Slide supporting material

Lesson 2: X.25, ISDN, Frame Relay, and TDM Hierarchy, SDH Transport

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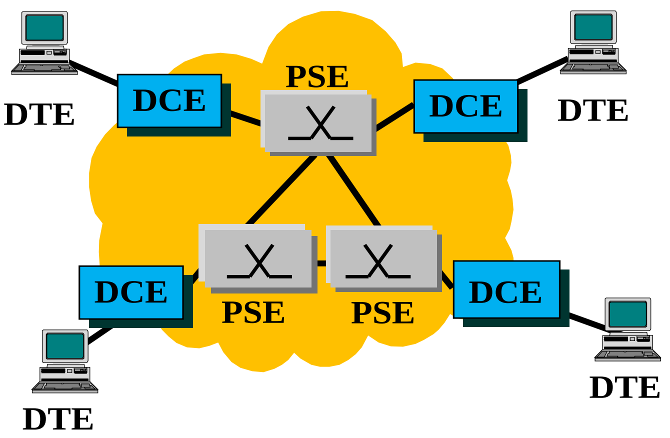
A First Historical Example of Geographical Networks: X.25 Networks

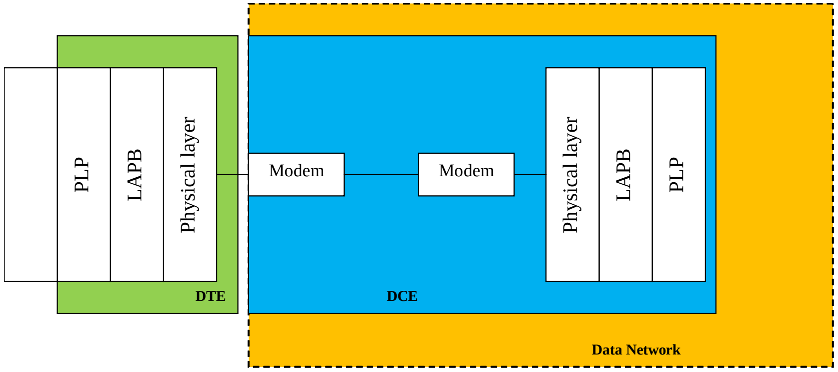
X-25 Networks

- X.25 ITU-T Recommendation was defined in 1976 and based on the **OSI protocol stack**.
- Interface for synchronous transmissions between the user terminal (Data Terminal Equipment, DTE) and the first network equipment (Data Circuit-terminating Equipment, DCE).
- No details are given for the protocols adopted in the network interconnecting DCEs.
  - X.75 protocol of ITU-T (specifying the protocols for communication between two packet-switched data networks) is a common choice inside the network.
- X.25 was conceived for networks whose **physical layer is prone to errors**.
- Typical applications include: automatic teller machine networks and credit card verification networks.

X-25 Networks (cont’d)

- X.25 is a **connection-oriented protocol** that defines the first three layers of the OSI architecture.
  1. **Physical layer**: it is based on the X.21 protocol that is similar to the serial transmissions of the RS-232 standard (ITU V.24).
  2. **Data link layer**: it employs the Link Access Protocol – Balanced (LAP-B), a subset of the HDLC (High Level Data Link Control) protocol in its balanced version (meaning that both parts can start a new transmission without needing the authorization of the other part). An **ARQ** scheme is adopted integrating **flow control** (use of a window scheme).
  3. **Network layer**: the Packet Layer Procedure (PLP) is adopted. PLP is a **connection-oriented** protocol needing an e2e call setup phase. The PLP layer communicates between DTE devices in units called packets.





X-25 Networks (cont'd)

- **X-25 adopts error control and flow control on a hop-by-hop basis.**
  - **Flow control** is needed to avoid overwhelming the receiver with data.
  - **Error control** is used to verify whether the received data is correct so that a retransmission can be requested in case of errors.
- Due to error and flow controls operated on each link (hop), we can understand that X.25 entails a **heavy overhead** (= not efficient).
- X.25 technology was implemented at the **beginning of '80** with very low bit-rate and poor-quality links:
  - The transmission capacity for a DTE typically ranges from 75 kbit/s to 192 kbit/s, up to 2 Mbit/s.
- In Italy, the X.25 network was named **ITAPAC** and operational from 1980s to 1990s.

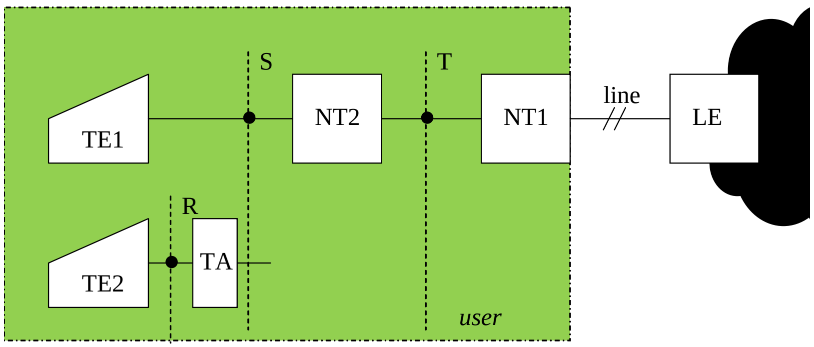
X-25 Networks (cont'd)

- **In-band signalling**: In X.25 information and control messages share the same protocol layers according to the classical OSI approach.
- LAPB is a bit-oriented protocol that ensures that frames are correctly ordered and error-free.
  - Each frame that is sent over a particular link is saved in a buffer until its information has been checked and the frame has been approved by the receiving node or subscriber.
  - LAPB employs an **ARQ scheme** to recover the erroneous frames on each link (in the LAPB frame there are two bytes used for error detection: Frame Check Sequence field). Both the Go-Back-N and the Selective Repeat schemes can be adopted to manage retransmissions. **A sliding window scheme** is integrated with the ARQ scheme to operate flow control.
  - The **store and forward method** is also applied to internal nodes of the network to recover errors.
- X.75 is a signaling system to connect packet-switched network elements (such as X.25) on international circuits. It permits the transfer of call control and network control information and user traffic. On layers 2 and 3, X.75 is almost identical to X.25.

ISDN: Digital Baseband Access, Digital Network

ISDN Introduction

- Integrated Services Digital Network (ISDN) has been standardized by ITU-T in 1980s with Recommendations of families E, I, and Q.
- **Digital access where bits are directly transmitted as baseband pulses.** This allows a unified system to support voice and different types of data traffic flows.
- ISDN supports both circuit-switching and packet-switching.
- There are some important reference points between the different blocks in the access architecture in the picture below, that is R, S, and T points.
- At user premises the **twisted pair** arrives at a Network Termination 1 (NT1). The Terminal Equipment (TE) uses a Network Termination 2 (NT2) to connect to NT1.



