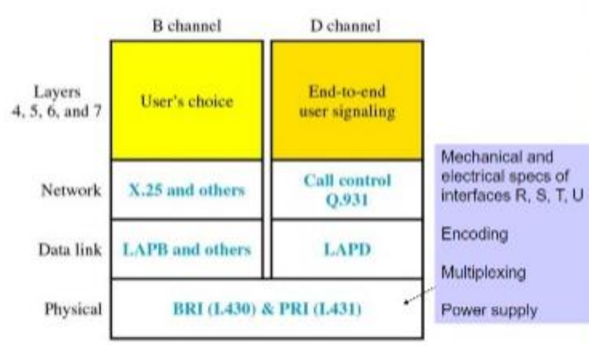
3 The ISDN layers:

- The ITU-T has devised an expanded model for the ISDN layers.
- Instead of a single 7 layer architecture like the OSI, the ISDN is defined in the three separate planes: the user plane, the control plane and the management plane.

Figure 16-11



- All 3 planes divided into 7 layers that correspond to the OSI model.

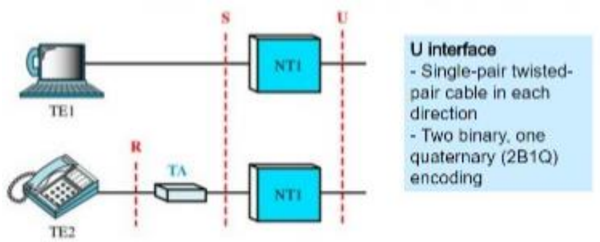


3.1 Physical layer:

- The ISDN physical layer specifications are defined by 2 ITU-T stds. I.430 for BRI access and I. 431 for PRI access.
- These stds. define all aspects of the BRI and PRI. Of these 4 are important.
- The mechanical & electrical specifications of interfaces R,S,T&U
- Encoding
- Multiplexing channels to make them carriable by the BRI& PRI digital pipes power supply.

3.1.1 Physical layer specifications for BRI:

- BRI consists of 2B channels & one D channel.
- A subscriber connects to the BRI using the R, S, & U interfaces (reference points).



3.1.2 R interface:

- This interface not defined by the ISDN.
- A subscriber can use any of the EIAstds (such as E1A-232, E1A-499, or E1A-530) or any of the V or X series standards (such as x .21).

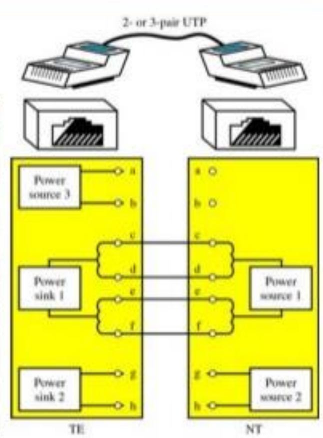
3.1.3 S interface:

- For S interface, ITU-T specifies the ISO std. ISO8887.
- This std. calls for four -, six-, or eight wire connections.

Figure 16-14

S Interface

Name	TE	NT
a	Power source 3	Power sink 3
b	Power source 3	Power sink 3
c	Transmit	Receive
d	Receive	Transmit
e	Receive	Transmit
f	Transmit	Receive
g	Power sink 2	Power source 2
h	Power sink 2	Power source 2

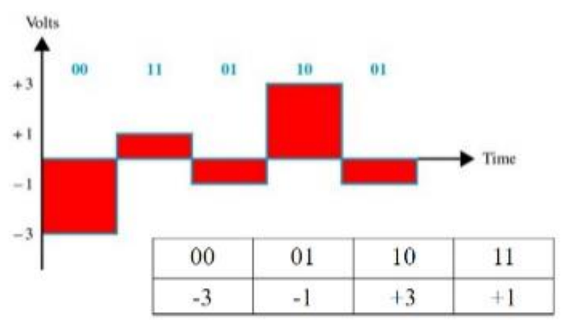


- The signal used in the sinterface is preudoternary encoding.

3.1.4 U-interface:

- For U interface (digital subscriber or local loop), the ITU –T specifies a single pair twisted – pair cable in each direction.
- Encoding for this interface uses a method called two binary, one quaternary (2B1Q).
- 2B1Q uses 4 voltage levels instead of two.
- The 4 voltage levels represent the bits 00, 01, 10 & 11.

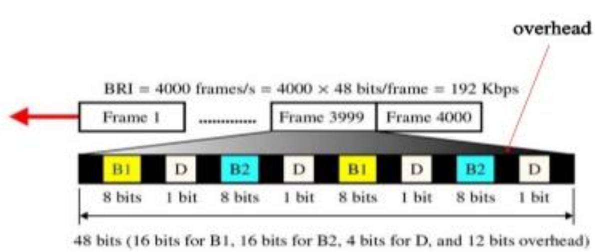
Figure 16-15
2B/1Q Encoding



3.1.5 BRI frame:

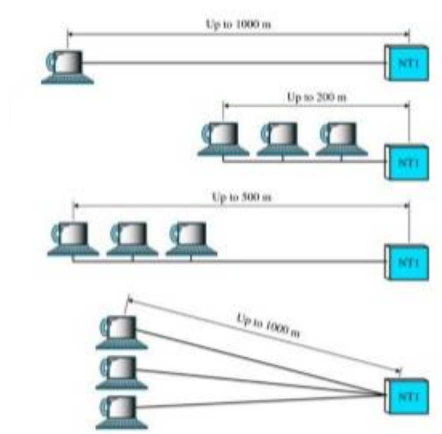
- Each B channel is sampled twice during each frame (8 bits per sample).
- The D channel is sampled four times during each frame (one bit per sample).
- The entire frame consists of 48 bits : 32 bits for the B Channels, 4 bits for the D channel, & 12 bits of overhead.

Figure 16-16
BRI Frame



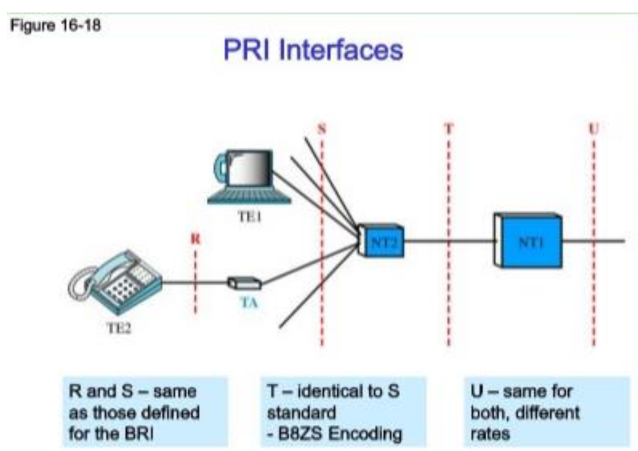
3.1.6 Connection and Topology:

- BRI services can be supported by either a bus or star topology.
- The main restriction governing the choice of topology for a BRI is the distance of the data devices from the NT1.
- In a point to point connection, each device can be as far as 1000 meters away from the NT1.
- In a multipoint connection, the maximum length of the line generally cannot be more than 200 meters.
- This restriction is necessary to ensure frame synchronization.
- If the synchronization between the frame and the devices is off, data dumped by one device can end up in a part of the frame devoted to data from another device, or to another kind of information altogether.
- Unavoidable propagation delays over distance can result in a shifted frame.
- IF the distance between the first and the last device on a link is great enough, data collection timing can deteriorate the frame.
- Clustering the devices means that propagation delays will impact the data from all devices almost equally, allowing the relationship between the data units to remain predictable for 500 meters.
- As many as 8 devices can be connected to an NT1.
- Of these, only 2 can access the B channels at one time, one exchange per channel.
- Every device, however can contend for access tot eh D channel.
- D Channels use a mechanism like CSMA to control access.
- Once device has access to D channel, it can request a B channel.
- If B channel is available, the connection is made by the D channel and the user may then send data.



3.1.7 Physical layer specifications for PRI:

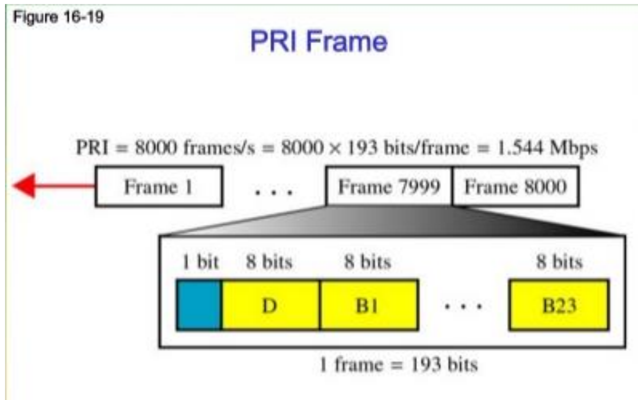
- The PRI consists of 23B channels and 1 D channel.
- Interfaces associated with PRI usage include R.S.T. & U.
- The R&Sstds. are the same as those defined for the BRI.
- The T std. is identical to S Std. with the substitution of B8ZS encoding.
- The U interface is the same for both stds. except that the PRI rate is 1.544 Mbps instead of 192 Kbps: 1.544 Mbps allows the PRI to be implemented using T-1 specifications.



3.1.8 PRI frame:

- The B and D channels are multiplexed using synchronous TDM to create a PRI frame.
- The frame formats identical to that defined for T-1 lines.

- PRI frame samples each channel, including the D channel, only once per frame.



3.1.9 Connection and Topology

- Can be the same as those described for the device to NT1 links in the BRI, or they can differ (depends on application).
- The link from NT2 to NT1 must always be point to point.
- If NT2 is LAN, its topology will be specified by LAN being used, If NT2 is a PBX, its topology will be specified by the PBX being used & so on.

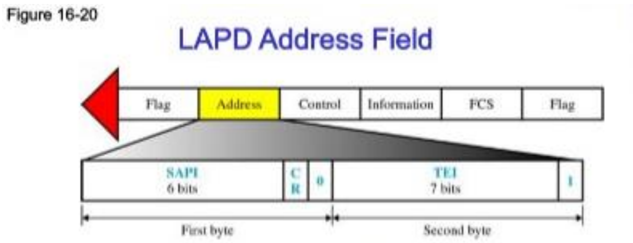
3.2 Data Link Layer:

- B and D channels use different data link protocols, B channels use LAPB protect.
- The D channel uses link access procedure for D channel (LAPD).
- First, LAPD can be used in either unacknowledged (without sequence numbering) or acknowledged (with sequence numbering) formats.
- The unacknowledged format is used only seldomly.

3.2.1 LAPD addressing:

- The address field of the LAPD is 2 bytes long.
- The first byte contains a b – bit field called a service access point identifier (SAP1) a 1-bit command / response field set to 0 if the frame

is a command and to 1 if the frame is a response I and a 1 bit field set to 0 to indicate that the address is continued in the next byte.



- The 2nd byte contains a 7 bit field called a terminal equipment identifier (TEI) and a one – bit field set to 1 to indicate that the address is complete.

3.2.2 <u>SAPI field:</u>

- Identifies the type of upper – layer service (n/w layer) using the frame.
- It indicates the intended use of the D channel.
- It is a 6 – bit field and can therefore define upto 64 different service access points.
- To date, only 4 of the possible bit combinations have been assigned.

  000000 - Call control for n/w layer (signaling use of D channel)

  000001 - Call control for upper layer (end to end signaling), not yet in use.

  010000 - Packet communication (data use of D channel).

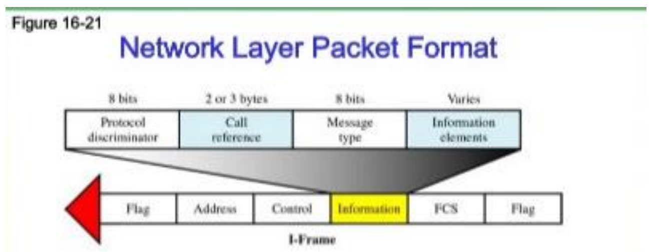
  111111 - Management.

3.2.3 <u>TE1 field:</u>

- The TE1 field is the unique address of the TE.
- It consists of 7 bits and can therefore identify upto128 different TEs.

3.3 Network Layer:

- Once a connection has been established by the D channel, the B Channel sends data using circuit switching, x .25, or other similar protocols.
- The functions are defined by the ITU-T Q. 931 standard.
- The network layer packet is called a message.
- A message is encapsulated in the information field of an LAPD I frame for transport across a link.



- The format of the message have 4 fields:

  Protocol discrimination (a single one byte field).

  Call reference (2-or3 byte field)

  Message type (a single one byte field)

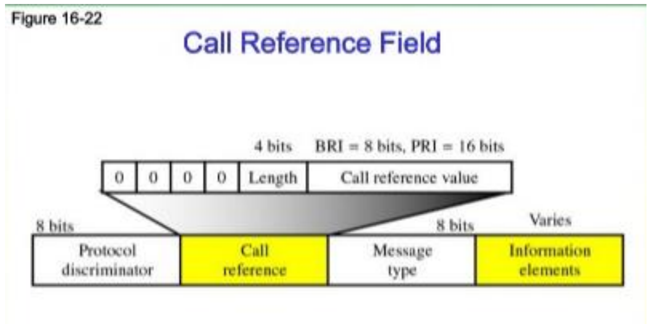
  Information elements (a variable number of variable length fields).

3.3.1 Protocol discriminator:

- This field identifies the protocol in use. For Q. 931, the value of this field is 00001000.

3.3.2 Call Reference:

- It is the sequence number of the call. The format is shown in figure.



3.3.3 Message Type:

- It is a one byte field that identifies the purpose of the message.
- 4 categories of message types: call establishment messages, call information messages, call clearing messages, & miscellaneous messages.

