Chapter-I

Telecommunication Networks

School Of Electrical And Computer Engineering
Telecommunication Network
By Melkamu.S

Outline

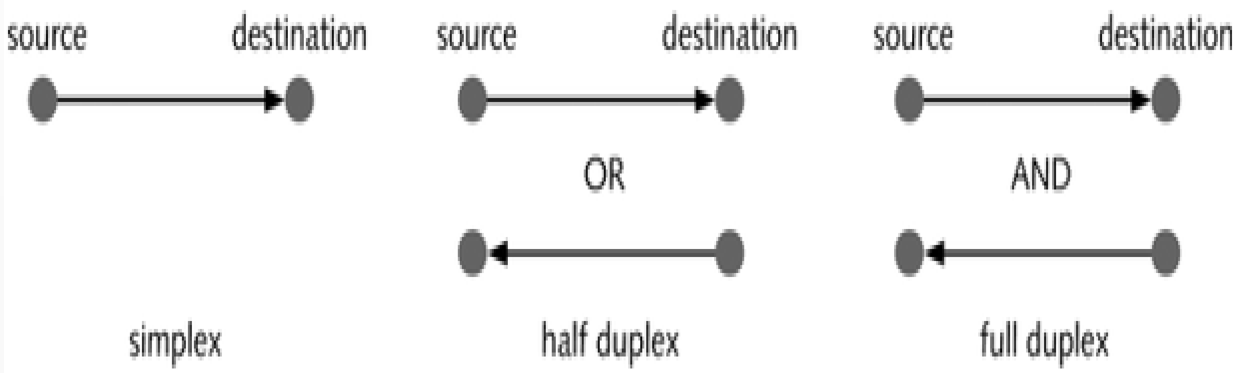
- Communication?
- Telecommunication?
- Network?
- Telecommunication Network?
- History and evolution of telecommunication network?

Communication?

- ***Communication*** is the sharing of information or messages between two or more entities.
- Elements of a Communication System
  - ***Source*** - the originator of the message, whether it is a person or machine.
  - ***Transmitter*** - the equipment that modifies the message (either data or voice) into the form required for transmission.
  - ***Communications channel*** - the means of carrying the signal from the source to the destination.
  - ***Transmission media*** - may be physical, like a copper wire or fiber optic cable, or atmospheric, like radio waves.
  - ***Receiver*** - is the device that captures the message from the communications channel and converts it into a form that the person or machine at the destination can understand.
  - ***Destination*** - the person or machine to whom the message is directed.

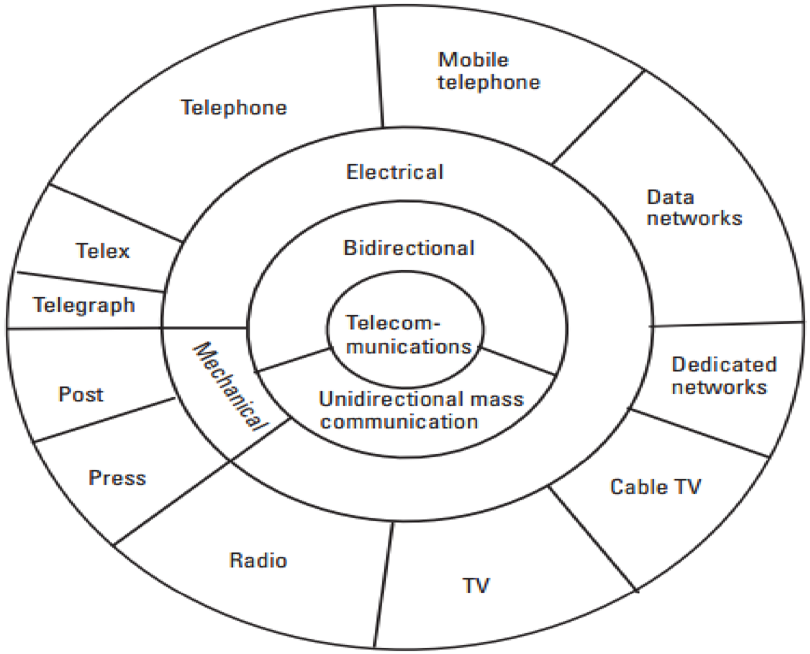
Block diagram for communication system

- **Flow of Messages**
  - *Simplex*
  - *Half-duplex communication*
  - *Full-duplex communication*



Telecommunication

- Communication that spans a distance.
- Telecommunications has been defined as a technology concerned with communicating from a distance or
- The transmission of signals over long distance, such as by telegraph, radio or television.
- It includes mechanical communication and electrical communication.



Con…

- **Voice telecommunication** - using electrical signals to transmit human voice across a distance, such as telephones and radio broadcasts.
- **Video telecommunication** - electrically-based transmission of moving pictures and sound across a distance.
- **Data telecommunication** - the use of electrical signals to exchange encoded information between computerized devices across a distance.

Significance of Telecommunications

- *Telecommunications services have an essential impact on the development of a community.*
- *The operations of a modern community are highly dependent on telecommunications.*

Some examples of services that depend on telecommunications:

- ✓ Aviation, booking of tickets;
- ✓ Banking, automatic teller machines, telebanking;
- ✓ Sales, wholesale and order handling;
- ✓ Credit card payments at gasoline stations;
- ✓ Booking of hotel rooms by travel agencies;
- ✓ Material purchasing by industry;
- ✓ Government operations, such as taxation

Network

- A Computer network consists of communications media, devices, and software needed to connect two or more computer systems or devices.
- Computer networks are essential to modern organizations for many reasons.
  - First, networked computer systems enable organizations to be more flexible and adaptable to meet rapidly changing business conditions.

Con…

- Second, networks enable companies to share hardware, computer applications, and databases across the organization.
- Third, networks make it possible for geographically dispersed employees and workgroups to share documents, ideas, opinions, and creative insights, encouraging teamwork, innovation, and more efficient and effective interactions.
- Finally, the network is increasingly the link between businesses and between businesses and their customers.