ISDN Introduction (cont'd)

- NT1 operates at OSI layer 1 (termination of the transmission line, clock management, channel multiplexing).
- NT2 contains the functionalities of layers 1, 2 and 3 (NT2 can be an ISDN Private automatic Branch eXchange, PBX).
- TE contains all seven layers of the OSI protocol stack.
- In Europe and Japan, the Operators own the NT1 and provide the S/T interface to customers. In North America, the U interface (i.e., the interface between NT1 and LE) is provided to customers, who own the NT1.
- The internal nodes of an ISDN network are called 'switches'.
- The GSM core network was based on ISDN.

ISDN Channels

- **Channel B at 64 kbit/s**
  - It transparently transports the flux of bits from one end to another in the network according to circuit-switching. Hence, the flux of channel B is transparently managed by the network (i.e., only the physical layer needs to managed for channel B in the switches of the network).
- **Channel D (at 16 or 64 kbit/s)**
  - This channel is packet-switched (non-transparent). Hence, at each node of the network, all the first three OSI layers (i.e., 1, 2 and 3) are needed to manage the flux coming from a D channel. Such channel is used both to send signalling messages between the user and the network and to transmit user packet data.

ISDN Access Structures

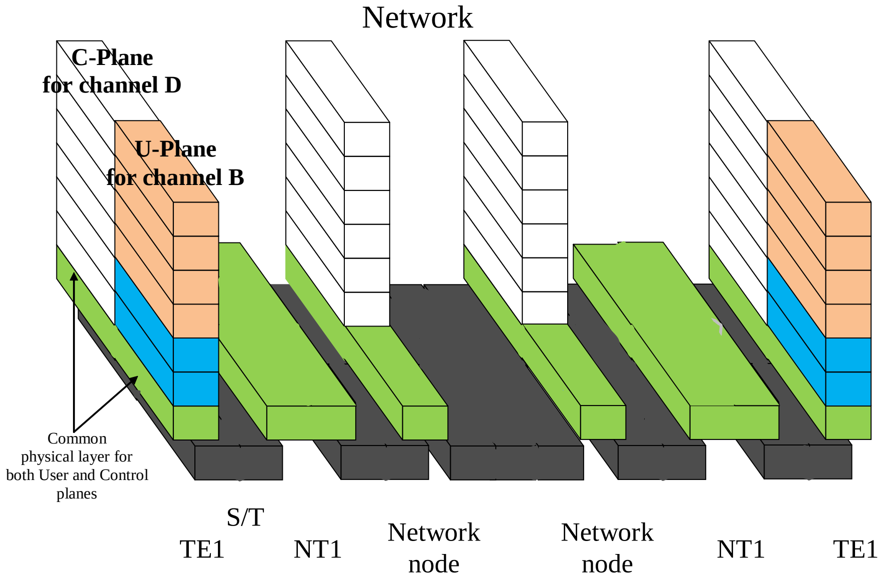
- **Basic Rate Interface (BRI)**
  - Two 64 kbit/s B channels plus one 16 kbit/s D channel (2B+D) for a total information rate of 144 kbit/s: 2B+D. This basic service is intended to meet the needs of most individual users.
- **Primary Rate Interface (PRI)**
  - 23 B channels in USA and 30 B channels in Europe plus one 64 kbit/s D channel (totally, 1536 kbit/s in USA and 1984 kbit/s in Europe): 23B+D or 30B+D.

ISDN Services from ITU-T Recommendation I.210

- **Bearer services** to transfer digital information between end-points (S or T) across the network (Recommendations from I.230 to I.233).
  - **Circuit services** (the network is a physical relay system) with transparent and non-transparent circuits at different bit-rates.
  - **Frame mode service** (the network operates as a relay at layer 2). The name is due to the fact that the packet data units are also named frames at layer 2. Two different cases are possible:
    - Frame switching, where the network uses a complete layer 2.
    - Frame relaying, where only part of layer 2 (i.e., the lower part) is implemented within the network.
  - Packet mode service (the network operates a relay at layer 3, i.e., a packet-switched network). Practically, only the virtual circuit service has been defined that uses at layer 3 the corresponding X.25 protocol.
- **Teleservices**: Teleservices involve OSI protocols from layer 1 to layer 7. Teleservices rely on bearer services for the transport of information from one end to another of the network. Examples: telephony, videotelephony …
- **Supplementary services**: Supplementary services are provided together with a bearer service or a teleservice. Examples: calling number notification, group calls, etc.

ISDN Protocol Stack according to Rec. I.320

- The OSI reference model (as well as X.25) conceived **'in-band' signaling**: e2e signaling is managed by the same protocol stack as information traffic.
  - This approach is incompatible with circuit-switching, where once a circuit is established, information is transparently conveyed by the network (relay system at level 1).
- The ISDN protocol stack is an evolution of OSI model with two parallel stacks: one for **information traffic** (User Plane, channel B with 'out-of-band' signaling) and the other for **signaling traffic** (Control Plane, channel D with ' in-band' signaling).



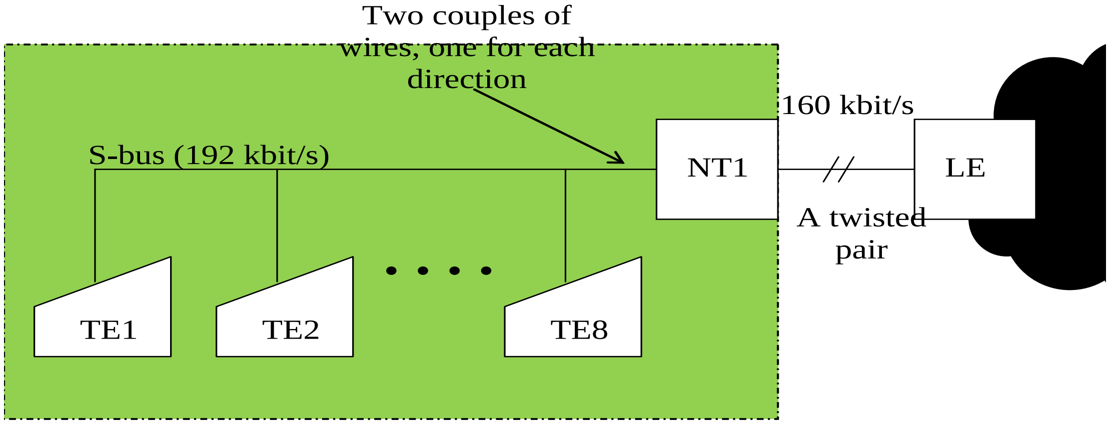
Example of protocol stacks at different interfaces for a circuit-switched ISDN connection

ISDN PHY Layer

- As specified in Recommendation ITU-T I.431, ISDN PRI uses the same layer 1 of the 2 Mbit/s **E1 numeric transmission** (ITU-T G.703 and ITU-T G.704 Recommendations).

- As specified in Recommendation ITU-T I.430, ISDN BRI layer 1 is based on a **passive bus with up to 8 TE1 connected** to NT1. In the link between NT1 and the local exchange of the network there is a full-duplex transmission at 144 kbit/s (2B+D) over a twisted pair copper cable. **At the customer site, the 2-wire U interface is converted into a 4-wire S/T interface by the NT1**.