3.3.4 Call Establishment Message:

- Setup – sent by the calling user to the n/w or by the n/w to the called user to initiate a call.
- Setup Acknolwedgement – sent by called user to n/w or by n/w to calling user to indicate that setup has been received (no connection).
- Connect – sent by called user to n/w or by n/w to calling user to indicate acceptance of the call.
- Connect acknowledgement – sent by the n/w to the called user to say that the desired connection has been awarded.
- Progress – sent by n/w to called user to indicate that call establishment is in progress.
- Alerting – sent by the called user to n/w or by n/w to calling user to indicate that the call user alert (ringing) has been initiated.
- Call Processing – sent by called user to n/w or by n/w to the calling user to indicate that the requested call establishment has been initiated and that no more information is needed.

3.3.5 Call Information Message:

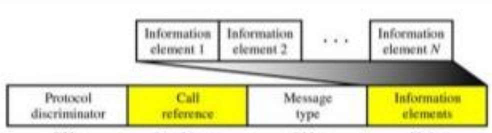
- Resume – sent by a user to the n/w to request that a responded call be resumed.
- Resume Acknowledgement – sent by the n/w to the user to acknowledge a request to resume the call.
- Suspend – sent by a user to request that the n/w suspend a call.
- Suspend acknowledgement – sent by n/w to user to acknowledge the requested suspension of the call.
- Suspend reject – sent by n/w to user to reject the requested suspension.
- User Information – sent by user to n/w to be delivered to the remote user.

3.3.6 Call Clearing Messages:

- Disconnect – sent by the called user to the n/w or by the n/w to the called user to clear the end to end connection (termination).
- Release – sent by a user or n/w to indicate the intention to disconnect and release the channel.
- Release complete – sent by a user or n/w to show that the channel has been released.
- Miscellaneous – other messages carry information defined in the protocols of specific services.

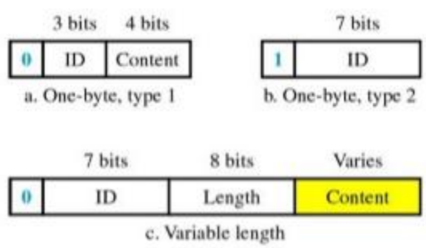
3.3.7 Information Elements:

- An information elements field carries specific details about the connection that are required for call establishment.



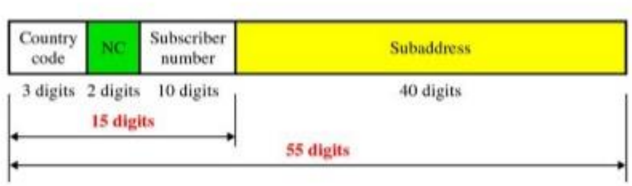
3.3.8 Information Element types:

- An information element consists of one or more bytes.
- A one byte information element can be of type 1 or type 2.
- In type 1, the 1st bit is 0, the next 3 bits identify the information being sent, and the remaining 4 bits carry the specific content or attribute of the element.
- Type 2 elements start with a 1 bit, the remainder of the bit reserved for the ID.
- In multi byte information element, the 1st bit of the 1st byte is 0 and the remainder of the byte is the ID.
- The 2nd byte defines the length of the content in bytes, the remaining bytes are content.



3.3.9 Addressing:

- An important type of information element is addressing.



- The country code consists of 3 digits.
- The NC field is the national code and consists of 2 digits.
- It identifies the specific n/w in countries with more than one ISDN n/w.
- The subscriber number is the 10 digit number familiar from national telephone numbers; a 3 digit area code & a 7 digit phone number.

- Together these, 15 digits define the access to a subscriber NT1.
- An NT1 may have multiple devices connected to it, either directly or indirectly through an NT2.
- Each device is identified by a sub-address.
- The ISDN allows upto 40 digits for a sub-address.

4 Broadband ISDN:

- When the ISDN was originally designed, data rates of 64 kbps to 1.544 Mbps were sufficient to handle all existing transmission needs.
- As applications using the telecommunications n/ws advanced, these rates proved inadequate to support many applications.
- In addition, the original bandwidths proved too narrow to carry the large numbers of concurrent signals produced by a growing industry of digital service providers.

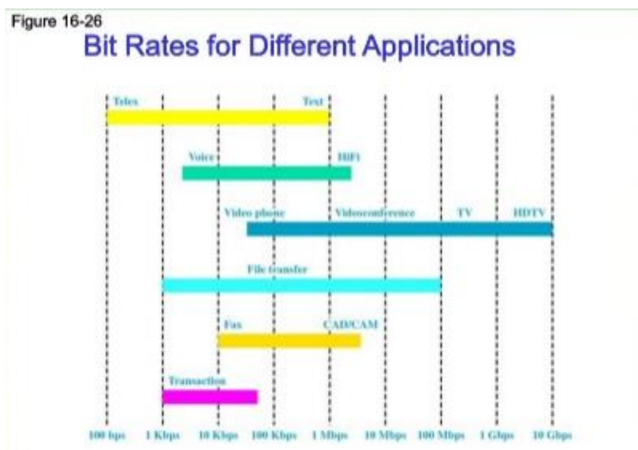


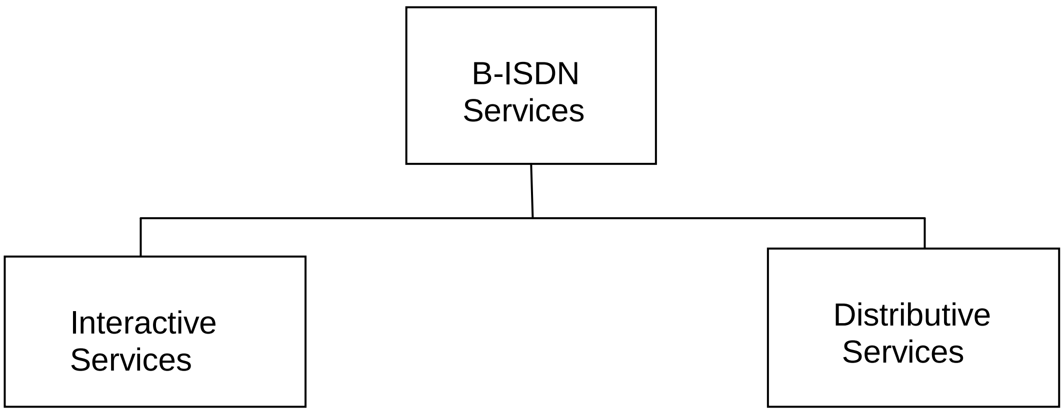
Figure 16-26
Bit Rates for Different Applications

- To provide the needs for next generation technology, an extension of ISDN called Broadband ISDN (B-ISDN) is under study.
- The original ISDN is now known as narrow band ISDN (N-ISDN).
- B-ISDN provides subscribers to the n/w with data rates in the range of 600 mbps, almost 400 times faster than the PRI rate.
- B-ISDN not yet implemented or standardized.

- B-ISDN is based on a change from metal cable to fiber – optic cable at all levels of tele-communications.

4.1 **Services:**

- B-ISDN provides 2 types of services interactive & distributive.



4.1.1 **Interactive Services:**

- Interactive services are those that required two – way exchange between either two subscribers or between a subscriber and a service provider.
- These services are of 3 types : conversational, managing & retrieval.

4.1.1.1 **Conversational:**

- Conversational services are those, such as telephone calls, that support real – time exchanges.
- These real time services can be used for telephony, video telephony, video conferencing, data transfer, and so on.

4.1.1.2 **Messaging:**

- Messaging services are store- and forward exchanges.
- These services are bidirectional, meaning that all parties in an exchange can use them at the same time.
- The actual exchange, may not occur in real time.
- One subscriber asking another for information may have to wait for an answer, even though both parties are available at the same time.

- These services include voice mail, data mail and video mail.

4.1.1.3 Retrieval:

- Retrieval services are those used to retrieve information from a central source, called an information centre.
- These services are like libraries: they must allow public access and allow users to retrieve information on demand.
- eg. A videotext that allows subscribers to select video data from an online library.
- Service is bidirectional because it requires action on the part of both the requester and the provider.

4.1.2 Distributor Services:

- These are unidirectional services sent from a provider to subscribers without the subscriber having to transmit a request each time a service is desired.
- These services can be without or with user control.

4.1.2.1 Without user control:

- These services are broadcast to the user without the user's having requested them or having control over either broadcast times or content. User choice limited to whether or not to receive the service at all. eg. commercial TV – programming content & times devided by provider alone.

4.1.2.2 With user control:

- These services are broadcast tot eh user in a round – robin fashion.
- Services are repeated periodically to allow the user a choice of times during which to receive them.

- Which services are broadcast at which times, is the option of the provide alone.
- With pay TV a program is made available in a limited number of time slots.
- A user wishing to view the program must activate his or her television to receive it, but he or she has no other control.

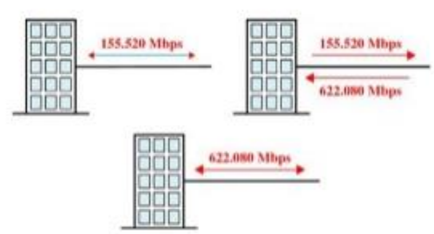
4.2 Physical Specifications:

- B- ISDN model is divided into layers that are different from N- ISDN and closely tied to the design of ATM.
- Physical aspects of B-ISDN not related to ATM include access methods, functional equipment groupings and reference points.

4.3 Access Methods:

Figure 16-28

B-ISDN Accesses



- B-ISDN defines 3 access methods designed to provide for 3 levels of user needs.

4.3.1 155.520 Mbps full duplex:

- This rate matches that of an OC – 3 Sonet link.
- It is high enough to support customers who need access to all N-ISDN services and to one or more regular video transmission services.
- This method is geared to fill the needs of most residential and many business subscribers.

4.3.2 155.520 Mbps output / 622.080 Mbps input:

- The outgoing rate is 155.520 Mbps (same as an OC – 3 Sonet Link), but the incoming rate is 622.080 Mbps (same as an Oc-12 sonet link).
- Designed to fill the needs of business that require the simultaneous receipt of multiple services and video conferencing but that are not service providers & don not broadcast distributive services.
- Input needs of these subscribers are far greater than their o/p needs.
- Providing only one rate would either limit their receipt of services or result in wasted link capacity.
- The asymmetrical configuration provides for a balanced use of resources.

4.3.3 622.080 Mbps full duplex:

- This is designed for business that provide and receive distributive services.

4.4 Functional Grouping:

- The functional groupings of equipment in the B-ISDN model are same as those for N-ISDN.
- Here they are called B-NT1, B-NT2, B-TE1, B-TE2 and B-TA.

4.5 Reference Points:

- B-ISDN uses the same reference points as N-ISDN (R, S, T, & U); some of these, are currently under scrutiny and may be redefined.

4.6 FUTURE OF ISDN:

- The N-ISDN was designed to replace the analog telephone system with a digital one for both voice and data transmission.

- N-ISDN has replaced the normal telephone line in some European countries but in United States this replacement was delayed and new technologies such as cable modem and ADSL evolved that make N-ISDN questionable.
- However ISDN can still be considered a good solution for several reasons:

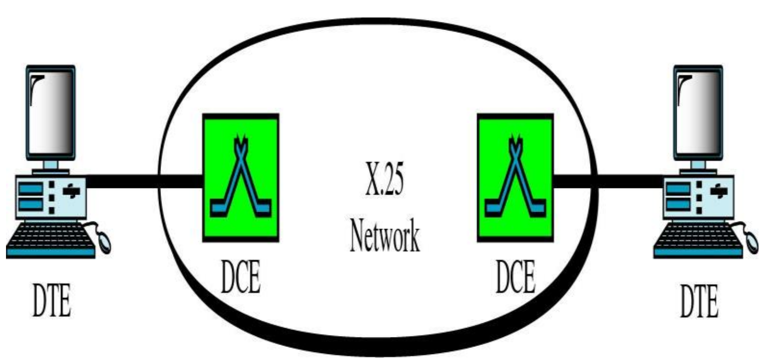
  First SIDN can be brought to a subscriber premise with minimum cost and the services available can satisfy the needs of many users.

  Second, new equipment has appeared on the market that allows a subscriber to use the entire bandwidth of an ISDN link (192 Kbps for BRI or 1.544 Mbps for PRI)

  Third, the protocol is flexible enough to be upgraded to higher data rates using new technology and new transmission media.

  Fourth, N-ISDN can be used as a fore runner for B-ISDN, the data rate of which is sufficient for several years to come.