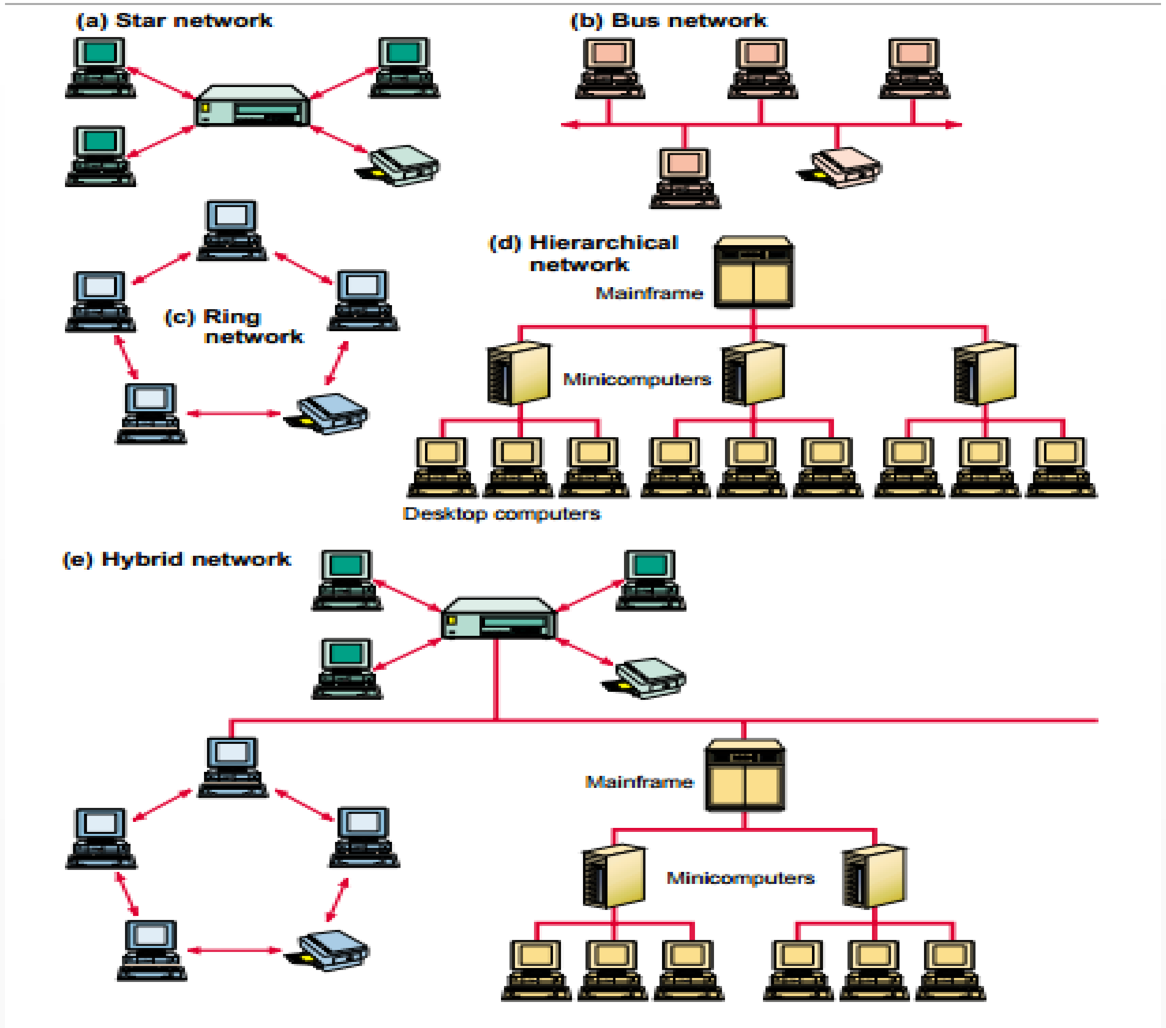
LAN

- A local area network connects two or more communicating devices within the same building.
- A LAN allows a large number of users to share corporate resources ,such as storage devices, printers, programs, and data files.
- Network have Topology and Protocols.
  - The topology of a network is the physical layout and connectivity of a network.
  - Specific protocols, or rules of communications, are often used on specific topologies,

Con…

- **Protocol** refers to the rules by which data communications take place over these channels

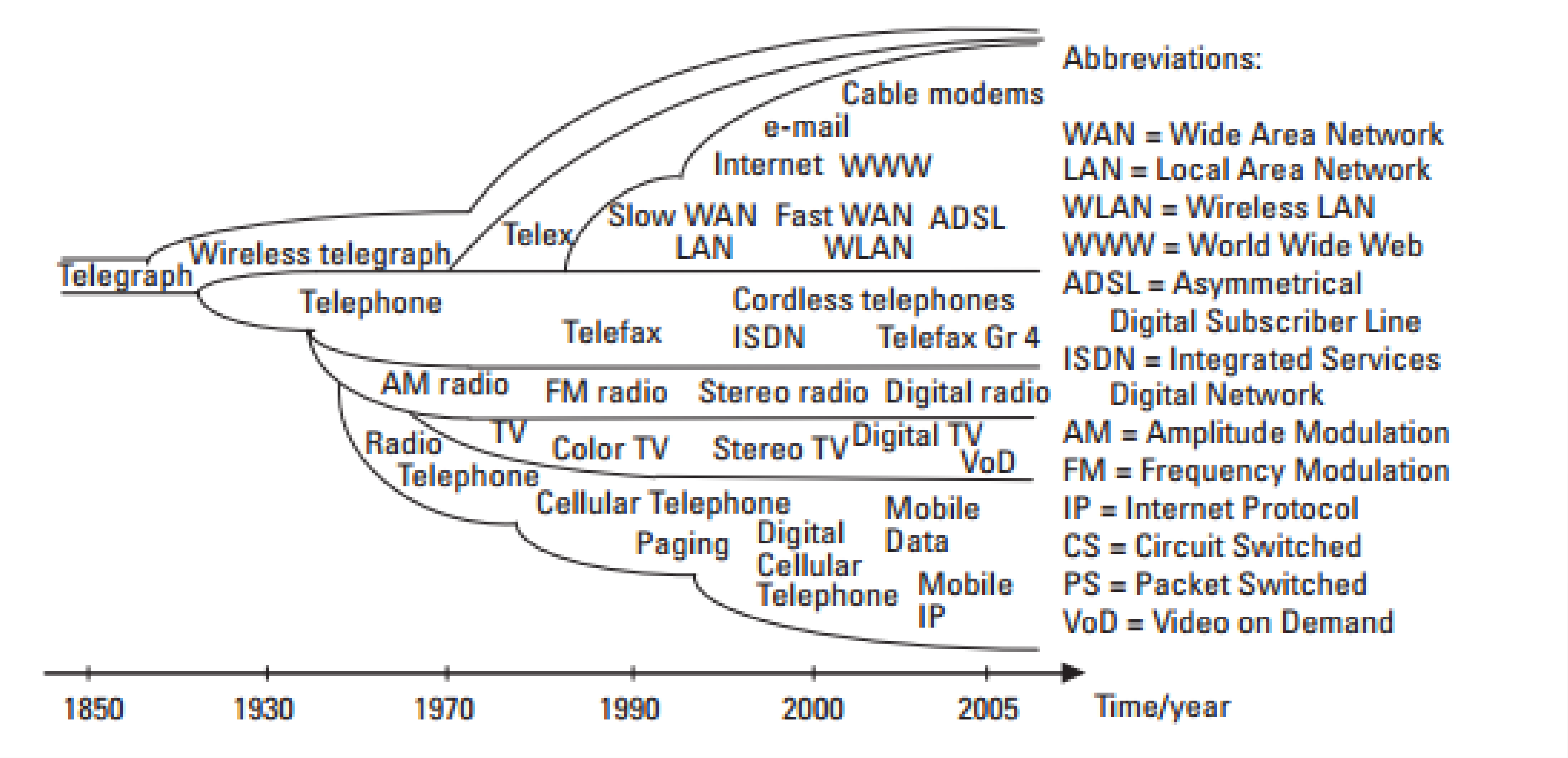
- There are five basic network topologies: star, bus, ring, hierarchical, and hybrid



WAN

- Wireless local area networks (WLANs). WLAN technologies provide LAN connectivity over short distances
- Are long-haul, broadband networks covering wide geographic areas.