Multi-point architecture



ISDN Layer 2

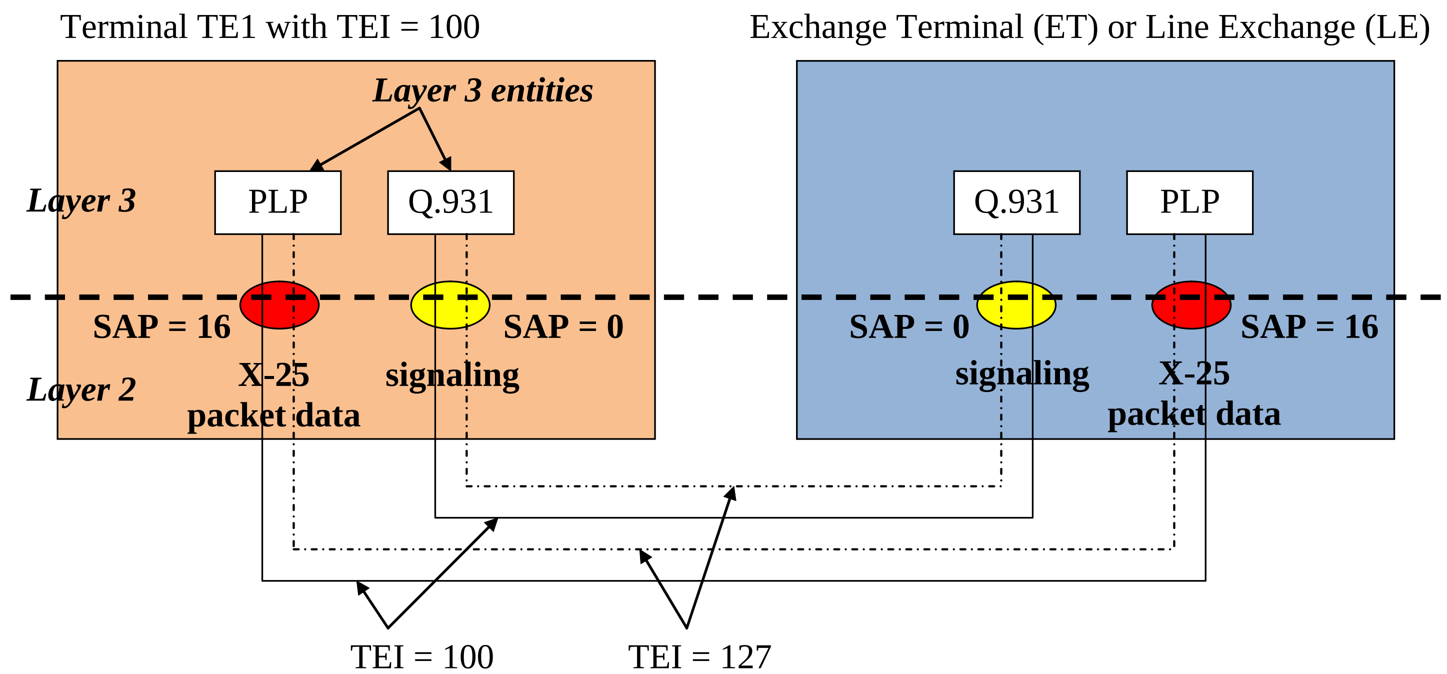
- The ISDN protocols specified for layers 2 and 3 are only valid for **D channels** (Recommendations ITU-T Q.920 and ITU-T Q.921).
- Layer 2 protocol is based on HDLC and on its frame structure. In particular, the protocol is named **Link Access Procedure on the D-channel (LAPD)**.
- Layer 2 has the specific task of allowing the communication between peer layer 3 entities. A layer 3 entity is identified by a Service Access Point (SAP). There are two types of SAPs, each denoted by a suitable SAP Identifier (SAPI): SAP = 0 for signaling and SAP = 16 for packet data traffic.
- To distinguish different TEs in a multi-point connection a suitable Terminal Endpoint Identifier (TEI) is used.
  - **Each layer 2 connection is therefore identified by SAPI + TEI**, that together form the Data Link Connection Identifier (DLCI), the address field of a LAPD frame.

ISDN Layer 3

- Layer 3 is specified by ITU-T Recommendations Q.930, Q.931 and Q.932 for signaling traffic carried by channel D.
  - These protocols have the task to manage the exchange of end-to-end information through the network on channels B.
  - In case of data packet traffic on channel D, PLP (layer 3 of X.25) is used.

ISDN: an Example of Layers 2&3

- Example of ISDN layer 2 LAPD addressing for sending packet data and signalling from a line exchange to a TE1:



BRI ISDN: Details on S/T and U-Interfaces

S/T Interface

- **4 wires, 2 wires per each direction.**
- **Time division frame with 48 bits in 250 μs** (bits divided among 2 B channels and a D channel) with corresponding gross bit-rate of 192 kbit/s. Additional bits (with respect to those needed for 2B+D, 144 kbit/s) are necessary for synchronism, signaling, framing, etc.
- **A pseudo-ternary AMI line code** is used for transmissions (to avoid the DC component in the signal in order to allow the coupling of transformers).

U Interface

- **Specified in ITU-T Recommendation G.961.**
- **Two Binary One Quaternary 2B1Q (ANSI T1.601) line coding** is used.
- **Two wires (twisted pair) and bidirectional transmissions**: duplex transmission shall be achieved through the use of Echo Cancellation (ECH) or Time Compression Multiplex (TCM).
- With the TCM or "burst mode" method transmissions on the 2 wire links are separated in time. **Blocks of bits (bursts) are sent alternatively in each direction (ping-pong transmissions).** Bursts are passed through buffers at each transceiver terminal such that the bit stream at the input and output of the TCM transceiver terminal is continuous at the rate R. The bit rate of the line has to be to be greater than 2R to provide for an idle interval between bursts, which is necessary to allow the **transmitter/receiver turn-around**.

Frame Relay

Frame-Relay Networks: Introduction

- These networks are based on **a layer 2 protocol, named Frame Relay**, that can be considered as a **special case of the (packet-switched) layer 2 protocol used in ISDN**.
- Frame relay was one of the "fast packet switching" technologies introduced in the early Nineties.
- Frame relay entails lower overhead and achieves higher performance than X.25 networks.
- The **ITU Recommendations** (coherent with ANSI standards) are: I.233, Q.922 Annex A, and Q.933.

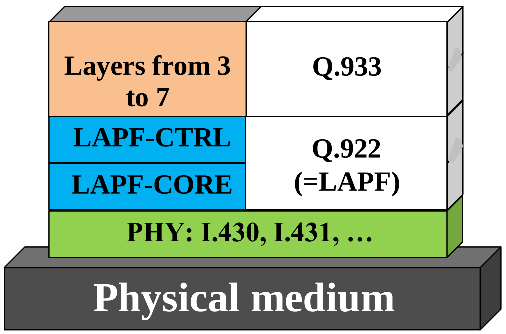
Frame-Relay Networks: Basic Characteristics

- **There is no correction/recovery and no flow control in the network links; both tasks are end-to-end performed.**
  - With the adoption of optical fibers, error rates are drastically reduced (from 10^{-6} to 10^{-9}), thus making useless to perform error recovery on each link (ARQ).
- Frame relay is a **connection-oriented protocol with virtual circuits** (an end-to-end connection must be established before data can be transferred).
- **Switching is performed at layer 2** (differently from X.25 networks, where switching was performed at layer 3).
- The protocol stack employs **a user plane (data, information flow) and a control plane (signaling)**.
  - Signaling is out-of-band as in ISDN and differently from X.25.