5 PACKET LAYER PROTOCOL X.25

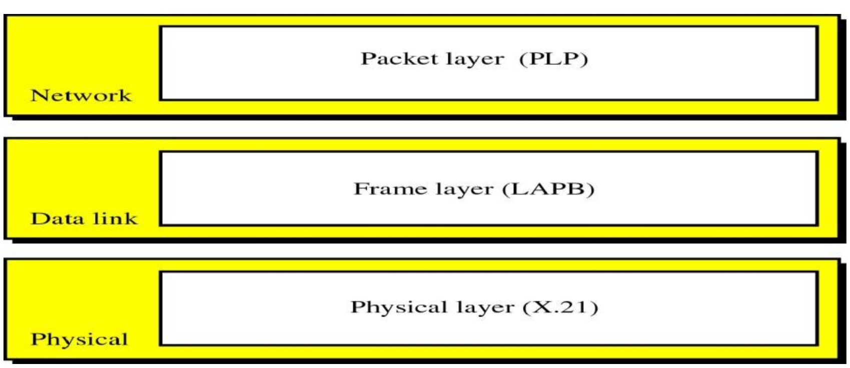


- It is a Packet-Switching wide area network.
- It is an interface between Data TerminalEquipment(DTE) and Data- Circuitterminating Equipment(DCE)
- It describes the procedures for data transmission between a DTE and a DCEfor terminal operation in the packet mode on public data networks.
- It is also known as Subscriber Network Interface(SNI) protocol.
- It uses VC approach to packet switching(SVC and PVC) and uses asynchronous(statistical) TDM to multiplex packet

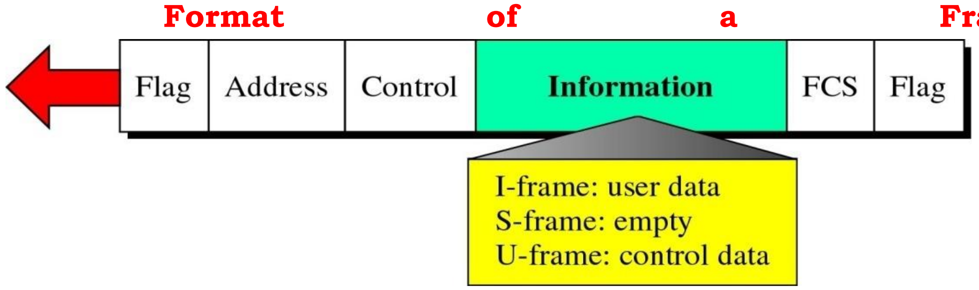
5.1 X.25 Layers

- The three layers are Physical , frame (data link) and packet (network) layer.
- Physical layer specifies X.21 protocol .
- At frame layer X.25 provides data link control using a bit-oriented protocol called Link Access Procedure , Balanced(LAPB).

5.1.1 X.25 Layers in Relation to the OSI Layers

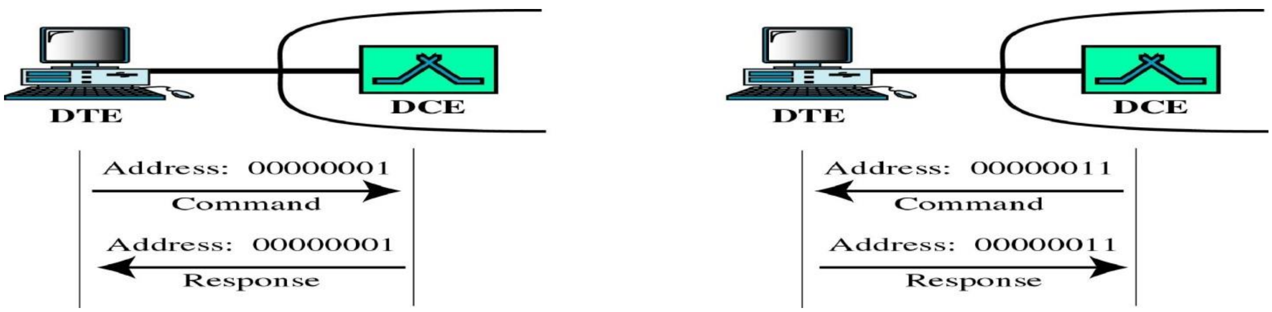


Format of a Frame



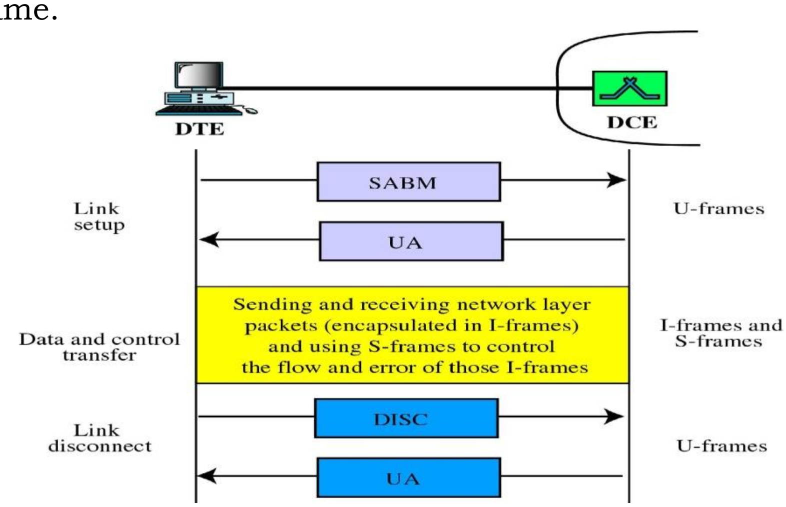
- I – Frames are used o encapsulate PLP packets from the network layer
- S – Frames are for flow and error control in the frame layer.
- U – Frames are used to set up and disconnect the links between a DTE and a DCE

5.1.2 Addressing at the Frame Layer



5.2 Three Phases of the Frame Layer

- **Link Setup** – The link between DTE and DCE must besetup before packets are transferred. Either DTE or DCE can setup the link by sending a SABM(Set Asynchronous Balanced Mode) frame, the responding party sends an UA(Unnumbered Acknowledgement) frame.
- **Data Transfer** – after the link has been established thetwo parties can send and receive network layer packets using I – frames and S – frames.
- **Link Disconnect** – When the network no longer needsthe link, one of the parties issue a Disconnect(DISC) frame and other party replies with an UA frame.

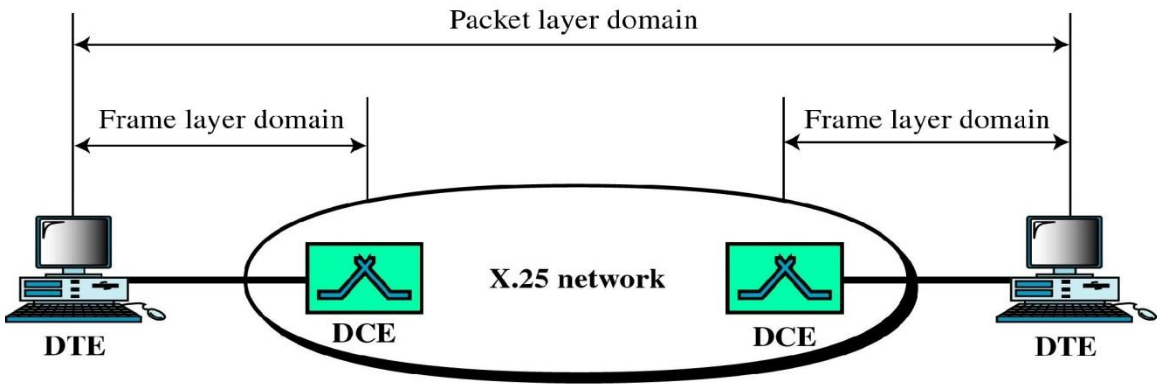


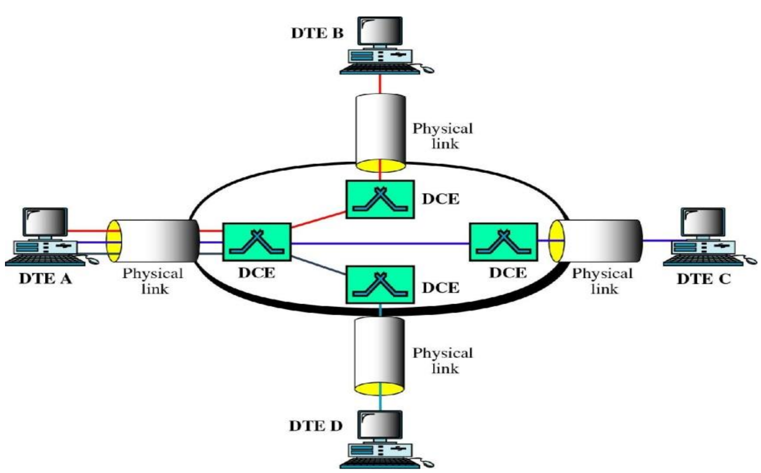
5.2.1 Packet Layer

- Packet Layer handles connection establishment, data transfer, connection termination , Virtual Circuit creation and negotiation of network services between two DTEs.
- Frame layer is responsible for connection between DTE and DCE, whereas packet layer is responsible for making connection between two DTE's.
- X.25 uses flow and error control at both frame and packet layer.

5.2.2 Frame Layer and Packet Layer Domains

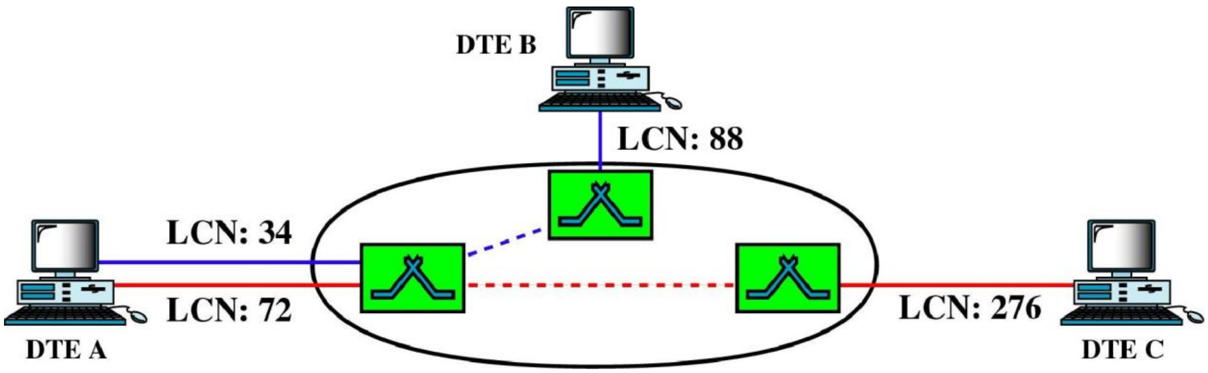
Virtual Circuits in X.25

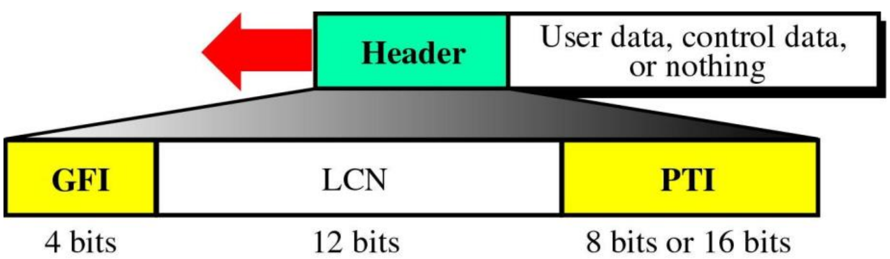




5.3 Virtual Circuit Identifiers

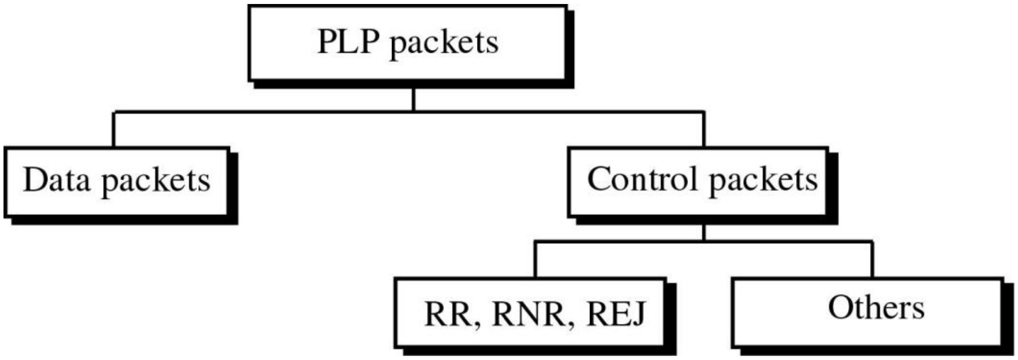
- Virtual circuit identifier in X.25 is called the Logical Channel Number (LCN)
- When VC is established there is always a pair of LCNs.
- One defining the VC between the local DTE and local DCE and the other one betwee DCE and remote DTE.