Evolution of Telecommunication



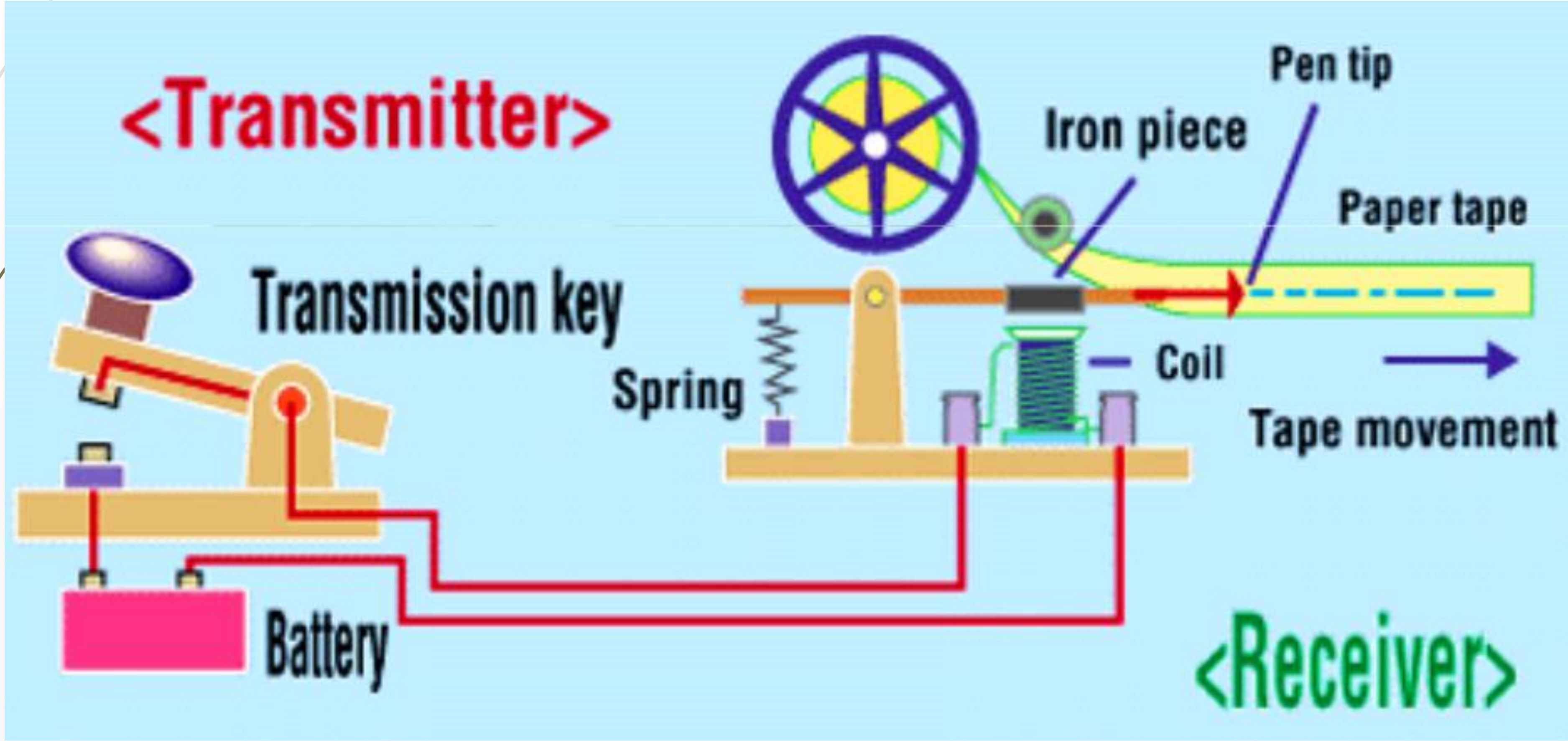
umbrella

Telegraphy

- Samuel Morse (1844) and William thomson (1855)
- from the two Greek words: Tele means "far" and graphene meaning "writing".
- So telegraphy is the long-distance transmission of messages via some signaling technology.
- It requires messages to be converted to a code, which is known to both sender and receiver.

Morse Telegraph

- Samuel Finely Breese Morse (1791 – 1872) Painter , Inventor.