Frame-Relay Networks: Protocol Stack

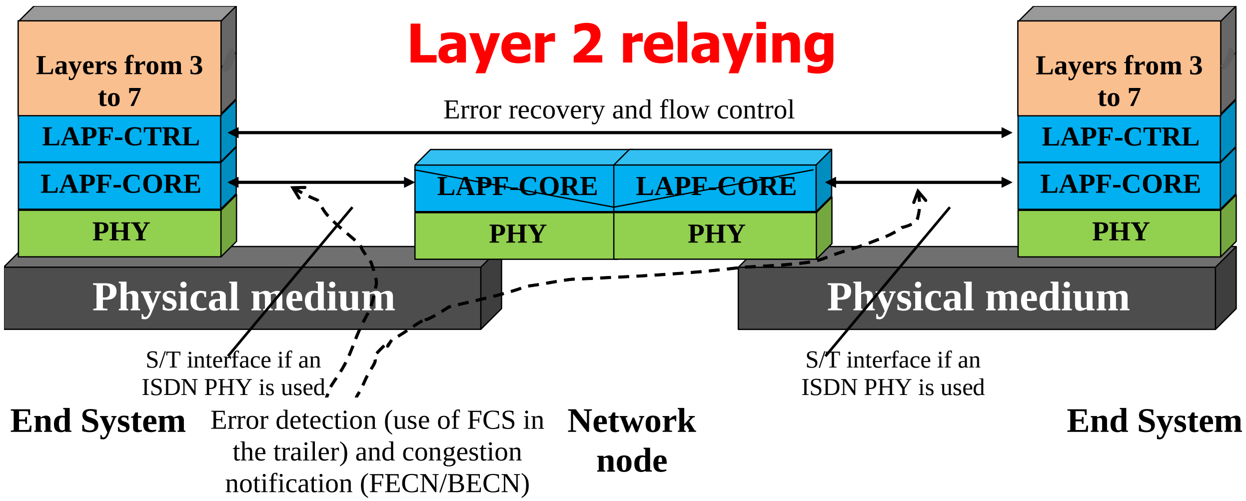
- **Layer 1 (PHY)**: it is common for user and control planes. It is based on typical ISDN physical resources (ISDN I.430 or ISDN I.431).
- **Layer 2**: the control plane adopts the full LAP-F protocol defined in Q.922, whereas the user plane LAP-F protocol is divided into two parts:
  - **Functions of LAP-F core** (Annex A of Q.922): framing, multiplexing/demultiplexing of virtual circuits, error detection, etc.
  - **Functions of LAP-F control**: **error recovery** (ARQ protocol) and **flow control**; in the typical frame relay service (and network) LAP-F control is only end-to-end operated.
  - **End hosts have both LAP-F core and LAP-F control; intermediate nodes only have LAP-F core in the frame relay service.**
- **Layer 3**: on the control plane the Q.933 protocol (management of virtual calls) is adopted. On the user plane, we have a simplified layer 3 protocol only at the end systems.

Frame-Relay Networks: Protocol Stack (cont'd)



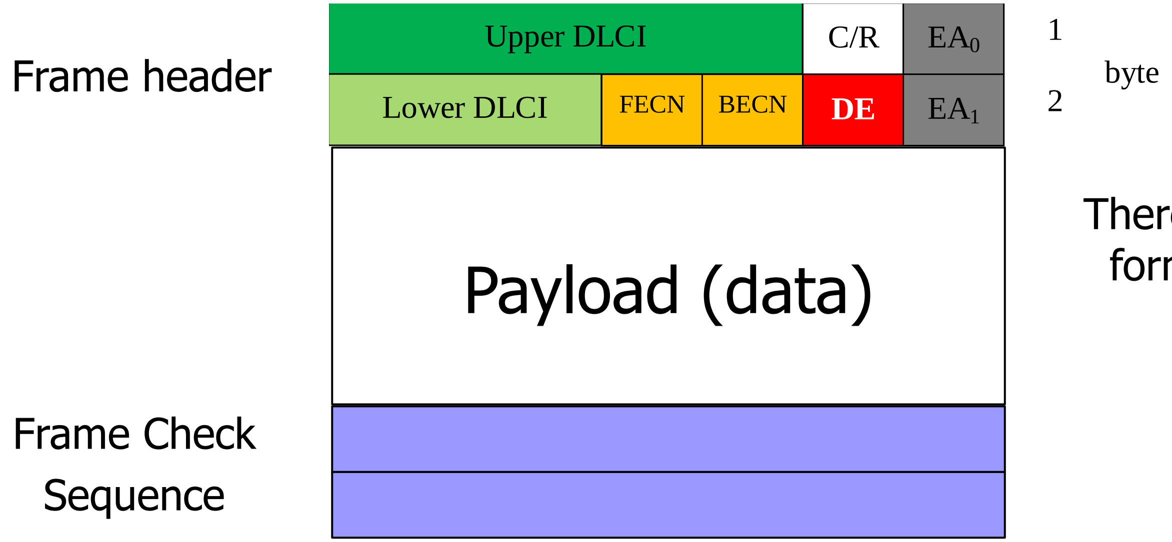
User and control plane at the end system.

Frame relay service: user plane protocols in network nodes and at the end system. Note that error recovery and flow control are end-to-end performed.



Frame-Relay Networks: Frame Format

- User and control planes convey data organized in layer 2 messages called **frames**.
- Frames are switched trough **virtual circuits** by means of the address field called Data Link Connection Identifier (DLCI). The DLCI field has only a local meaning; it can be changed at each node according to the path defined during the set-up phase.



There are different frame formats; this is just an example.

Frame-Relay Networks: Frame Format (cont'd)

- DLCI of different length: 10, 16 or 23 bits
  - DLCI = 0 is reserved for a channel that conveys signalling for all the virtual connections on the same link. DLCI field with all bits equal to 1 is for a channel that transports management information on the link.
- Address Extension (EA) bit at the end of each byte in the address field
  - EA = 0 except for the last byte of the address field where EA = 1.
- Forward Explicit Congestion Notification (FECN) bit
  - if it is set to 1 by an internal node of the frame relay network, it denotes a congestion situation on the related link on the path towards the destination of the frame.
- Backward Explicit Congestion Notification (BECN) bit
  - If it is set to 1 by an internal node of the frame relay network, it denotes a congestion situation on the link where the frame is sent, but in the opposite direction.
- Discard Eligibility (DE) bit
  - If it is set to 1 by an access node to the frame relay network, it authorizes to discard with priority the related frame (with respect to those with DE = 0) in congested nodes.