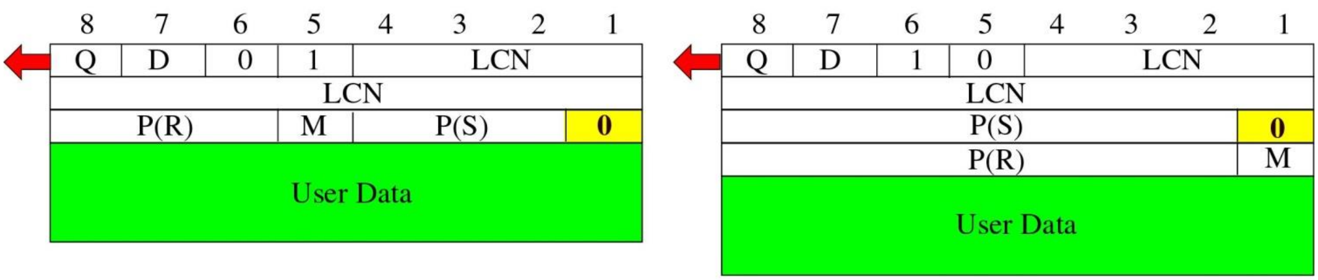


- **General Format identifier(GFI)**
- **Logical Channel Number(LCN)**
- **Packet Type Identifier (PTI)**

5.4 Categories of PLP Packets

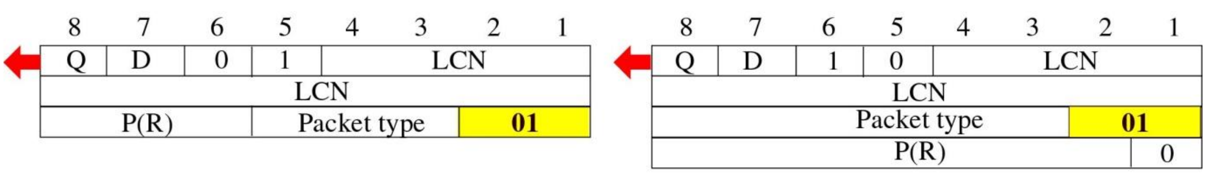


5.4.1 Data Packets in the PLP Layer



a. Three-bit sequence number b. Seven-bit sequence number

RR, RNR, and REJ Packets



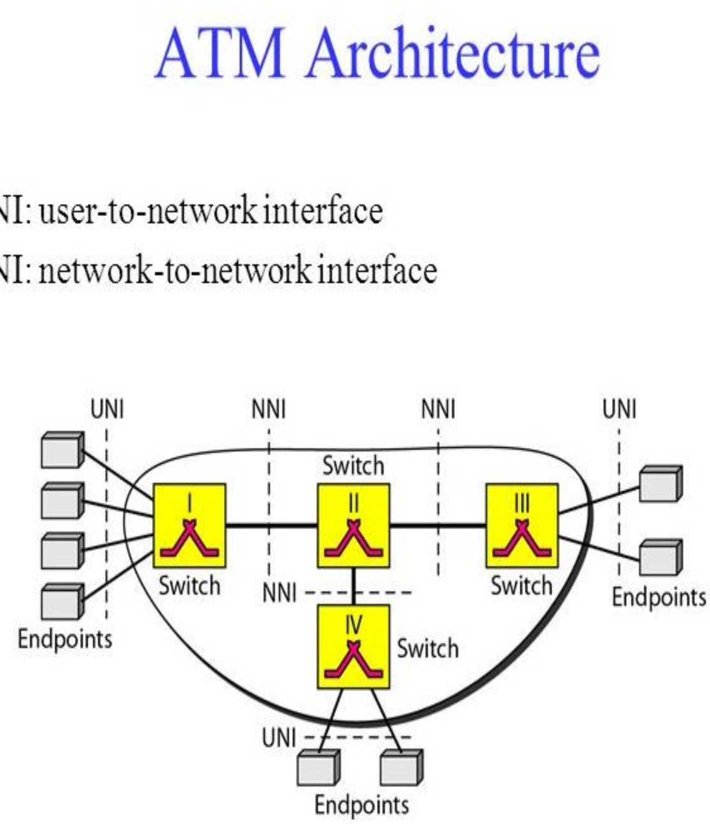
a. Three-bit sequence number b. Seven-bit sequence number

6 ATM

- Asynchronous Transfer mode (ATM) is the cell relay protocol designed by the ATM Forum and adopted by ITU-T.
- The combination of ATM and B-ISDN will allow high speed interconnection of all the world's network.

7 ATM Architecture:

- ATM is a cell – switched n/w.
- The user access devices, called the end points are connected through a user to network interface (UNI) to the switches inside the network.
- The switches are connected through network to network interfere (NNIs).



7.1 Virtual Connection:

- A Transmission path (TP) is the physical connection (wire, cable, satellite, etc.) between an end point and a switch or between two switches.

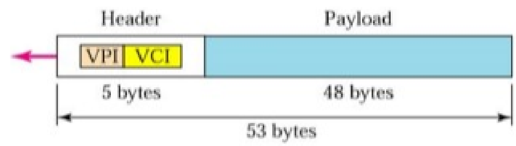
- A transmission path is divided into several virtual paths.
- A virtual path (VP) provides a connection or a set of connection between two switches.
- Cell networks are based on virtual circuits (VCs).
- All cells belonging to a single message follow the same virtual circuit and remain in their original order until they reach their destination.
- In the figure, 8 end points are communicating using 4 VCs.
- The First 2 VCs seems to share the same virtual path from switch I to switch III so it is reasonable to bundle these 2 VCs together to form 1 VP.
- Similarly, other 2 VCs share the same path from switch I to switch IV, so can combine them to form 1 VP.

7.2 Identifiers:

- In a virtual circuit n/w, to route data from one end point to another, the virtual connections need to be identified.
- So ATM uses a hierarchical identifier with 2 levels: a virtual path identifier (VPI) and a virtual circuit Identifier (VCI).
- VPI defines the specific VP and VCI defines the particular VC : inside the VP.
- VPI is the same for all virtual connections that are bundled into one VP.
- A virtual connection is defined by a pair of numbers : VPI, VCI.
- The lengths of the VPIs for UNI & NNI inter/ access are different.

7.3 Cells:

- The basic data unit in an ATM / n/w is called a cell
- A cell is only 53 bytes long with 5 bytes allocated to header and 48 bytes carrying payload.



7.4 Connection Establishment and Release:

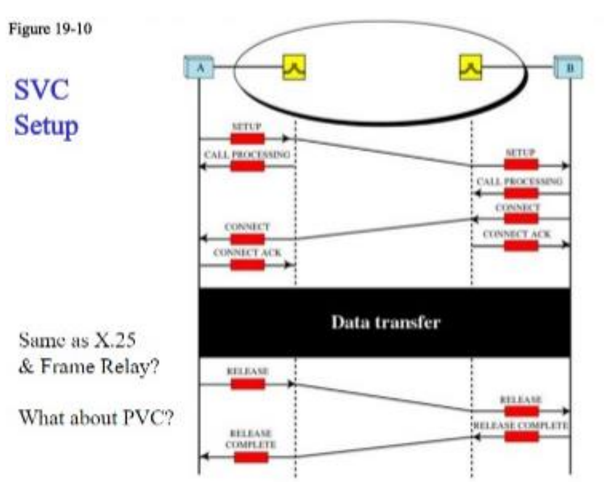
- Like x.25 and frame relay, ATM uses 2 types of connections, PVC & SVC.

7.5 PVC:

- A permanent virtual circuit (PVC) connection is established between 2 end points by the network provider.
- The VPIs and VCIs are defined for the permanent connections & the values are entered for the tables of each switch.

7.6 SVC:

- In a switched virtual circuit (SVC) connection, each time an end point wants to make a connection with another end point, a new virtual circuit should be established.
- ATM cannot do the job by itself, but needs n/w layer addresses and the services of another protocol (such as B-ISDN or IP).
- The actual mechanism depends on the n/w layer protocol.

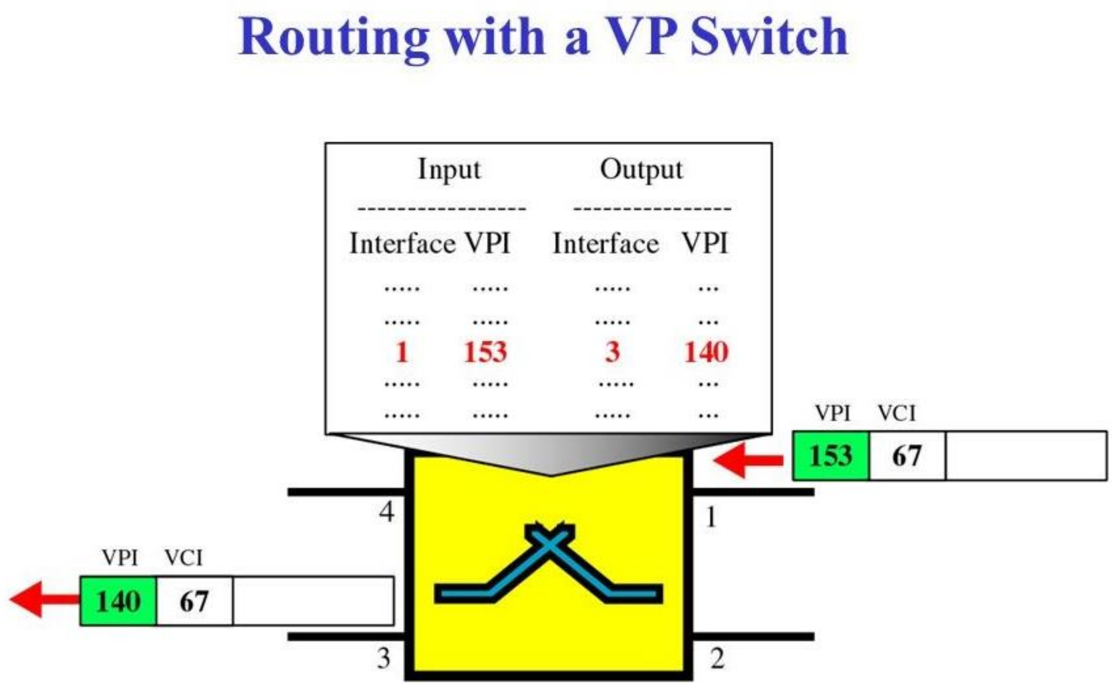


7.7 Switching:

- ATM uses switches to route the cell from a source end point to the destination end point.
- ATM uses 2 type of switches : VP and VPC.

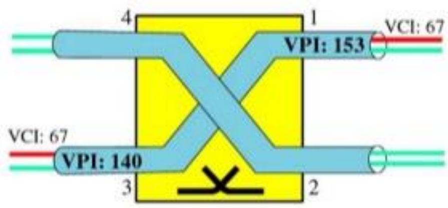
7.8 VP Switch:

- A VP switch routes the cell using only the VPI.



- A cell with a VPI of 153 arrives at switch interface 1.
- The switch checks its switching table, which stores four pieces of information per row: arrival interface number, incoming VPI, corresponding outgoing interface number, the new VPI.
- The switch finds the entry with interface 1 and VPI 153 and discovers that the combination corresponds to output interface 3 with VPI 140.
- It changes the VPI in the header to 140and sends the cell out through interface 3.

A Conceptual View of a VP Switch

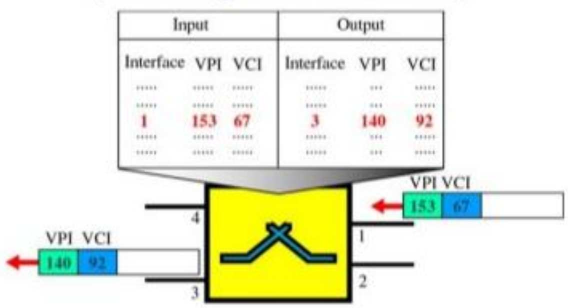


- The VPI and change, but the VCIs will remain the same.

7.9 <u>VPC switch:</u>

- A VPC switch routes the cell using both the VPIs and the VCIs.
- The routing requires the whole identifier.

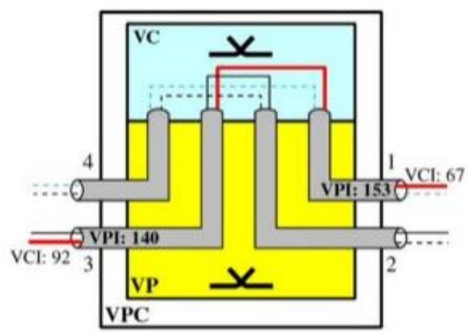
Routing with a VPC Switch
(Combining VP & VC Switches)



- A cell with a VPI of 153 & VCI of 67 arrives at switch interface 1.
- The switch checks its switching table, which stores six pieces of information per row: arrival interface number, incoming VPI, incoming VCI, corresponding outing interface number, the new VPI and the new VCI.
- The switch finds the entry with the interface 1, VPI 153, & VCI 67 & discovers that the combination corresponds to o/p interface 3, VPI 140 & VCI 92.
- It changes the VPI & VCI in the header to 140 & 92, respectively, & sends the cell out through interface 3.

- The whole idea behind dividing a virtual connection identifier into 2 parts is to allow hierarchical routing.
- Host of the switches in a typical ATM n/w are VP switches, they only route using VPs.
- The switches at the boundaries of the n/w, those that interact directly with the end point devices, use both VPIs & VCIs.

A Conceptual View of a VPC Switch



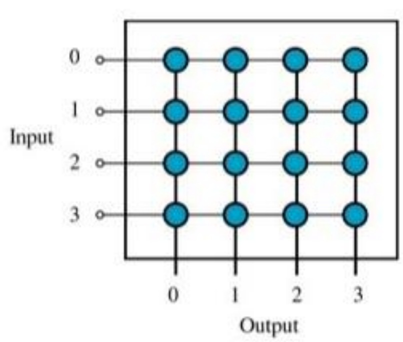
7.10 Switch Fabrics:

- In ATM, there is a need for switches that can receive and route cells as fast as possible.
- In addition, the switches in ATM must be synchronized, although there may be no cells in some slots.
- The switch has a clock & delivers one cell to the output at each tick.

7.11 Crossbar switch:

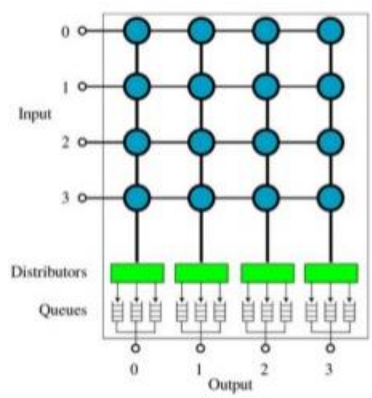
- The simplest type of switch for ATM is the crossbar switch.