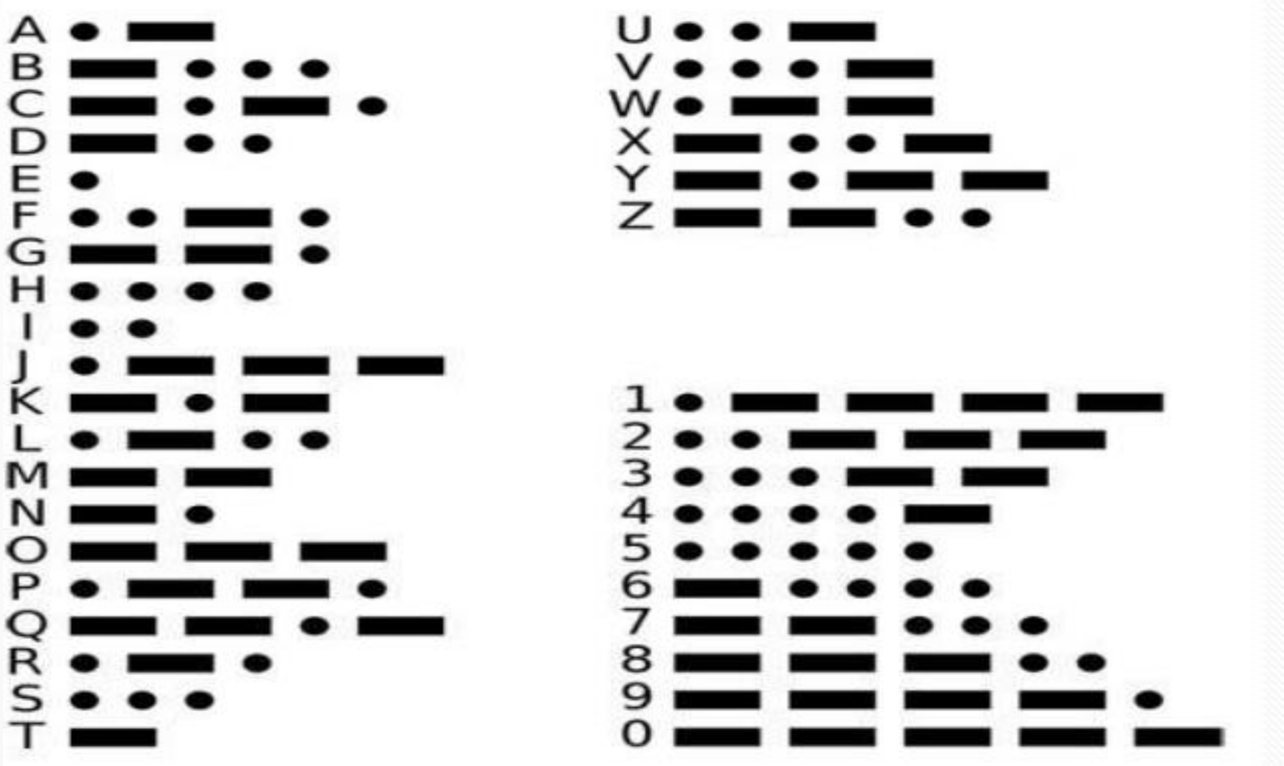


Sequences of operations

- When a finger presses the key, electricity flows from the battery through the wires and causes an impulse to flash to the electromagnet at the receiving end.
  - The impulse causes a burst of magnetism that causes two pieces of iron to be attracted to each other and make a click.
  - If the operator depresses and releases the key quickly, a short click is produced, a "dot".
  - If the operator holds down the key for a count of about three , a longer click is produced, a "dash".
  - Each letter of the alphabet is assigned its own configuration of dots and dashes.

Con…

- International Morse codes



Standardization

- Communication networks are designed to serve a wide variety of users who are using equipment from many different vendors.
- Standards are necessary to achieve interoperability, compatibility, and required performance in a cost effective manner.
- Open standards are needed to enable the interconnection of systems, equipment, and networks from different manufacturers, vendors, and operators.
- *Example International Telecommunication Union (ITU), American National Standards Institute (ANSI).*

Telecommunication Network

- An overall telecommunications network defined as a long distance communication and interconnected via networks.

ANY DOUBT AND COMMENT

???

WELCOME

Chapter-II

Basics of Telecom Networks

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Outline

- ***Introduction***
- ***Technology of communication***
- ***Operation of a Conventional Telephone***
- ***Signaling between Exchange and Telephone***
- ***Telephone Numbering***

INTRODUCTION

- The basic purpose of a telecommunications network is to transmit user information in any form to another user of the network.
- These users of public networks, for example, a telephone network, are called subscribers.
- User information may take many forms, such as voice or data, and subscribers may use different access network technologies to access the network, for example, fixed or cellular telephones.

Technology of communication

There are three technologies needed for communication through the network.

- Transmission,
- Switching, And
- Signaling.

Transmission

Transmission is the process of transporting information between end points of a system or a network.

Transmission systems use different media for information transfer from one point to another such as:

- **Copper cables**, such as those used in LANs and telephone subscriber lines;
- **Optical fiber cables**, such as high-data-rate transmission in telecommunications networks;
- **Radio waves**, such as cellular telephones and satellite transmission;
- **Free-space optics**, such as infrared remote controllers.

Switching

- In the very beginning of the history of telephony all telephones connected to each other by cables.
- However, as the number of telephones grew, it was necessary to switch signals from one wire to another.
- Now software-controlled digital exchanges developed.

Functions of switching system

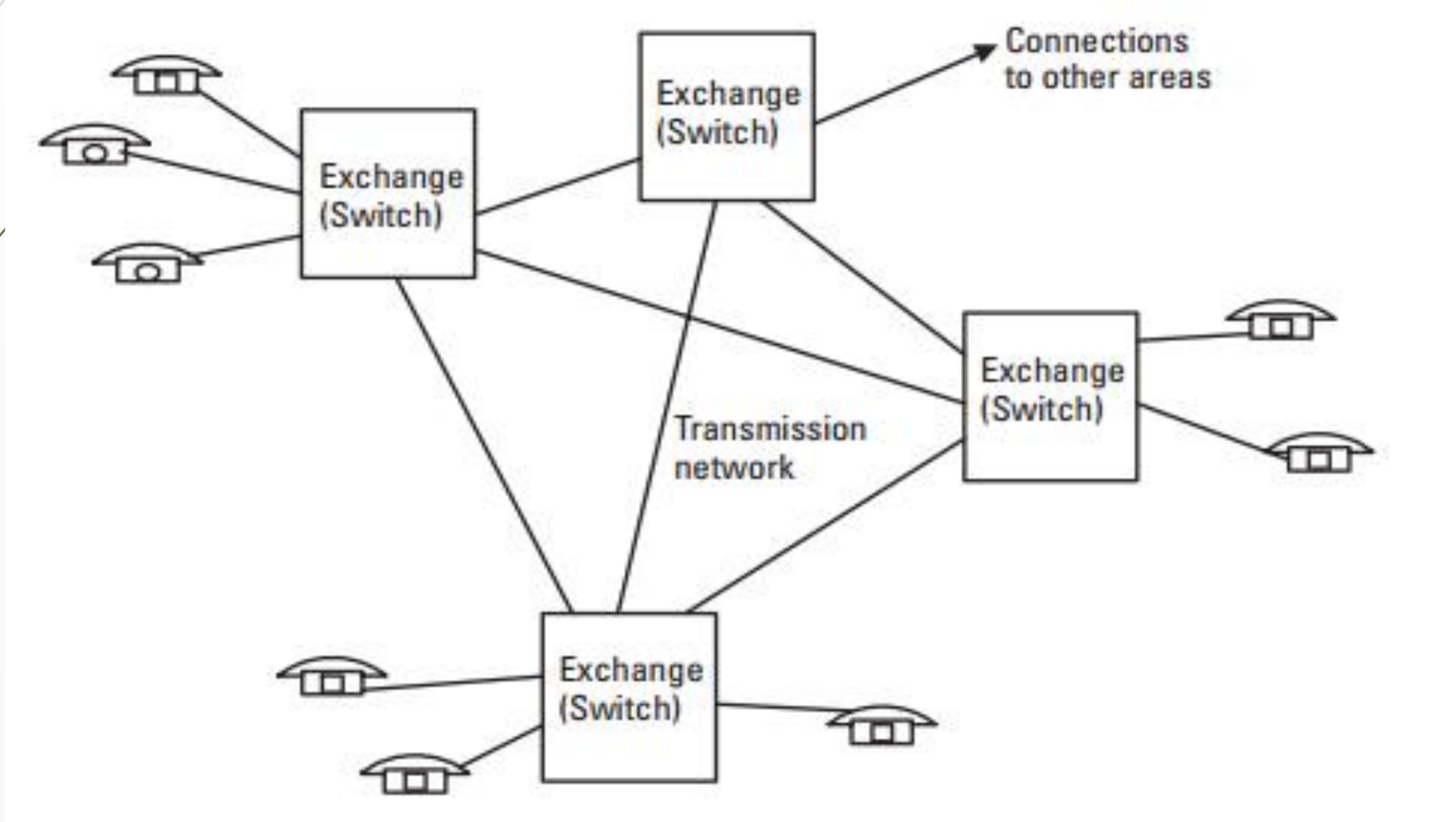
- **Identify:** The local switching center must react to a calling signal from calling subscriber and must be able to receive information to identify the required destination terminal.
- **Addressing:** The switching system must be able to identify the called subscriber from the input information (train of pulses or multiple frequency). The address may be in same local center or some other exchange.
- **Finding and path setup**: Once the calling subscriber destination is identified and the called subscriber is available, an accept signal is passed to the switching system and calling subscriber. Based on the availability, suitable path will be selected.
- **Busy testing:** If the number dialed by the calling subscriber is wrong or the called subscriber is busy or the terminal may be free but no response, a switching system has to pass a corresponding voice message or busy tone after waiting for some time.