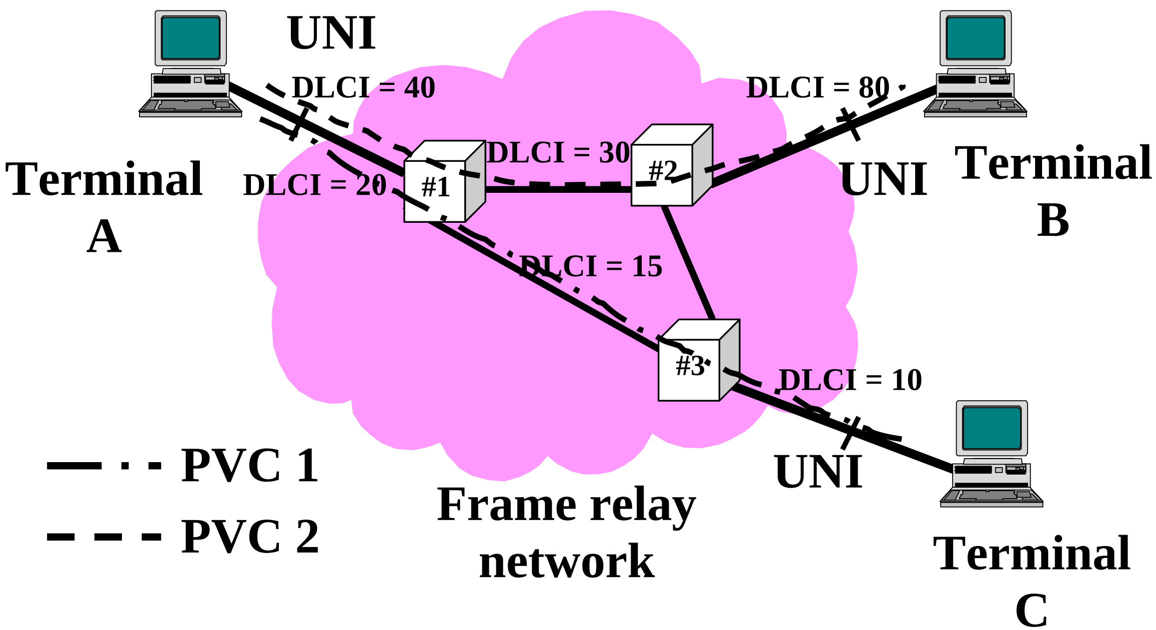
Frame-Relay Networks: Frame Format (cont’d)

- Frames are produced by a source with FECN = 0, BECN = 0, DE = 0.
- The DE bit can be modified at the first (access) node of the frame relay network.
- FECN and BECN bits can be modified at any node in the frame relay network.

Frame-Relay Networks: Addressing and Switching

End-users are interconnected using Virtual Circuits, which can be either Permanent Virtual Circuits (PVC) or Switched Virtual Circuits (SVC).

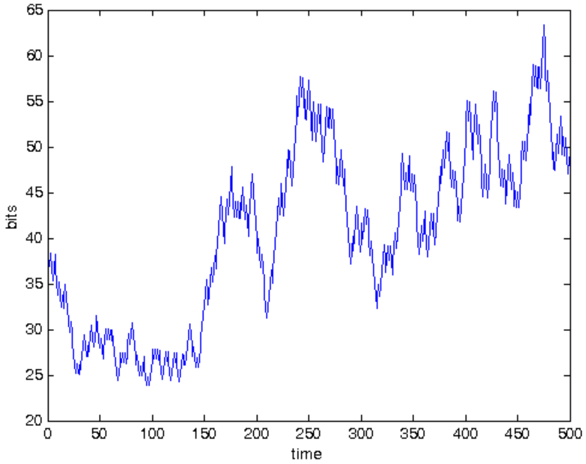
Switching is performed on the basis of the DLCI.



... an example

Frame-Relay Networks: Traffic Burstiness

- Sources generate traffic that typically has not a constant bit-rate, but a variable one with possible impulses. A traffic with impulses is said to be bursty.



- Traffic burstiness causes sudden congestion at the buffers of the nodes and consequent high delays and packet losses.

Frame-Relay Networks: Traffic Regulation (policer)

- Let us consider a variable bit-rate traffic source with an access line to the network with capacity, Access bit-Rate (AR), much greater than the maximum traffic load that can be accepted in the network.

- During the connection establishment phase, the following flow control parameters are defined [according to a certain Service Level Agreement (SLA)] to monitor (**policer**) the input traffic:
  - **Measurement interval**, T_c, i.e., the time interval on which we measure the input traffic to determine whether it is conformant to specifications or not. T_c is the time basis (periodicity) according to which the input traffic is monitored.
  - **Committed burst size**, B_c, that denotes the maximum number of bits that the network is able to accept and convey in a time T_c from a given source.
  - **Excess burst size**, B_e, that represents the maximum number of excess bits (with respect to the B_c value) that the network will try to convey at destination in T_c without any special guarantee.

Frame-Relay Networks: Traffic Regulation (cont'd)

- Committed Information Rate (CIR):

$$CIR = \frac{B_c}{T_c} \quad \left[\frac{\text{bit}}{\text{s}}\right]$$

- Excess Information Rate (EIR):

$$EIR = \frac{B_e}{T_c} \quad \left[\frac{\text{bit}}{\text{s}}\right]$$

- Frames sent in a given T_c interval and requiring the extra capacity (of the B_e bits in T_c) are marked with DE = 1, so that they can be discarded at an intermediate node if there is congestion.

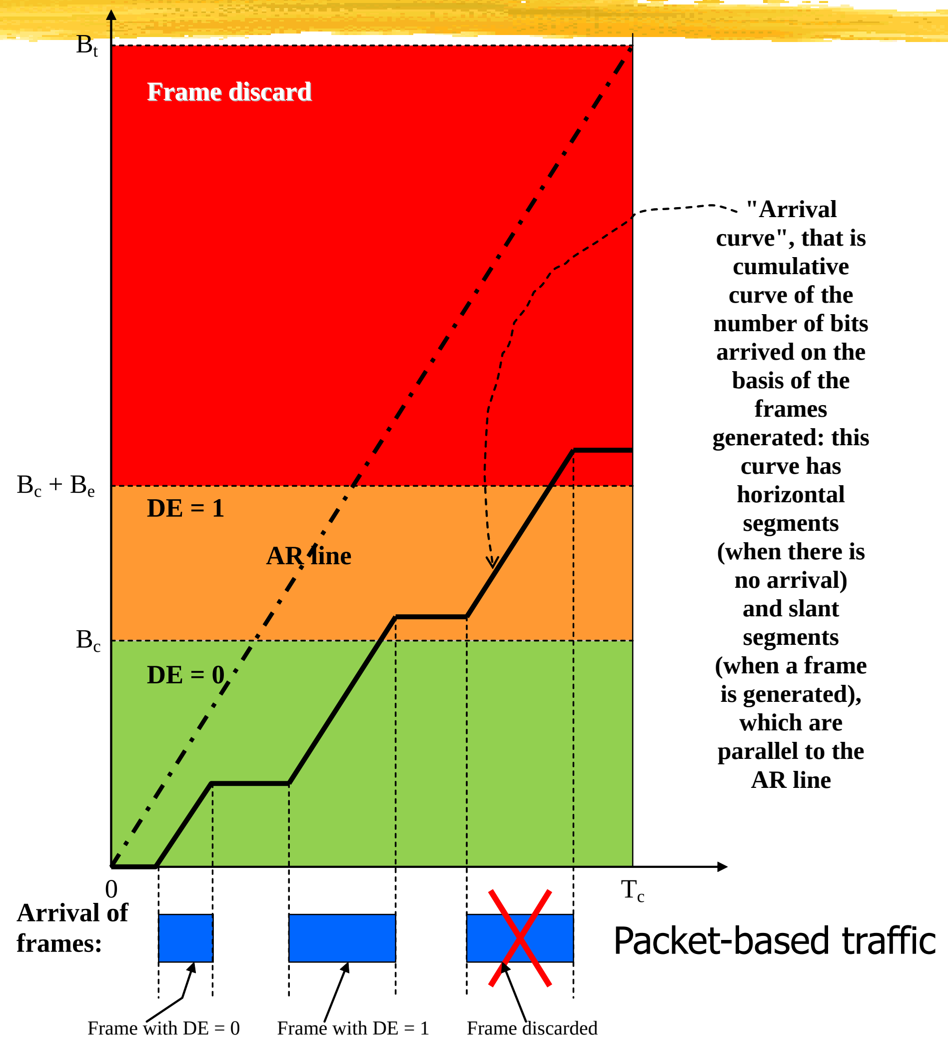
- The access capacity AR must fulfil the following condition:

$$CIR + EIR \leq AR \quad \left[\frac{\text{bit}}{\text{s}}\right]$$

Frame-Relay Networks: Traffic Regulation (cont'd)

Note that higher values of T_c are preferable for users since they allow sending bursts of data (burstiness).

However, from the network standpoint, lower T_c values are preferable since they permit a better control on the traffic injected into the network.



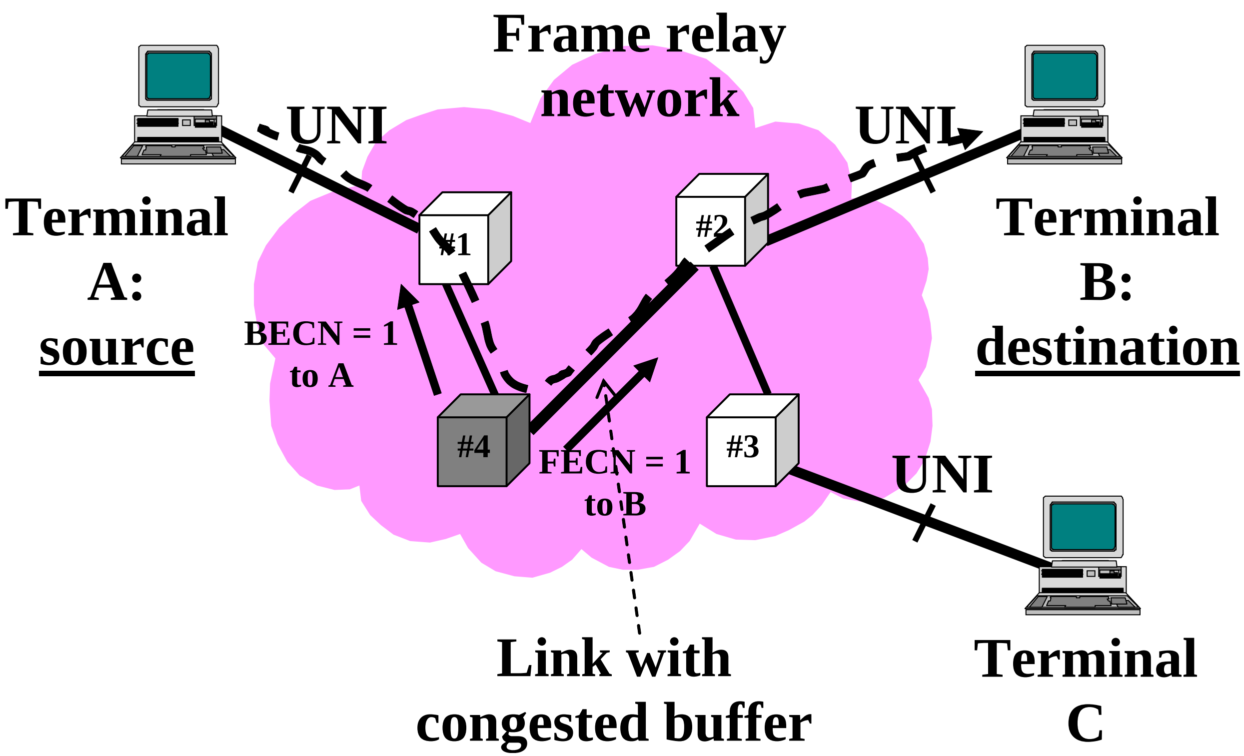
Frame-Relay Networks: Congestion Control

- Congestion control is a crucial part of telecommunication networks.
  - The occurrence of congestion leads to the discard of frames, unpredictable delays, low network throughput, and the possible need of retransmissions.
- In Frame Relay, congestion control is operated by the end-user and by the network:
  - Each node in the network is in charge of monitoring congestion and reporting it to the terminals by means of a mechanism described in the following slide.
  - The terminals have the responsibility to react accordingly.

Frame-Relay Networks: Congestion Control (cont'd)

- If a link (buffer) is congested, the related node can discard frames starting from those with **DE = 1** (as set by the access policer).
- Each node controls the occupancy of its buffers; when a threshold value is exceeded for the buffer of a given link, a congestion notification is made for **all the virtual channels that use this link**.
  - **FECN is set to 1** for all the frames that from this node are sent through the bottleneck link. FECN can be used by the destination device in the case that its upper layer protocols can control the traffic injected by the source through an end-to-end feedback signaling.
  - **BECN is set to 1** for all the frames that are received by the node through the bottleneck link. BECN notifies the sender that there is congestion in the network and that a bit-rate reduction is needed.

Frame-Relay Networks: Congestion Control (cont'd)



Congestion Notification in IP Networks, a Note

- **An FECN-like approach is also used by the IP protocol.**
  - **Explicit Congestion Notification (ECN)** is an extension to the Internet Protocol and to the Transmission Control Protocol (TCP) and is defined in RFC 3168 (2001). ECN allows end-to-end notification of network congestion without dropping packets.
  - ECN is an optional feature that is only used when both endpoints support it and are willing to use it.
  - ECN uses the two least significant (right-most) bits of the DiffServ field in the IPv4 or IPv6 header (see Lesson No. 14).
  - At the receiving endpoint, this congestion indication is handled by the upper layer protocol (i.e., TCP) and needs to be echoed back to the transmitting node in order to signal it to reduce its transmission rate.

A Summary on Flow/Cong. Control in Frame Relay

- **LAP-F core**
  - Policer regulating the traffic entering the network at layer 2 on the basis of the contractual traffic conditions: use of the DE bit.
  - Buffer management for congestion at nodes on the basis of DE of the received frames.
  - Congestion notifications at layer 2 based on flags FECN and BECN that can be set at intermediate nodes if they are congested.
- **LAP-F control**
  - End-to-end congestion control at layer 3 or above on the basis of FECN.