Crossbar Switch



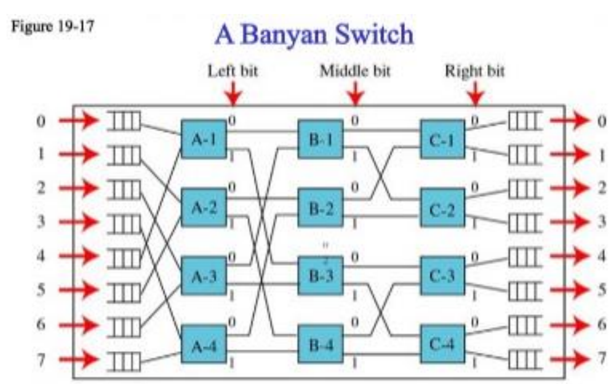
7.12 Knockout Switch:

- The problem with the crossbar switch is the collision that result when 2 cells arriving at different inputs need to go out the same output.
- Knockout switch uses distributors & queues to direct the cells to different queues at the output.
- But still this switch is inefficient – with n i/ps& n o/ps, we still need n^2 cross points.

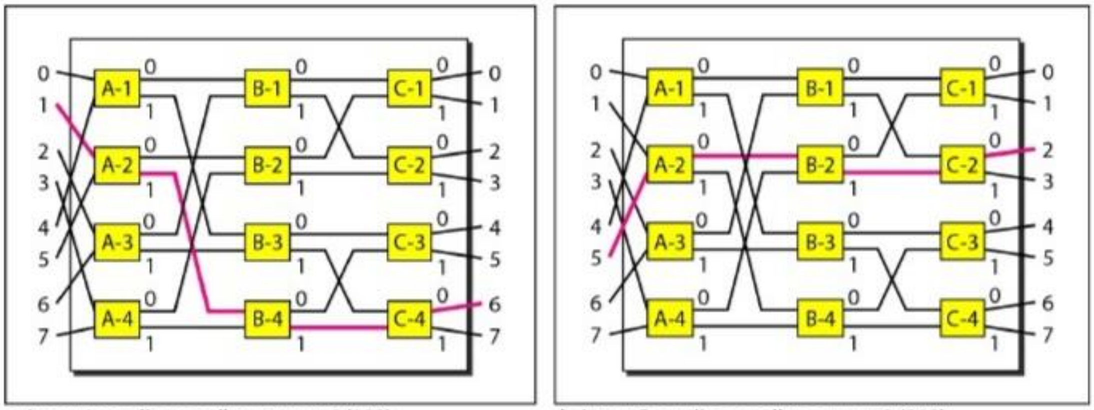


7.13 Banyan Switch:

- A more realistic approach is a switch called a banyan switch.
- A banyan switch is a multistage switch with micro switches at each stage that route the cells based on the o/p port represented as a binary string.
- The 1st stage routes the cell based on the high order bit of the binary string.
- The 2nd Stage routes the cells based on the second high order bit, so on.



- The number of stages shown in figure is $\log_2 (8) = 3$.



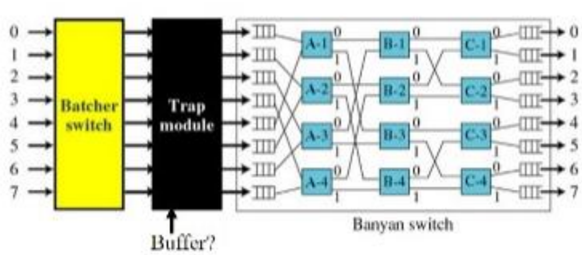
a. Input 1 sending a cell to output 6 (110) b. Input 5 sending a cell to output 2 (010)

- I part(a), a cell has arrived at i/p port 1 and should go to o/p port 6 (no in binary).
- The first microswitch (A-2) routes the cell based on the 1st bit (1), the 2nd micro switch (B-4) routes the cell based on the 2nd bit (1), and the 3rd micro switch (C-4) routes the cell based on the 3rd bit (0).

7.14 Batcher-Banyan Switch:

- Disadvantage of banyan switch is the possibility of internal collision ever when 2 cells are not heading for the same o/p port.
- K.E. Batcher designed a switch that comes before the banyan switch & sorts the incoming cells according to their final destination. The combination is called the Batcher – banyan switch.
- Another hardware module called the trap is added between the Batcher switch and the banyan switch.

Batcher-Banyan Switch

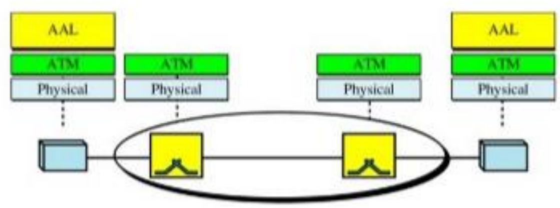


- The trap module prevents duplicate cells (cells with the same o/p destination) from passing to the banyan switch simultaneously.
- Only one cell for each destination is allowed at each tick, if there are more than one, they should wait for the next tick.

8 ATM Layers:

- ATM defines 3 layers - They are, from top to bottom, the application adaptation layer, the ATM layer, and the physical layer.
- The end points we all 3 layers while the switches use only the 2 bottom layers.

ATM Layers in End-Point Devices and Switches



8.1 Application Adaptation Layer (AAL):

- The AAL allows existing n/ws (such as packet n/ws) to connect to ATM facilities.
- AAL protocols accept transmissions from upper –layer services (eg. packet data) & map them into fixed – sized ATM cells.
- These transmissions can be of any type (voice, data, audio, video) and can be of variable or fixed rates.
- At the receiver, this process is reversed – segments are reassembled into their original formats & passed to the receiving service.

8.1.1 Data Types:

- ATM designers, identified 4 types of streams i.e. data streams.

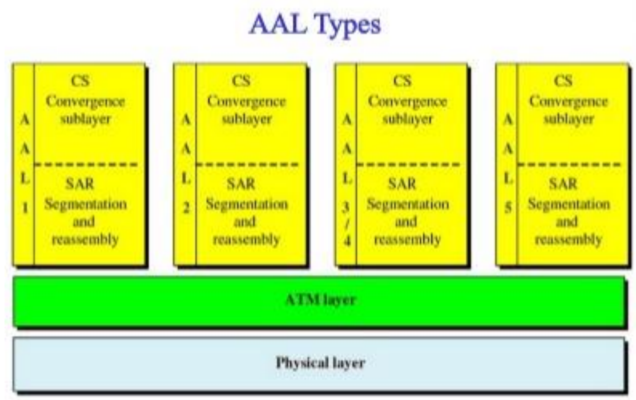
8.1.2 Constant-bit-rate (CBR):

- CBR data refers to applications that generate & consume bits at a constant rate.
- In this type of application, transmission delays must be minirral&transmission must simulate real time. eg. real time voice (telephone calls) & real time video (television).

8.1.3 Variable bit rate (VBR):

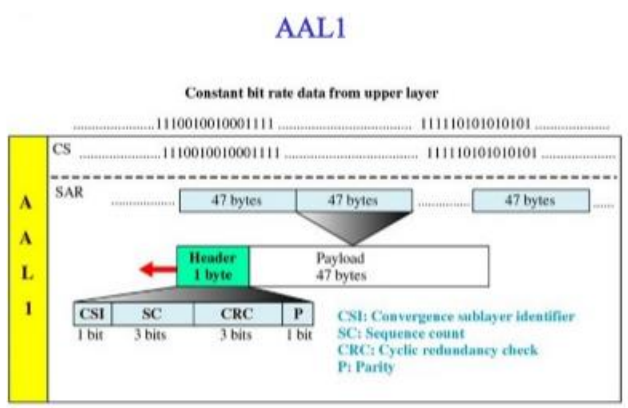
- VBR data refers to applications that generate & consume bits at variable rates.
- In this type of application, the bit rate varies from section to section of the transmission, but within established parameters. eg. compressed voice, data & video.
- Connection oriented packet data refers to conventional packet applications (such as x .25 & TCP protocol of TCP/IP) that use virtual circuits.
- Connectionless packet data refers to applications that use a datagram approach to routing (such as the IP protocol in TCP/ IP).
- The ITU – T recognized the need for an additional category, one that cuts across all of the above data types but is adapted for point to point rather than multipoint or internet work transmissions.
- The sublayer designed to meet the needs of this type of transmission is called the simple & efficient adaptation layer (SEAL).

8.1.4 Convergence & Segmentation:



- Each of the AAL categories is actually 2 layers: the convergence sublayer (CS) and the segmentation & reassembly (SAR) sublayer.

8.2 AAL1:



Note: CSI used for signaling while SC for error & flow control

- Supports applications that transfer information at constant bit rates, such as video. A voice, & allows ATM to connect existing digital telephone n/ws such a DS-3 or E-1.

8.2.1 Convergence Sublayer (CS):

- Divides the bit stream into 47 byte segments & passes them to the SAR sublayer below.

8.2.2 Segmentation and Reassembly (SAR):

- This layer accepts a 47 byte payload from the CS & adds a one byte header.
- The result is a 48 byte data unit that is then passed to the ATM layer, where it is encapsulated in a cell.

The header at this layer consists of 4 fields:

8.2.3 Convergence sublayer identifier (CSI):

- The one bit CSI field will be used for signaling purposes that are not yet clearly defined.

8.2.4 Sequence Count (SC):

- 3 bit SC field is a modulo 8 sequence number to be used for ordering and identifying the cells for end to end error & flow control.

8.2.5 Cyclic redundancy check (CRC):

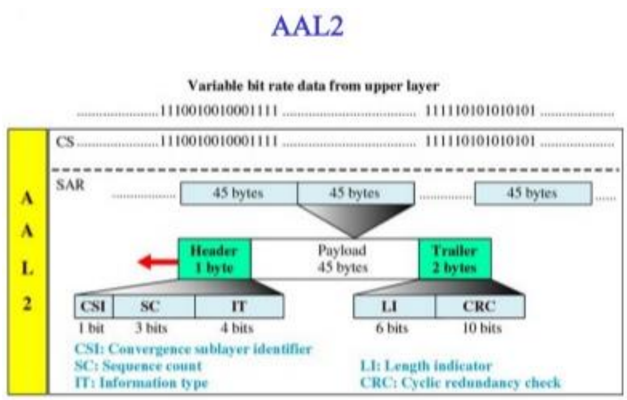
- The 3 bit CRC field is calculated over the first 4 bits using the 4 bit divisor x^3+x+1.
- They are intended not only to detect a single or multiple bit error, but also to correct single bit errors.
- In non real time applications, cell can be retransmitted. In real time applications, retransmission is not an option.
- With no retransmission, the quality of the received data deteriorates.
- Large number of missing cells can destroy intelligibility.
- Automatic correction of single bit header errors dramatically reduces the number of cells that are missing and is ‡ a boon to quality of service.

8.2.6 Parity (P):

- The one bit P field is a standard parity bit calculated over the first 7 bits of the header.

- A parity bit can detect odd number of errors but not even number of errors.
- If one single bit is in error, both CRC&P bit will detect it.
- If there are two bits in error, CRC will detect them & the P bit will not.
- In this case, CRC correction is invalid and the cell is discarded.

8.3 AAL2



Note: IT says where in message & LI points to how much padding

- AAL2 is intended to support variable bit rate applications.

8.3.1 Convergence Sublayer (CS)

- The format for reordering the received bit stream & adding overhead is not defined here.
- Different applications may use different formats.

8.3.2 Segmentation and Reassembly (SAR):

- Functions at this layer accept a 45 byte payload from the CS and add a one byte header and two byte trailer.
- The result is a 48 byte data unit that is then paned to the ATM larger, where it is encapsulated in a cell.
- The overhead at this layer consists of 3 fields in the header & two fields in the trailer.

8.3.3 ConvergeneSublayer identifier (CSI):

- The one bit CSI field will be used for signaling purposes that are not yet clearly defind.

8.3.4 Sequence Count (SC):

- The 3 bit SC field is a modulos sequence number to be used for ordering and identifying the cell for end to end error and flow control.

8.3.5 Information Type (IT):

- The IT bits identify the data segment as falling at the beginning, middle or end of the message.

8.3.6 Length indicator (LI):

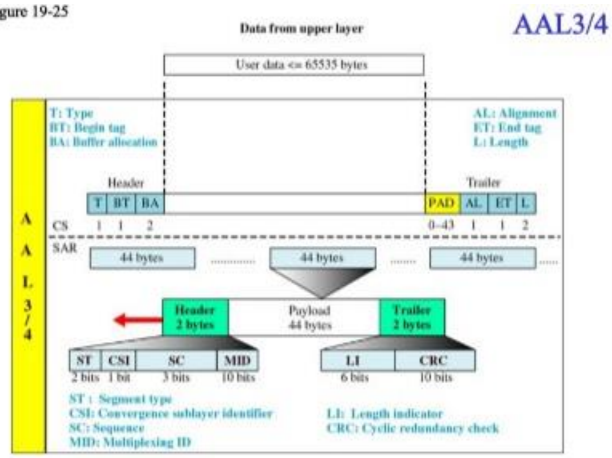
- The first 6 bits of the trailer are used with the final segment of a message (when the IT in the header indicates the end of the message) to indicate how much of the final cell is data and how much is padding.
- If the original bit stream is not evenly divisible by 45, dummy bits are added to the last segment to makeup the difference.
- This field indicates where in the segment those bits start.

8.3.7 CRC:

- The last 10 bits of the trailer are a CRC for the entire data unit.
- This can also be used to correct single bit errors in the data unit.

8.4 AAL ¾ :

Figure 19-25



- Initially, AAL3ws intended to support connection oriented data services and AAL4 to support connectionless services.
- They have been combined now into a single format called AAL ¾ .