Con…

- **Supervision:** Once the path is setup between calling and called subscriber, it should be supervised in order to detect answer and clear down conditions and recording billing information.
- **Clear down:** When the established call is completed, the path setup should be disconnected. By clear signal, the switching system must disconnect the path setup between calling and called subscriber.
- **Billing:** A switching system should have a mechanism to meter to count the number of units made during the conversation. This information and if any should be sent to the calling subscriber.

Con…

- Modern exchanges usually have quite a large capacity—tens of thousands subscribers—and thousands of them may have calls ongoing at the same time.



Signaling

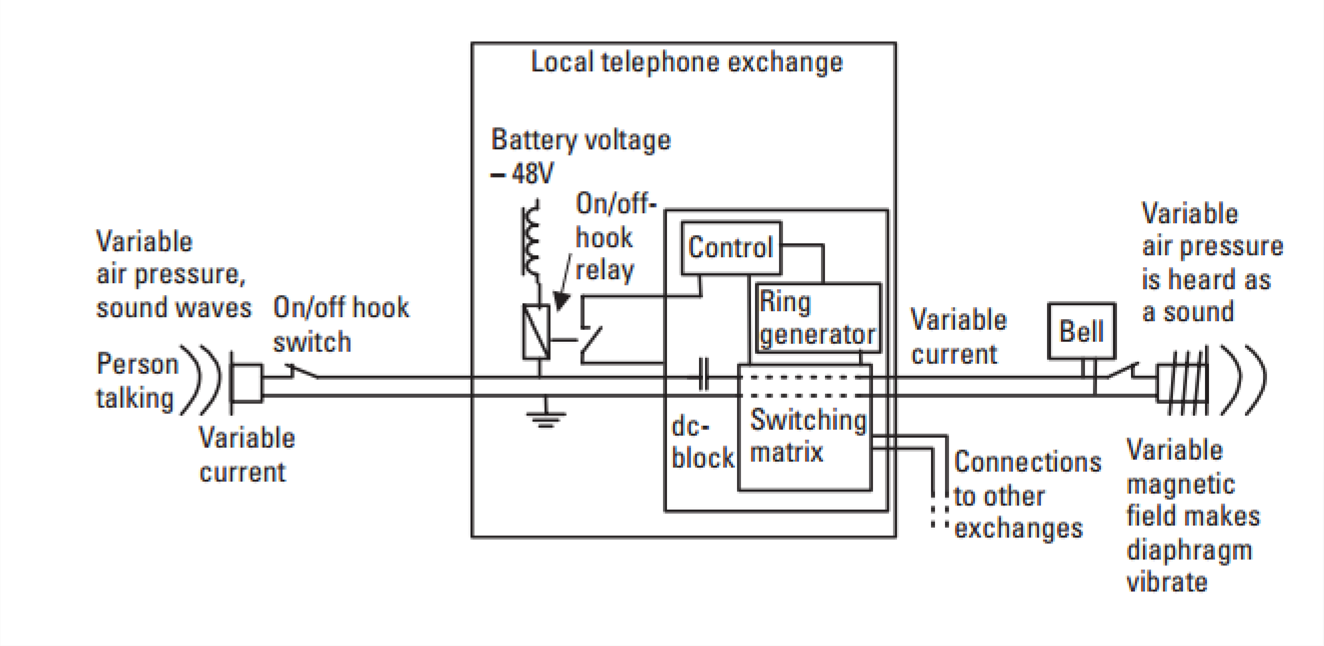
Signaling is the mechanism that allows network entities (customer premises or network switches) to establish, maintain, and terminate sessions in a network.

Some examples of signaling examples on subscriber lines are:

- ***Off-hook condition:*** The exchange notices that the subscriber has raised the telephone hook (dc loop is connected) and gives a dial tone to the subscriber.
- ***Dial:*** The subscriber dials digits and they are received by the exchange.
- ***On-hook condition***: The exchange notices that the subscriber has finished the call (subscriber loop is disconnected), clears the connection, and stops billing.

Operation of a Conventional Telephone

- Home telephone receives the electrical power from the local exchange via two copper wires.
- This subscriber line, which carries speech signals as well, is a twisted pair called a local loop.



Microphone

- When we raise the hook of a telephone, the on/off hook switch is closed and current starts flowing on the subscriber loop through the microphone that is connected to the subscriber loop.
- The microphone converts acoustic energy to electrical energy.
- When sound waves pressed the carbon grains more tightly, loop resistance decreased and current slightly increased.
- The variable air pressure generated a variable alternating current to the subscriber loop.
- This variable current contained voice information.

Earphone

- Alternating current, generated by the microphone, is converted back into voice at the other end of the connection.
- The earphone has a diaphragm with a piece of magnet inside a coil.
- The coil is supplied by alternating current produced by the microphone at the remote end of the connection.
- The current generates a variable magnetic field that moves the diaphragm that produces sound waves close to the original sound at the transmitting end.

Signaling between Exchange and Telephone

Telephone exchanges supply dc voltage to subscriber loops, and telephone sets use this supplied voltage for operation.