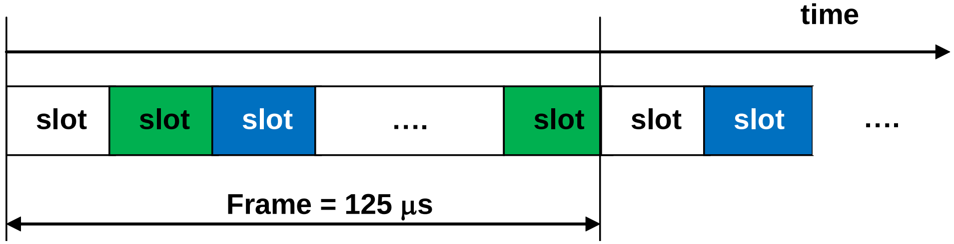
Plesiochronous and Synchronous Multiplexing Hierarchies

PCM Voice Codec

- Human voice ranges from about 20 Hz to about 14 kHz. When the telephone system was designed it was decided to reduce this bandwidth for reasons of economy.
  - Net phone bandwidth from 300 Hz to 3400 Hz.
  - Voice is channelized at **4 kHz** (net band plus guard-bands).
  - One voice sample every $T_c = 1/8000$ s = 125 μs (Nyquist sampling theorem).
  - A **companding** (logarithmic) characteristic is used to compress the dynamics of voice samples. Two companding laws are possible, referred to as **A-law** for Europe and **μ-law** for USA and Japan. The obtained value is quantized with 8 bits (7 bit in USA). Hence, **8 bits every 125 μs correspond to a bit-rate of 64 kbit/s**; this is **the classical voice codec of the Pulse Code Modulation (PCM)** system, specified by ITU-T G.711 Recommendation.
  - **The frame duration of 125 μs is used at all levels of the time-division multiplexing hierarchy** (both USA and ITU-T standards: PDH and SDH/SONET) for the transport of multiplexed voice traffic flows. **The frame duration represents the time-basis for resource allocation to different users**.

Voice Digital Multiplexing: TDM

- The procedure according to which the signals of different users are transmitted through the same physical resource (e.g., a cable, an optic fiber, etc.) without generating mutual interference is called **multiplexing**.
- **Time Division Multiplexing (TDM)** at different hierarchical levels is used in digital telephony and data communications. Let us refer below to the **TDM signal at the first level of the hierarchy**.
  - There is a **frame structure of 125 μs**. All signals are transmitted in the same bandwidth, but at distinct times organized in slots.
  - Slots may be permanently assigned or assigned on demand to users.
  - **A slot conveys the digitized representation of a voice sample (1 byte).**



First Level of the ITU-T TDM Hierarchy: E1 and PCM-30

- **E1 (also named PCM-30)** has a capacity of 2.048 Mbit/s and uses **line encoding** in order both to eliminate the DC component from the digital baseband transmission and to help a fast synchronization to the signal.
- The periodic use of **one timeslot (i.e., 8 bits)** per frame corresponds to a capacity of 64 kbit/s.
- Timeslots are numbered from 0 to 31.
- The **E1 signal can be structured** or unstructured.
- Let us describe the organization of a structured E1 signal:
  - **Time slot 0**: Carries framing information in a frame alignment signal as well as remote alarm notification, five national bits, and optional Cyclic Redundancy Check (CRC) bits.
  - **Time slot 16**: Carries out-of-band signaling. Note that every time slot in an E1 is a ‘clear channel’, that is no bits are robbed from a data time slot for signaling purposes**.**
  - The **other 30 time slots** are used for information channels at 64 kbit/s.

he ITU-T TDM nd PCM-30

The PCM code generated by the codec function has a Non-Return to Zero (NRZ) format. It cannot be directly transmitted on a transmission line because the signal contains a DC component and lacks of timing information. A **line coding** needs to be adopted to convert the NRZ code to a pseudo-ternary code that is more suitable for transmissions [e.g., Alternate Mark Inversion (AMI), Bipolar with N Zero Substitution (BNZS), and High Density Bipolar 3 (HDB3) coding]. These schemes eliminate the DC component of NRZ, thereby eliminating the troublesome '**DC wander' phenomenon**: the DC component can cause signal distortion in the circuits with AC coupling. Line coding also provides the means to detect errors, and enhances the synchronization between transmitter and receiver through the reduction of timing jitter.

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TDM Hierarchy: E1, E2, E3, E4, etc.

Level	North America	Japan	ITU
0	64 kbit/s (DS0)	64 kbit/s	64 kbit/s
1	1.544 Mbit/s (T1/DS1)	1.544 Mbit/s (J1)	2.048 Mbit/s (E1)
2	6.312 Mbit/s (DS2)	6.312 Mbit/s (J2)	8.448 Mbit/s (E2)
3	44.736 Mbit/s (T3/DS3)	32.064 Mbit/s (J3)	34.368 Mbit/s (E3)
4	139.264 Mbit/s (DS4)	97.728 Mbit/s (J4)	139.264 Mbit/s (E4)
5	400.352 Mbit/s	565.148 Mbit/s	565.148 Mbit/s

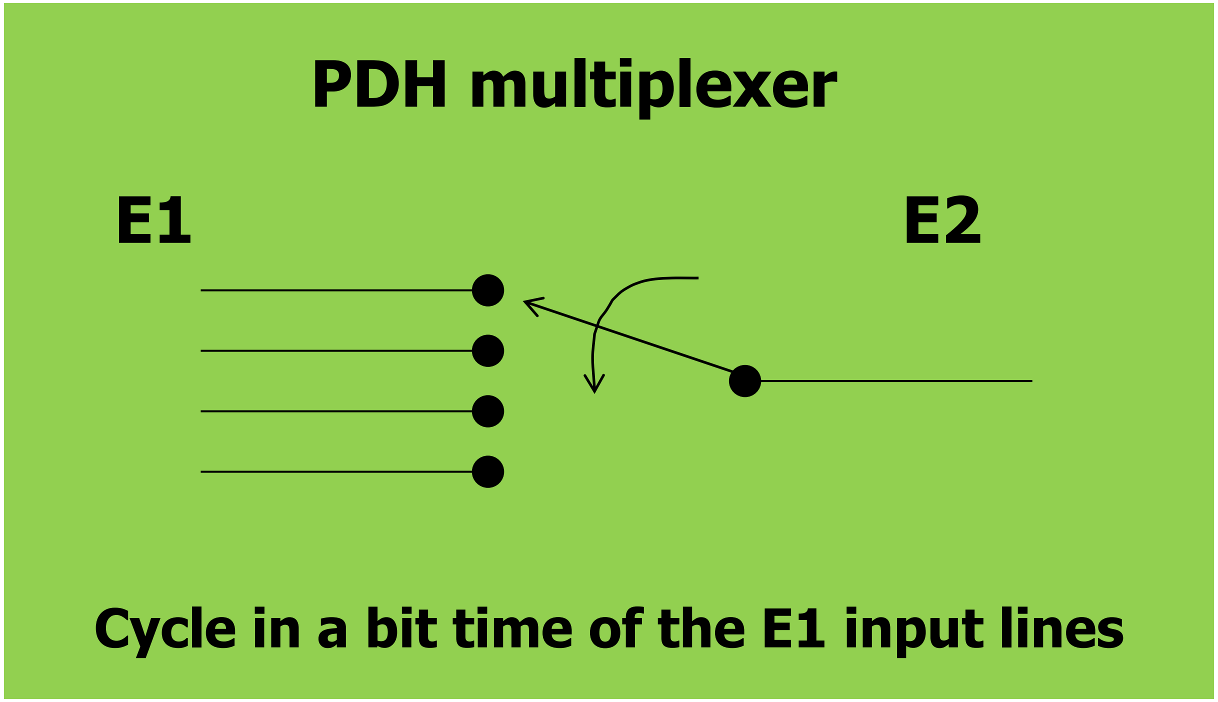
- Referring to the ITU-T standard: 32 voice channels (practically, 30 voice channels plus two control channels) are multiplexed to obtain an E1 signal; 4 E1 are multiplexed to have an E2; 4 E2 are multiplexed to have an E3; 4 E3 are multiplexed to have an E4; 4 E4 are multiplexed to obtain an E5.
- Apart the first level of the TDM multiplexing hierarchy, all the other levels are obtained by grouping 4 bearers of the lower level.