8.4.1 Convergence Sublayer (CS):

- This layer accepts a data packet of no more than 65,535 (216-1) bytes from on upper layer service and adds a header and trailer.
- The header and trailer indicate the beginning & end of the message, as well as how much of the final frame is data and how much is padding.
- Because packets vary in length, padding may be required to ensure that segments are of the same size and that the final control fields fall where the receiver expects to find them.
- Once the header, trailer and padding are in place, the CS passes the message in 44 byte segments to the SAR larger.
- Note: CS header and trailer are added to the beginning and end of original packet, not to every segment, the middle segments are pared to the SAR layer without added overhead.
- ATM retains the integrity of the original packets and keeps the ratio of overhead to data bytes low.
- the AAL ¾ CS header & trailer fields are as follows:

8.4.2 Type (T):

- The one byte T field is a holdover from the previous version of AAL3 and is set to 0 in this format.

8.4.3 Begin tag (BT):

- The oen byte BT field serves as a beginning flag.
- It identifies the 1st cell of a segmented packet and provides synchronization for the receiving clock.

8.4.4 Buffer Allocation (BA):

- The two byte BA field tells the receiver what size buffer is needed for the coming data.

8.4.5 Pad (PAD):

- Padding is added when necessary to fill out the final cell(s) in a segmented packet.
- Total padding for a packet can be between 0 x 43 bytes and is added to the last or the last two segments.
- There are 3 possible padding scenarios:
  a) When the number of data bytes in the final segment is exactly 40, no padding is required.
  b) When the number of data bytes in the final segment is less than 40, we add padding bytes (40 to 1) to bring the total to 40.
  c) When the number of data byte available for the final segment is between 41 & 44, we add padding bytes (43 to 40) to bring the total to 84. The 1st 44 bytes make a complete segment & next 40 bytes & trailer make the last segment.

8.4.6 Alignment (AL):

- The one byte AL field is included to make the rest of the trailer four bytes long.

8.4.7 Ending tag (ET):

- The one byte field serves as an ending flag for synchronization.

8.4.8 Length (L):

- The two byte L field indicates the length of the data unit.

8.4.9 Segment and Reasembly:

- Functions at this larger accept a 44 byte payload from the CS and add a 2 byte header and 1 2 byte trailer.

  The header and trailer at this sublayer consist of 6 fields:

8.4.10 Segment type (ST):

- The 2 bit ST identifier cells whether the segment belongs to the beginning, middle at end of a message, or is a single – segment message.

8.4.11 Convergence Sublayer Identifier (CSI):

- The one bit CSI field will be used for signaling purposes that are not yet clearly defined.

8.4.12 Sequence Count:

- The 3 bit SC field is a modulo 8 sequence number to be used for ordering and identifying the cells for end to end error & flow control.

8.4.13 Multiplexing identification (MID):

- The 10 bit MID field identifies cells coming from different data flows and multiplexed on the same virtual connection.

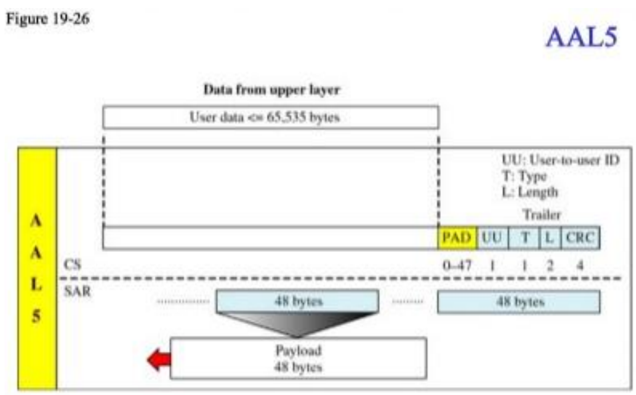
8.4.14 Length indicator (LI):

- The first 6 bits of the trailer are used in conjunction with ST to indicate how much of the last segment is message and how much is padding.

8.4.15 CRC:

- The last 10 bits of the trailer are a CRC for the entire data unit.

8.5 AAL5



- AAL5 assumes that all cells belonging to a single message travel sequentially and that the rest of the functions usually provided by the CS & SAR headers are already included in the upper layers of the sending application.
- Only padding and a 4 field trailer are added at the CS.

8.5.1 Convergence Sublayer:

- This accepts a data packet of no more than 65,535 bytes from an upper layer service and adds an 8 byte trailer as well as any padding required to ensure that the position of the trailer falls where the receiving equipment expects it.
- Then CS panes message in 48 bytes segments to the SAR layer.
- Segments therefore consist of 48 bytes of data or, in the case of last segment, 40 bytes of data and of overhead (trailer).

- Fields added at the end of the message include the following.

8.5.2 Pad (PAD)

- The total padding for a packet can be between 0 & 47 bytes.
- Rules for padding same as AAL 3/4 , with the difference that body segments must equal 48 bytes rather than 44.

8.5.3 User-to-user ID (UU):

- Use of the one byte UU field is left to the discretion of the user.

8.5.4 Type (T):

- The one byte T field is reserved but not yet defined.

8.5.5 Length (L):

- The 2 byte L field indicates how much of the message is data & how much is padding.

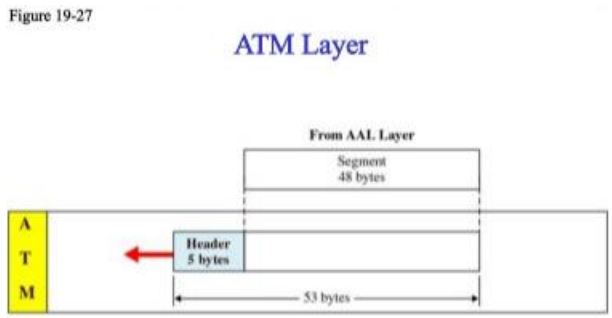
8.5.6 CRC

- The last 4 bytes are an error check for the entire data unit.

8.5.7 Segmentation and Reassembly:

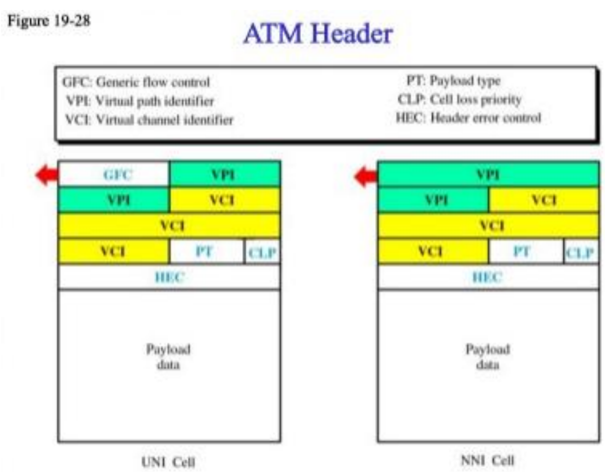
- No header or trailer is defined for the SAR level.
- Instead it passes the message in 48 byte segments directly to the ATM layer.

8.5.7.1 ATM layer:



- The ATM layer provides routing, traffic management, switching and multiplexing services.
- It processes outgoing traffic by accepting 48 byte segments from the AALsublayers and transforming them into 53 byte cells by the addition of a 5-6 byte header.

8.5.7.2 Header Format:



- ATM uses 2 formats for this header, one for user to network interface (UNI) cells & another for network to network interface (NNI) cells.

8.5.7.3 Generic flow control (GFC):

- The 4 bit GFC field provides flow control at the UNI level.
- In the NNI header, these bits are added to the VPI.
- The longer VPI allows more virtual paths to be defined at the NNI level.

8.5.7.4 Virtual path identifier (VPI)

- The VPI is an 8 bit field in a UNI cell and a 12 bit field in an NNI cell.

8.5.7.4 Virtual Channel Identifier (VCI):

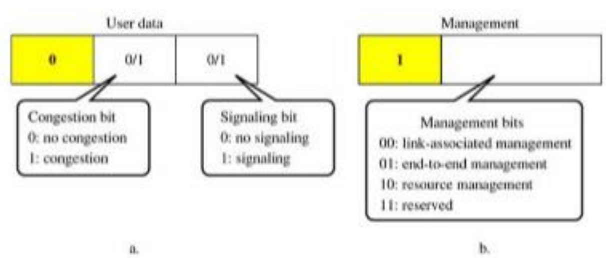
- The VCI is a 16 bit field in both frames.

8.5.7.5 Payload type (PT):

- In the 3 bit PT field, the 1st bit defines the payload as user data or managerial information.

Figure 19-29

Payload Type (PT) Fields



8.5.7.6 Cell loss Priority (CLP)

- The one bit CLP field is provided for congestion control.
- When links become congested, low – priority cells may be discarded to protect the quality of service for higher priority cells.
- This bit indicates to a switch which cells may be dropped and which must be retained.
- A cell with its CLP bit set to 1 must be retained as long as there are cells with a CLP of 0.

8.5.7.7 Header error correction (HEC):

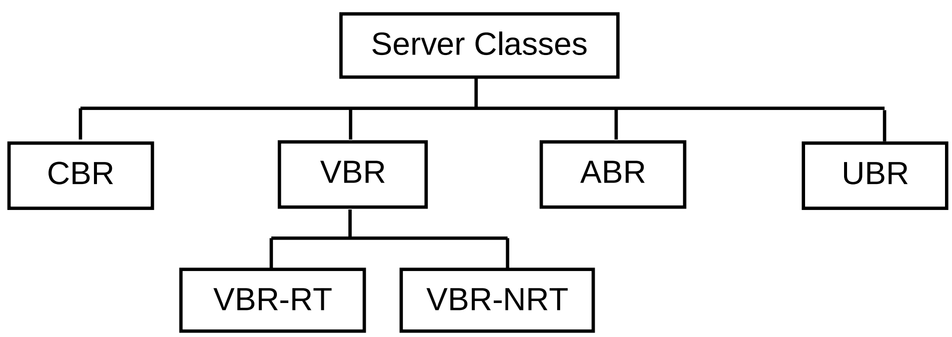
- The HEC is a code computed for the first 4 bytes of the header.
- It is a CRC using the divisor x^8+x^2+x+1 that is used to correct single bit errors and a large class of multiple bit errors.

8.5.7.8 Physical Layer:

- This layer defines the transmission medium, bit transmission, encoding and electrical to optical transformation.
- It provides convergence with physical transport protocols, such as SONET and T-3, as well as the mechanisms for transforming the flow of cells into a flow of bits.

8.6 Service Classes

- The ATM Forum defines 4 service classes.



8.6.1 CBR:

- the constant bit rate (CBR) class is designed for customers that need real time audio or video services; services is similar to T-line.

8.6.2 VBR:

- The variable bit rate (VBR) class is divided into 2 subclasses: Real time (VBR-RT) and Non real time (VBR-NRT).
- VBR – RT is designed for those users that need real time services (voice, video) and use compression techniques to create a variable bit rate.

8.6.3 ABR:

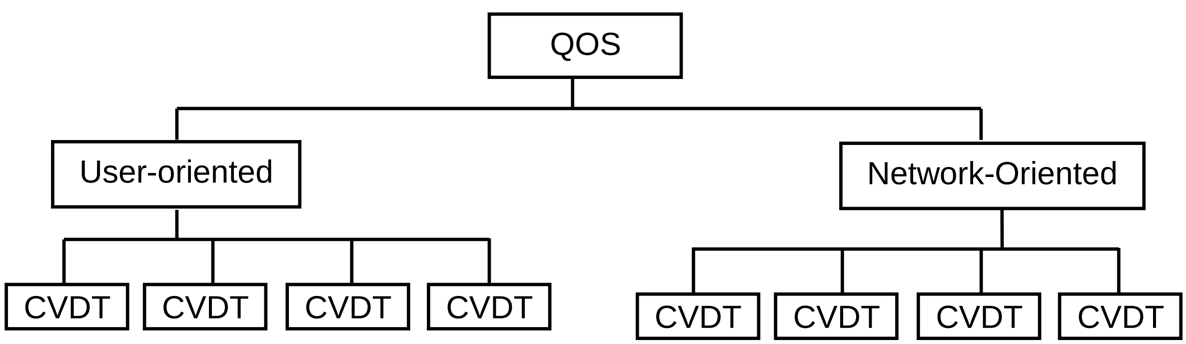
- The available bit rate (ABR) class delivers cells at a minimum rate.
- If more network capacity is available, this minimum rate can be exceeded.

8.6.4 VBR:

- The unspecified bit rate (UBR) class is a best effort delivery service that does not guarantee anything.

8.6.5 Quality of service (QOS):

- QOS defines a set of attributes related to the performance of the connection.
- For each connection, the user can request a particular attribute.
- Each service class is association with a set of the attributes.



8.6.5.1 User-Related Attributes:

- These attributes define how fast the user wants to send data.
- The following are some user – related attributes.

8.6.5.2 SCR (Sustained Cell Rate):

- SCR is the average cell rate over a long time interval.
- The actual cell rate may be lower or higher than this value, but the average should be equal to or less than the SCR.

8.6.5.3 PCR (Peak Cell Rate)

- The PCR defines the senders maximum cell rate.
- The users cell rate can sometimes reach this peak, as long as the SCR is maintained.

8.6.5.4 MCR (Minimum Cell Rate)

- MCR defines the minimum cell rate acceptable to the sender. eg. If MCR is 50,000, the n/w must guarantee that the sender can send atleast 50,000/- cells per second.

8.6.5.5 CVDT (Cell Variation Delay Tolerance):

- CVDT is a measure of the variation in cell transmission times.
  eg. if CVDT is 5ns, this means that the difference between the minimum and maximum delays in delivering the cells should not exceed 5 ns.

8.6.5.6 Network – Related Attributes:

- These attributes are those that define characteristics of the network.
- The following are some network related attributes.