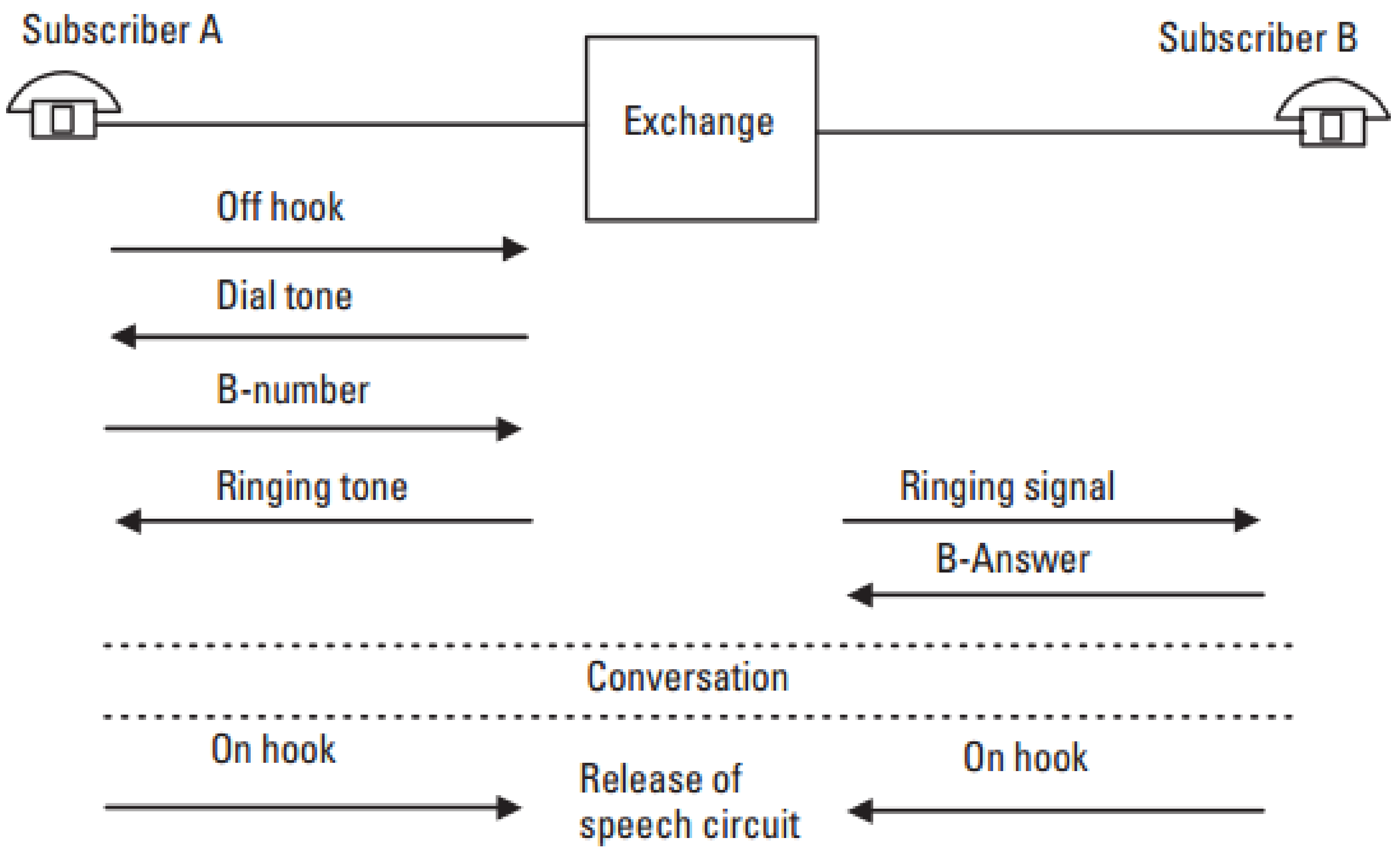
Setup and Release of a Call

- Each telephone has a switch that indicates an **on- or off-hook** condition.
- When **the hook is raised**, the switch is closed and an approximately 50 mA of current starts flowing. This is detected by a **relay** giving information to the **control unit** in the exchange.
- The control unit is an efficient and reliable computer (or a set of computers) in the telephone exchange, It activates signaling circuits, which then receive dialed digits from subscriber A.

Con…

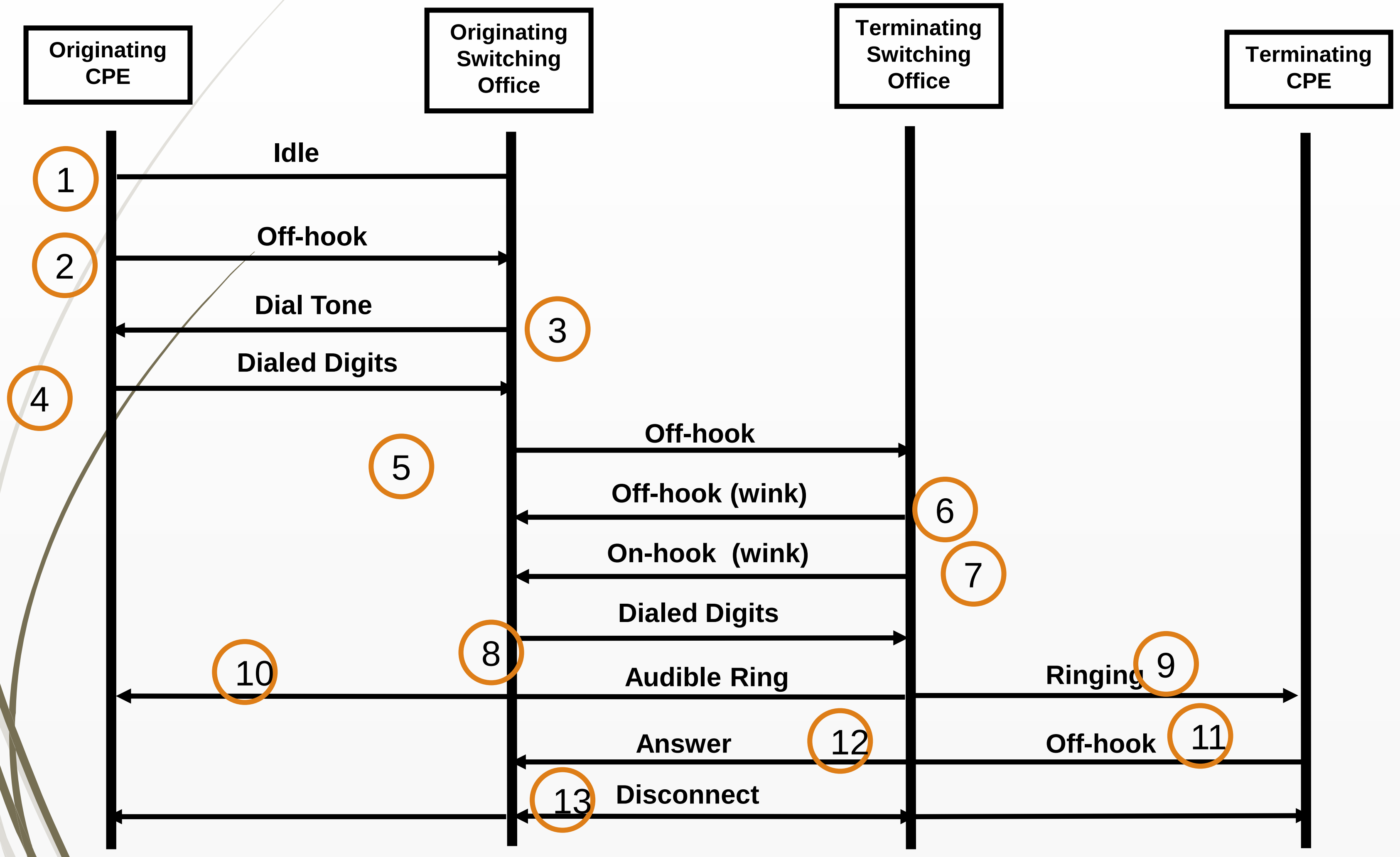
- When the call is being routed to subscriber B, the telephone exchange supplies to the subscriber loop a *ringing voltage* and the bell of subscriber B's telephone starts ringing.
- The ringing voltage is often about 70V ac with a 25-Hz frequency, which is high enough to activate the bell on any telephone.
- The ringing voltage is switched off immediately when an off-hook condition is detected on the loop of subscriber B, and then an end-to-end speech circuit is connected and the conversation may start.