Plesiochronous Digital Hierarchy (PDH)

- We refer to a **copper medium (cable)**.
- The exact data rate of the 2.048 Mbit/s E1 data stream is controlled by a clock in the equipment generating the multiplexed data. The exact rate is allowed to vary of +/- 50 ppm (tolerance of bit timing). **Different 2.048 Mbit/s E1 data streams can probably run at slightly different rates.**
- E1 streams are multiplexed in groups of 4 to achieve E2 signals. **With PDH, multiplexing is achieved by taking 1 bit from stream #1 (i.e., sampling line #1), followed by 1 bit from stream #2, then #3, and then #4 and so on, cyclically in one bit time of E1**. The resulting E2 data stream is at 8.448 Mbit/s.
  - Since the four E1 signals may have some discrepancy in the relative timings, it may occur that the multiplexer will look for the next bit of an E1 flow, when it is not yet arrived. Hence, to compensate for these absences the transmitting multiplexer **adds additional bits called "justification" or "stuffing" bits**. This allows the receiving multiplexer to correctly reconstruct the original data for each of the 4 E1 streams.
  - The PDH multiplexing approach entails some **'problems' when a given flow has to be extracted from a higher level hierarchy**, for instance an E1 flow from an E2 signal. If the E1 multiplexed flows were truly synchronous, each E1 flow would be regularly spaced in time. However, the insertion of justification bits disrupts such characteristic: **it is impossible to demultiplex a single E1 flow simply on the basis of synchronous timing**. The only solution is to demultiplex the whole structure to determine whether justification bits are present. The whole structure must then be multiplexed again if it has to be retransmitted.
  - **'Plesiochronous' from the Greek 'almost synchronous'.**

Plesiochronous Digital Hierarchy (PDH) – cont'd

- An example of PDH multiplexing is provided in the picture below for the case from four E1 to one E2:

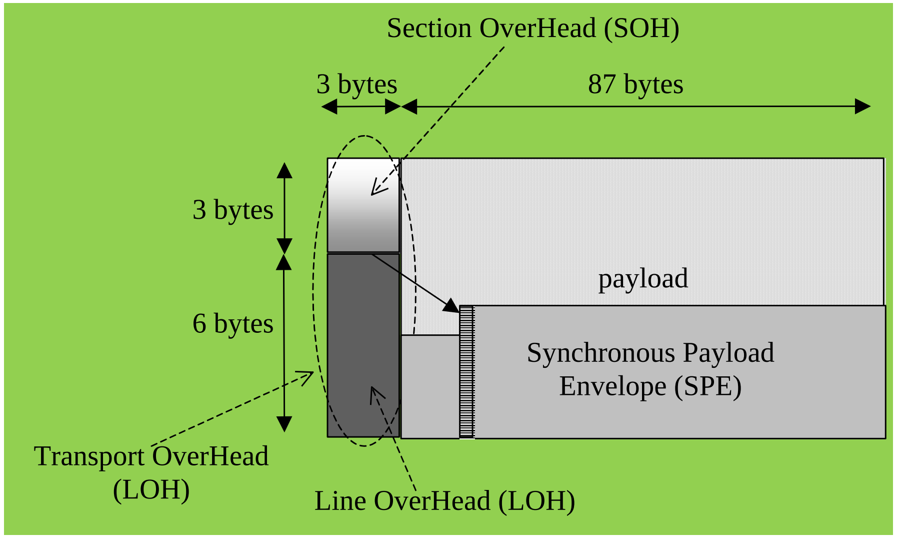


Synchronous Digital Hierarchy (SDH)

- SDH was defined in **ITU-T Recommendations** G.707, G.708 and G.709. It is suitable for **optical fiber** medium.
- In SDH the transmission of data is organized in **frames of 125 μs**, but the **multiplexing hierarchy is different with respect to PDH**.
- Differently from PDH, **SDH transport networks are tightly synchronized**: atomic clocks are used to maintain clocks synchronized in the networks (perfect synchronization is however impossible in large geographical networks).
- SDH employs a new approach to multiplex tributary signals onto a higher order one: **pointers** are used to individuate tributaries in the SDH payload.
- If a tributary signal clock slips over time with respect to the multiplexer clock, the SDH multiplexer simply recalculates the pointer from frame to frame.
- SDH allows the **direct synchronous multiplexing**: distinct slower signals can be directly multiplexed onto a higher-speed SDH signal without intermediate stages of multiplexing.
- Synchronous Transfer Mode (STM) denotes the electrical specification of the various levels of the SDH hierarchy. **The base signal for SDH is STM-1**.
- The **USA SONET** multiplexing system is similar to the SDH one.

An Example from the SONET Hierarchy

- The picture below shows an example of SONET frame structure
  - **The transmission of bytes of the matrix is from top row and moving from left to right.** The first three columns are used for section and line overhead (i.e., Transport OverHead, TOH) in relation to optical fiber network.
  - The **data payload** uses the remaining 87 columns with a column used for Path OverHead (POH).
  - A **pointer** in TOH identifies the start of the payload that is referred to as the Synchronous Payload Envelope (SPE).



Synchronous Digital Hierarchy (SDH), cont'd

- SDH and SONET hierarchies compared:

Optical Level	Electrical Level	Line Rate (Mbit/s)	Payload Rate (Mbit/s)	Overhead Rate (Mbit/s)	SDH Equivalent
OC-1	**STS-1**	51.840	50.112	1.728	-
OC-3	**STS-3**	155.520	150.336	5.184	**STM-1**
OC-12	**STS-12**	622.080	601.344	20.736	**STM-4**
OC-48	**STS-48**	2488.320	2405.376	82.944	**STM-16**
OC-192	**STS-192**	9953.280	9621.504	331.776	**STM-64**
OC-768	**STS-768**	39813.120	38486.016	1327.104	**STM-256**

Thank you!

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