8.6.5.7 CLR (Cell Loss Ratio):

- The CLR defines the fraction of cells lost during transmission or delivered so late that they are considered lost.
  eg. IF sender sends 100 cells and one of them is last, the CLR is, CLR = 1 / 100 = 10^{-2}

8.6.5.8 CTD (Cell Transfer Delay):

- The CTD is the average time needed for a cell to travel from source to destination.

- The maximum CTD & the minimum CTD are also considered attributes.

8.6.5.9 CDV (Cell Delay Variation):

- It is the difference between the CTD maximum and the CTD minimum.

8.6.5.10 CER (Cell Error Ratio):

- It defines the fraction of the cells delivered in error.

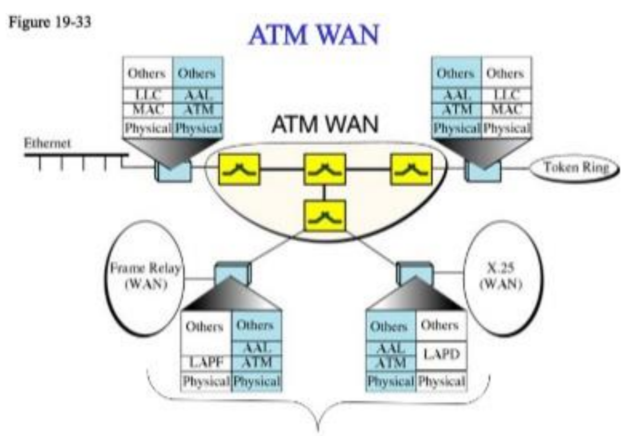
8.6.5.11 Traffic Descriptors:

- The mechanisms by which the service classes and QOs attributes are implemented are called the traffic descriptors.
- A traffic descriptor defines how the system enforces & polices & traffic.
- The algorithm to implement traffic descriptors is called the generalized cell rate algorithm (GCRA). It uses variations of leaky bucket alg. for each type of service class.

8.7 ATM Applications:

- ATM is used in both LANs & WANs.

8.7.1 ATM WANs:



- ATM is basically a WAN technology that delivers cells over a long distance.
- Here, ATM is mainly used to connect LANs or other WANs together.

- A router serves as an end point between ATM n/w & other n/w.
- The router has 2 stacks of protocols : one belonging to the ATM and other belonging to the other protocol.

8.7.2 ATM LANs:

- The high data rate of the technology (155x622 Mbps) used in high speed LANs.
- In the figure, part a shows a switched Ethernet, part b shows an ATM LAN.
- Both use a switch to route packets or cells between computers.
- Similarity is only at surface level so a lot of issues need to be resolved. Some of them are.

8.7.3 Connectionless Versus Connection oriented:

- LANs such as Ethernet are connectionless protocols.
- A station sends data packets to another station whenever the packets are ready. There is no connection establishment or connection termination phase.
- ATM is a connection – oriented protocol.
- A station that needs to send cells to another station should first establish a connection and after all of the cells are sent, terminate the connection.

8.7.4 Physical address verses virtual connection identifiers:

- A connectionless protocol (Ethernet) defines the route of a packet through source and destination addresses.
- A connection oriented protocol (ATM) defines the route of a cell through virtual connection identifiers (VPIs & VCIs).

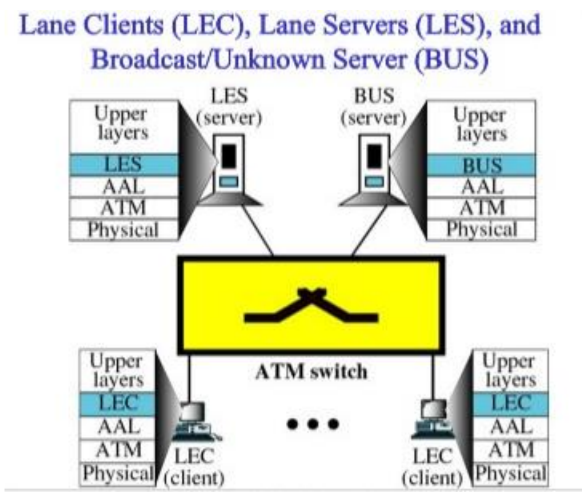
8.7.5 Multicasting and Broadcasting delivery:

- LANs such as Ethernet can both multicast and broadcast packets : a station can send packets to a group of stations or to all stations.
- There is no easy way to multicast or broadcast on an ATM n/w although point to multipoint connections are available.

8.8 LANE:

- Local Area Network Emulation (LANE) enables an ATM switch to behave like a LAN switch : it provides connectionless service, lets the station to use their traditional address instead of VPI /VCI & allows broadcast delivery.
- It is based on client / server approach, all stations use LANE client (LEC) s/w & 2 servers use tow different LANE server s/w called LES & BUS.
- LEC s/w is installed on each station on top of the 3 ATM protocols.
- The upper – layer protocols are unaware of the existence of the ATM technology.
- These protocols send their requests to LEC for a LAN service such as connectionless delivery using MAC unicast, multicast, or broadcast address.
- The LEC, however, just interprets the request and uses the services of either LES or BUS to do the job.
- The LANE server (LES) s/w is installed on the LES server.
- When a station receives a frame to be sent to another station using a physical address, LEC sends a special frame to the LES server.
- The server creates a virtual circuit between the source and destination station.
- Source station uses the virtual circuit to send the frames to the destination.
- Multicasting and broadcasting require the use of another server called the broadcast / unknown server or BUS.

- If a station needs to send a frame, the frame first goes to the BUS server, this server has permanent virtual connections to every station.
- Server creates copies of the received frame & sends a copy to a group of stations or to all stations, simultating a multicasting or broadcasting process.
- The server can also deliver a unicast frame by sending the frame to every station.
- In this case the destination address is unknown.
- This is sometimes more efficient then getting the connection identifier form the LES server.



9 Congestion-Control

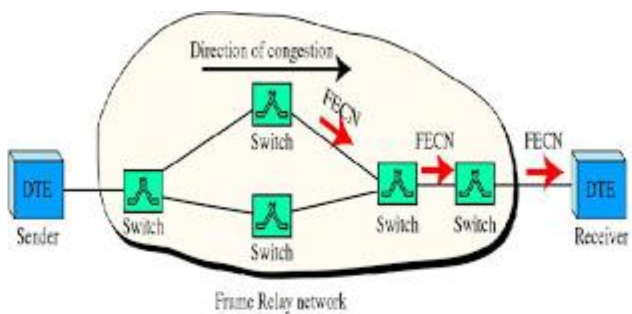
Congestion is an important issue that can arise in packet switched network. Congestion is a situation in Communication Networks in which too many packets are present in a part of the subnet, performance degrades. Congestion in a network may occur when the load on the network *(i.e.* the number of packets sent to the network) is greater than the capacity of the network *(i.e.* the number of packets a network can handle.)

9.1 Congestion-Control Mechanisms

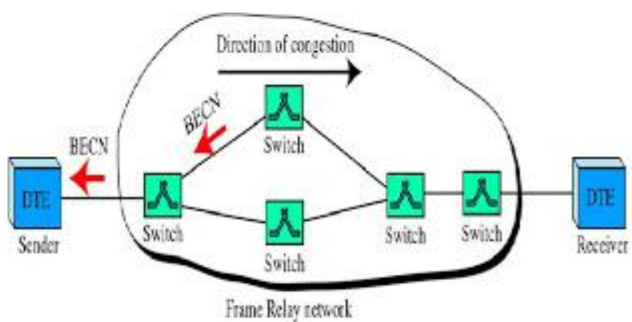
Frame Relay reduces network overhead by implementing simple congestion-notification mechanisms rather than explicit, per-virtual-circuit flow control.Frame Relay typically is implemented on reliable network media, so data integrity is not sacrificed because flow control can be left to higher-layer protocols. Frame Relay implements two congestion-notification mechanisms:

- Forward-explicit congestion notification (FECN)
- Backward-explicit congestion notification (BECN)

FECN and BECN each is controlled by a single bit contained in the Frame Relay frame header. The Frame Relay frame header also contains a Discard Eligibility (DE) bit, which is used to identify less important traffic that can be dropped during periods of congestion.



- The FECN *bit*is part of the Address field in the Frame Relay frame header.TheFECN mechanism is initiated when a DTE device sends Frame Relay frames into the network. If the network is congested, DCE devices (switches) set the value of the frames' FECN bit to 1.



- The BECN bitis part of the Address field in the Frame Relay frame header.DCE devices set the value of the BECN bit to 1 in frames traveling in the opposite direction of frames with their FECN bit set.This informs the receiving DTE device that a particular path through the network is congested.

Frame Relay Discard Eligibility (DE):

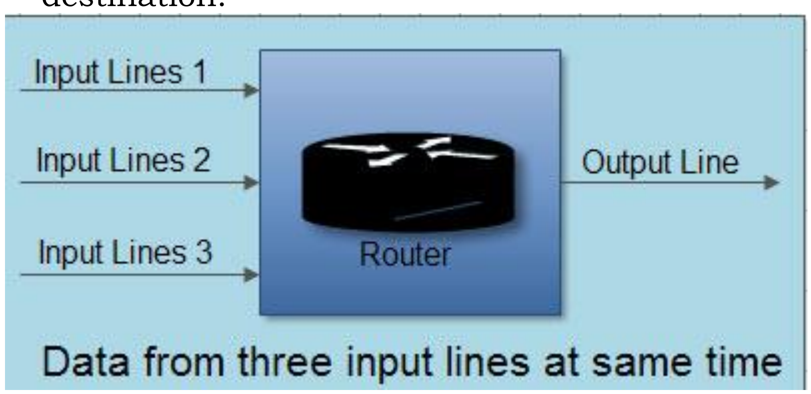
The Discard Eligibility (DE) *bit* is used to indicate that a frame has lower importance than other frames. The DE bit is part of the Address field in the Frame Relay frame header. DTE devices can set the value of the DE bit of a frame to 1 to indicate that the frame has lower importance than other frames.

9.2 Frame Relay error-checking

Frame Relay uses a common error-checking mechanism known as the cyclic redundancy check (CRC).The CRC compares two calculated values to determine whether errors occurred during the transmission from source to destination.Frame Relay reduces network overhead by implementing error checking rather than error correction.

9.3 The various causes of congestion in a subnet are

1. 1.The input traffic rate exceeds the capacity of the output lines. If suddenly, a stream of packet start arriving on three or four input lines and all need the same output line. In this case, a queue will be built up. If there is insufficient memory to hold all the packets, the packet will be lost. Increasing the memory to unlimited size does not solve the problem. This is because, by the time packets reach front of the queue, they have already timed out (as they waited the queue). When timer goes off source transmits duplicate packet that are also added to the queue. Thus same packets are added again and again, increasing the load all the way to the destination.



2.The routers are too slow to perform bookkeeping tasks (queuing buffers, updating tables, etc.).

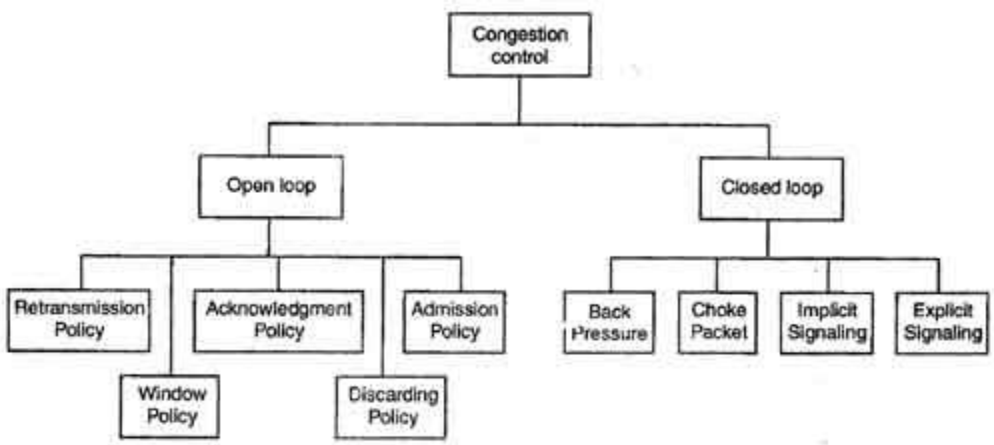
3.The routers' buffer is too limited.

4.Congestion in a subnet can occur if the processors are slow. Slow speed CPU at routers will perform the routine tasks such as queuing buffers, updating table etc slowly. As a result of this, queues are built up even though there is excess line capacity.

5.Congestion is also caused by slow links. This problem will be solved when high speed links are used. But it is not always the case. Sometimes increase in link bandwidth can further deteriorate the congestion problem as higher speed links may make the network more unbalanced.Congestion can make itself worse. If a route!" does not have free buffers, it start ignoring/discarding the newly arriving packets. When these packets are discarded, the sender may retransmit them after the timer goes off. Such packets are transmitted by the sender again and again until the source gets the acknowledgement of these packets. Therefore multiple transmissions of packets will force the congestion to take place at the sending end.

9.4 How to correct the Congestion Problem:

Congestion Control refers to techniques and mechanisms that can either prevent congestion, before it happens, or remove congestion, after it has happened. Congestion control mechanisms are divided into two categories, one category prevents the congestion from happening and the other category removes congestion after it has taken place.



Types of Congestion Control Methods

These two categories are:

1. Open loop
2. Closed loop

9.4.1 Open Loop Congestion Control

• In this method, policies are used to prevent the congestion before it happens.

• Congestion control is handled either by the source or by the destination.

• The various methods used for open loop congestion control are:

9.4.1.1 Retransmission Policy

• The sender retransmits a packet, if it feels that the packet it has sent is lost or corrupted.

• However retransmission in general may increase the congestion in the network. But we need to implement good retransmission policy to prevent congestion.