Con…



Subscriber A
Exchange
Subscriber B
Off hook
Dial tone
B-number
Ringing tone
Ringing signal
B-Answer
Conversation
On hook
Release of speech circuit
On hook

General procedure from originating CPE to terminating CPE



Telephone Numbering

An international telephone connection from any telephone to any other telephone is made possible by unique identification of each subscriber socket in the world.

- **International Prefix:-**used for international calls, may differ from country to country.
- **Country Code:-**one to four numbers that define the country of called subscriber, are not needed for national calls because their purpose is to make the subscriber identification unique in the world.
- **Trunk Code, Trunk Prefix, or Area Code:-**defines the area inside the country where the call is to be routed. In the case of cellular service, the trunk code is used to identify the home network of the subscriber instead of the location.
- **Subscriber Number**:-is a unique identification of the subscriber inside a geographical area
- **Operator Numbers**:-a subscriber will need to dial additional digits to select a service provider

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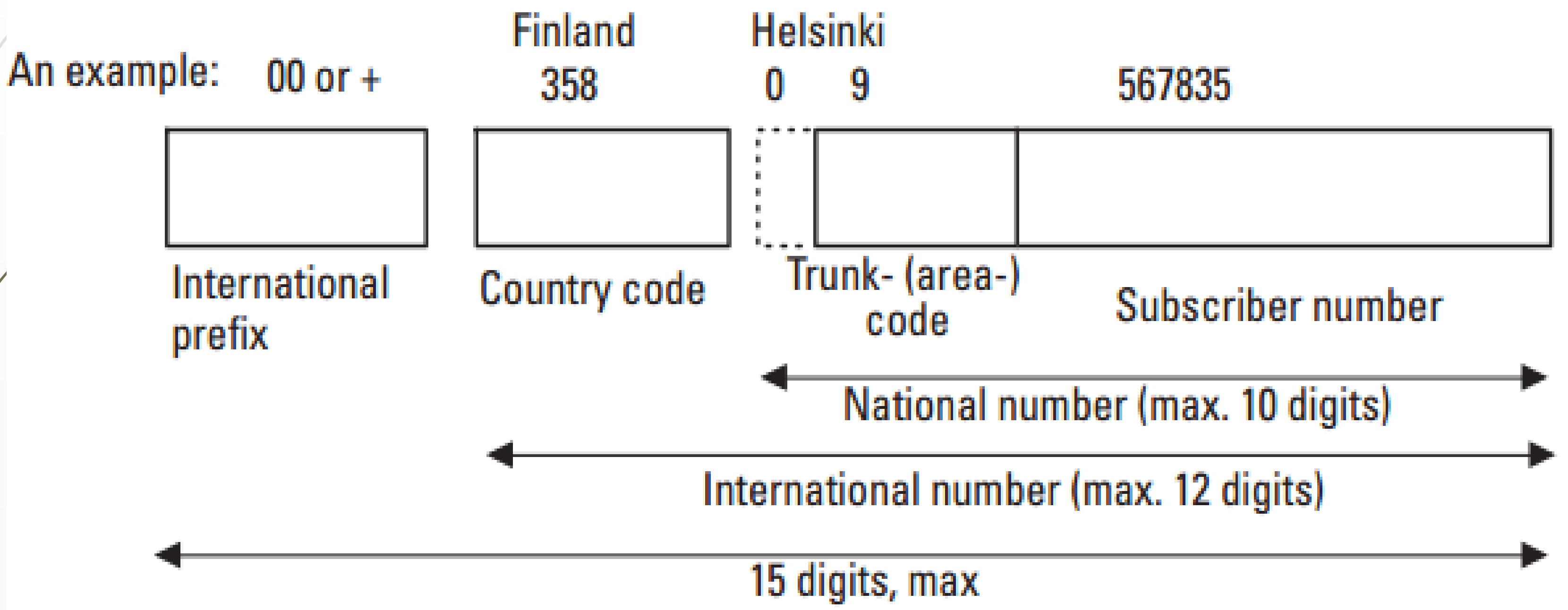

An example:
00 or +
Finland
358
Helsinki
0
9
567835
International prefix
Country code
Trunk- (area-) code
Subscriber number
National number (max. 10 digits)
International number (max. 12 digits)
15 digits, max

Quiz(10%)

If we have subscriber A and subscriber B, if subscriber A want to call for subscriber B and when subscriber A is off hook condition at that time subscriber B was off hook what will be going on

1. Subscriber A
2. Exchange
3. The local loop
4. Billing
5. Ring tone and ring signal

ANY DOUBT AND COMMENT

???

WELCOME

CHAPTER-III

PUBLIC SWITCHED TELEPHONE NETWORK (PSTN) *AND* PUBLIC LAND MOBILE NETWORKS

OUTLINE

- PSTN
- Mobile to land call
- Land to mobile call
- Theory of traffic engineering