• The retransmission policy and the retransmission timers need to be designed to optimize efficiency and at the same time prevent the congestion.

9.4.1.2 Window Policy

• To implement window policy, selective reject window method is used for congestion control.

• Selective Reject method is preferred over Go-back-n window as in Go-back-n method, when timer for a packet times out, several packets are resent, although some may have arrived safely at the receiver. Thus, this duplication may make congestion worse.

• Selective reject method sends only the specific lost or damaged packets.

9.4.1.3 Acknowledgement Policy

• The acknowledgement policy imposed by the receiver may also affect congestion.

• If the receiver does not acknowledge every packet it receives it may slow down the sender and help prevent congestion.

• Acknowledgments also add to the traffic load on the network. Thus, by sending fewer acknowledgements we can reduce load on the network.

• To implement it, several approaches can be used:

1. A receiver may send an acknowledgement only if it has a packet to be sent.
2. A receiver may send an acknowledgement when a timer expires.
3. A receiver may also decide to acknowledge only N packets at a time.

9.4.1.4 Discarding Policy

• A router may discard less sensitive packets when congestion is likely to happen.

• Such a discarding policy may prevent congestion and at the same time may not harm the integrity of the transmission.

9.4.1.5 Admission Policy

• An admission policy, which is a quality-of-service mechanism, can also prevent congestion in virtual circuit networks.

• Switches in a flow first check the resource requirement of a flow before admitting it to the network.

• A router can deny establishing a virtual circuit connection if there is congestion in the "network or if there is a possibility of future congestion.

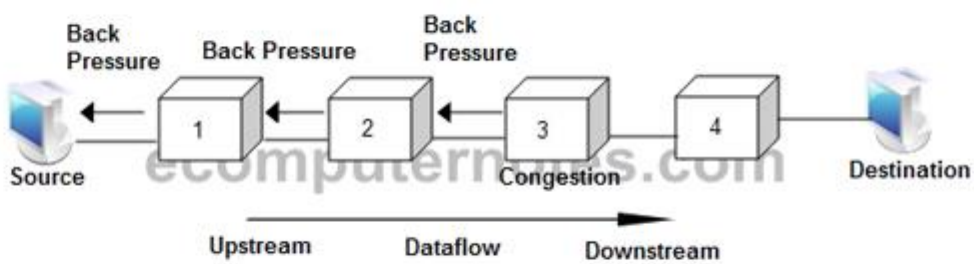
9.4.2 Closed Loop Congestion Control

• Closed loop congestion control mechanisms try to remove the congestion after it happens.

• The various methods used for closed loop congestion control are:

9.4.2.1 Backpressure

• Backpressure is a node-to-node congestion control that starts with a node and propagates, in the opposite direction of data flow.



Backpressure Method

• The backpressure technique can be applied only to virtual circuit networks. In such virtual circuit each node knows the upstream node from which a data flow is coming.

• In this method of congestion control, the congested node stops receiving data from the immediate upstream node or nodes.

• This may cause the upstream node on nodes to become congested, and they, in turn, reject data from their upstream node or nodes.

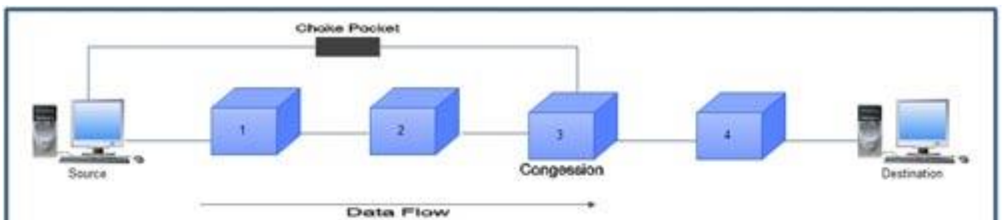
• As shown in fig node 3 is congested and it stops receiving packets and informs its upstream node 2 to slow down. Node 2 in turns may be congested and informs node 1 to slow down. Now node 1 may create congestion and informs the source node to slow down. In this way the congestion is alleviated. Thus, the pressure on node 3 is moved backward to the source to remove the congestion.

9.4.2.2. Choke Packet

• In this method of congestion control, congested router or node sends a special type of packet called choke packet to the source to inform it about the congestion.

• Here, congested node does not inform its upstream node about the congestion as in backpressure method.

• In choke packet method, congested node sends a warning directly to the source station *i.e.* the intermediate nodes through which the packet has traveled are not warned.



Choke Packet method

9.4.2.3. Implicit Signaling

• In implicit signaling, there is no communication between the congested node or nodes and the source.

• The source guesses that there is congestion somewhere in the network when it does not receive any acknowledgment. Therefore the delay in receiving an acknowledgment is interpreted as congestion in the network.

• On sensing this congestion, the source slows down.

• This type of congestion control policy is used by TCP.

9.4.2.4. Explicit Signaling

• In this method, the congested nodes explicitly send a signal to the source or destination to inform about the congestion.

• Explicit signaling is different from the choke packet method. In choke packed method, a separate packet is used for this purpose whereas in explicit signaling method, the signal is included in the packets that carry data .

• Explicit signaling can occur in either the forward direction or the backward direction .

• In backward signaling, a bit is set in a packet moving in the direction opposite to the congestion. This bit warns the source about the congestion and informs the source to slow down.

• In forward signaling, a bit is set in a packet moving in the direction of congestion. This bit warns the destination about the congestion. The receiver in this case uses policies such as slowing down the acknowledgements to remove the congestion.

10. Congestion control algorithms

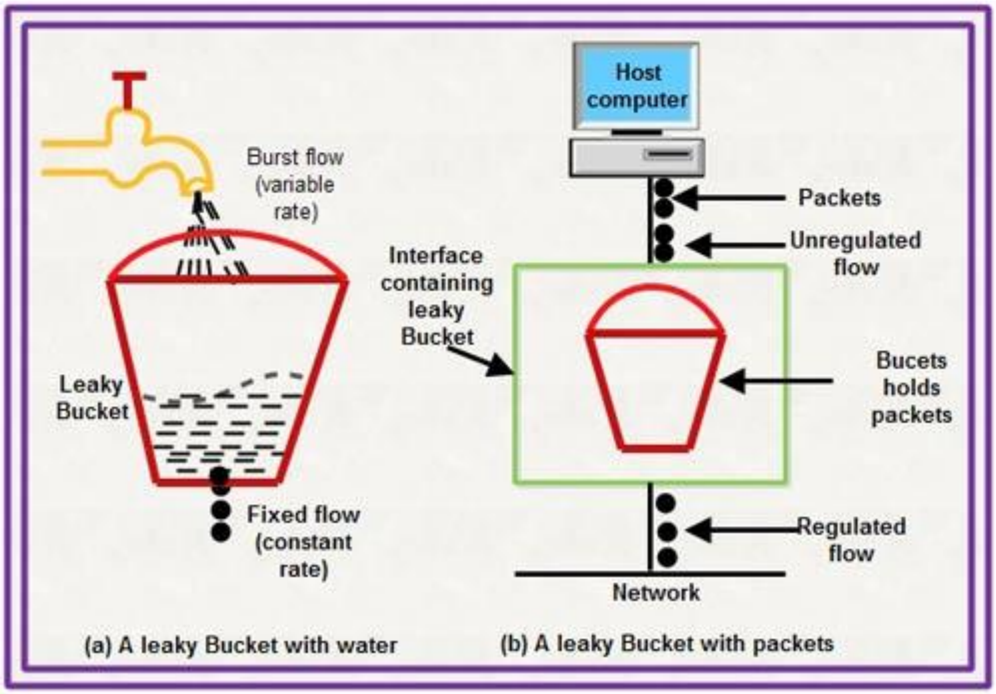
10.1. Leaky Bucket Algorithm

• It is a traffic shaping mechanism that controls the amount and the rate of the traffic sent to the network.

• A leaky bucket algorithm shapes bursty traffic into fixed rate traffic by averaging the data rate.

• Imagine a bucket with a small hole at the bottom.

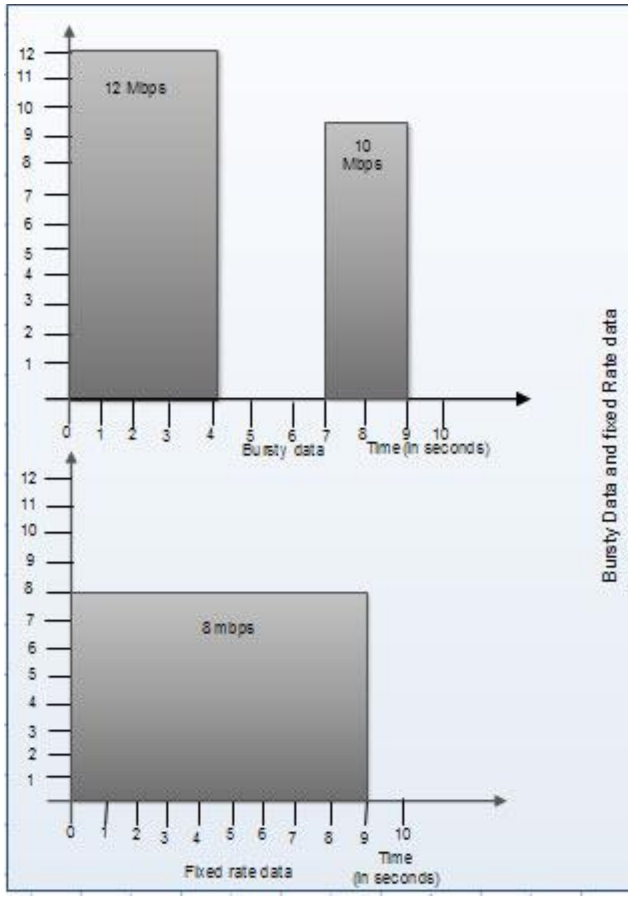
• The rate at which the water is poured into the bucket is not fixed and can vary but it leaks from the bucket at a constant rate. Thus (as long as water is present in bucket), the rate at which the water leaks does not depend on the rate at which the water is input to the bucket.



• Also, when the bucket is full, any additional water that enters into the bucket spills over the sides and is lost.

• The same concept can be applied to packets in the network. Consider that data is coming from the source at variable speeds. Suppose that a source sends data at 12 Mbps for 4 seconds. Then there is no data for 3 seconds. The source again transmits data at a rate of 10 Mbps for 2 seconds. Thus, in a time span of 9 seconds, 68 Mb data has been transmitted.

If a leaky bucket algorithm is used, the data flow will be 8 Mbps for 9 seconds. Thus constant flow is maintained.



2. Token bucket Algorithm

• The leaky bucket algorithm allows only an average (constant) rate of data flow. Its major problem is that it cannot deal with bursty data.

• A leaky bucket algorithm does not consider the idle time of the host. For example, if the host was idle for 10 seconds and now it is willing to sent data at a very high speed for another 10 seconds, the total data transmission will be divided into 20 seconds and average data rate will be maintained. The host is having no advantage of sitting idle for 10 seconds.

• To overcome this problem, a token bucket algorithm is used. A token bucket algorithm allows bursty data transfers.

• A token bucket algorithm is a modification of leaky bucket in which leaky bucket contains tokens.

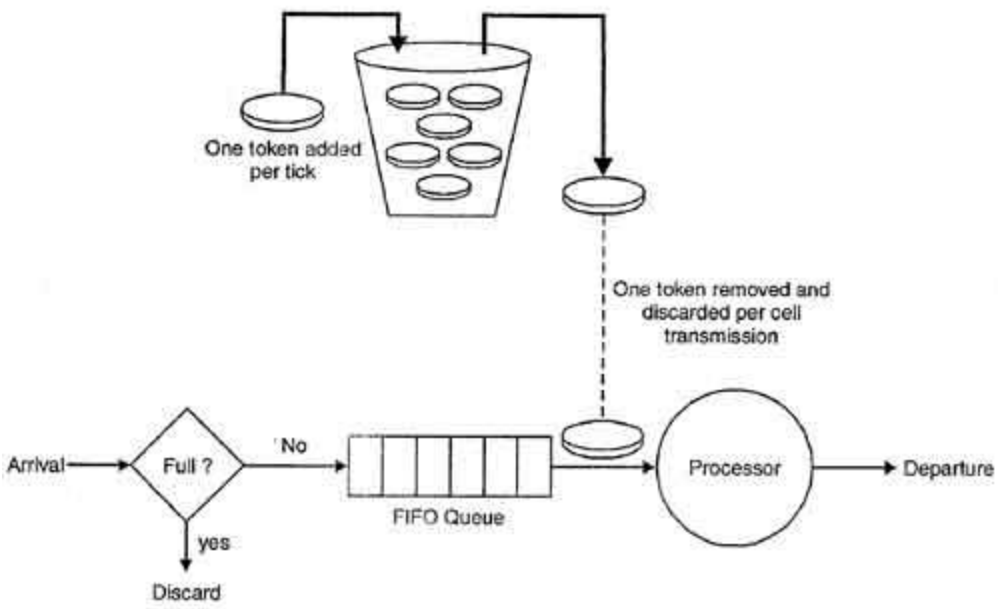
• In this algorithm, a token(s) are generated at every clock tick. For a packet to be transmitted, system must remove token(s) from the bucket.

• Thus, a token bucket algorithm allows idle hosts to accumulate credit for the future in form of tokens.

• For example, if a system generates 100 tokens in one clock tick and the host is idle for 100 ticks. The bucket will contain 10,000 tokens.

Now, if the host wants to send bursty data, it can consume all 10,000 tokens at once for sending 10,000 cells or bytes.

Thus a host can send bursty data as long as bucket is not empty.



Token bucket algorithm