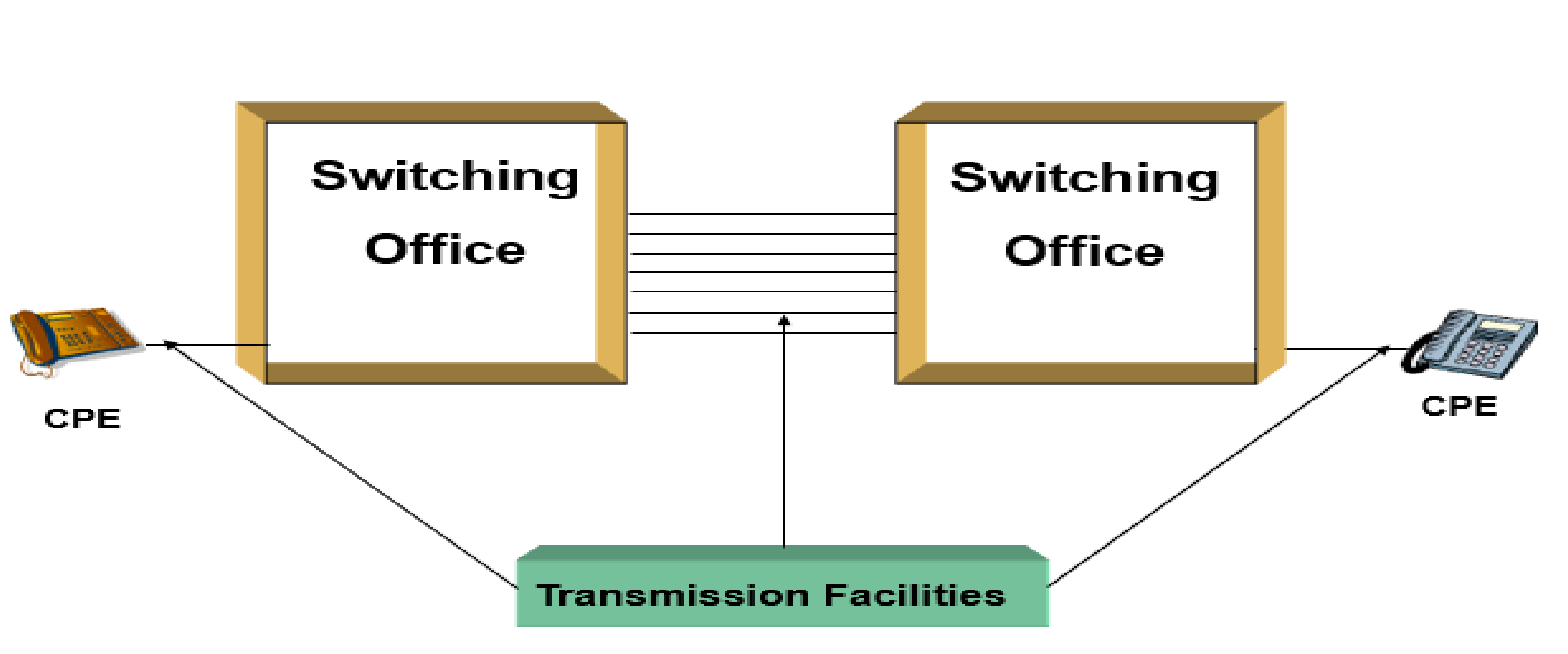
PSTN

- *PSTN* is the worldwide collection of interconnected public telephone network that was designed primarily for analog telephone.
- It is Circuit switching network.
- A dedicated circuit is established for the duration of a telephone call.
- It uses signaling number 7.SS7 as a signaling protocol.
    - In telecommunication signaling is used for controlling communication
    - SS7 is used to set up and terminate a telephone call

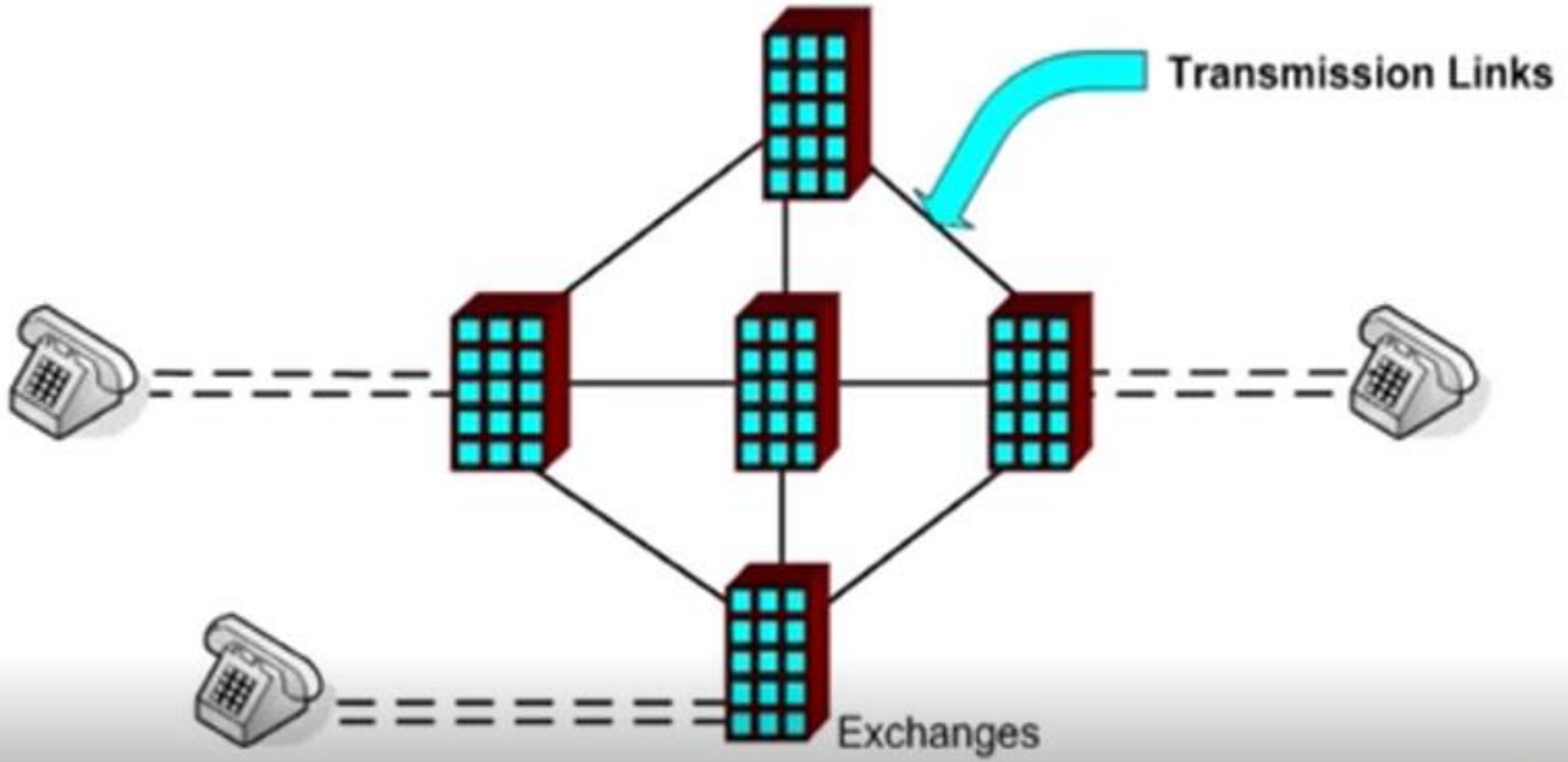
Con…

Major Components of the Public Switched Telephone Network (PSTN):

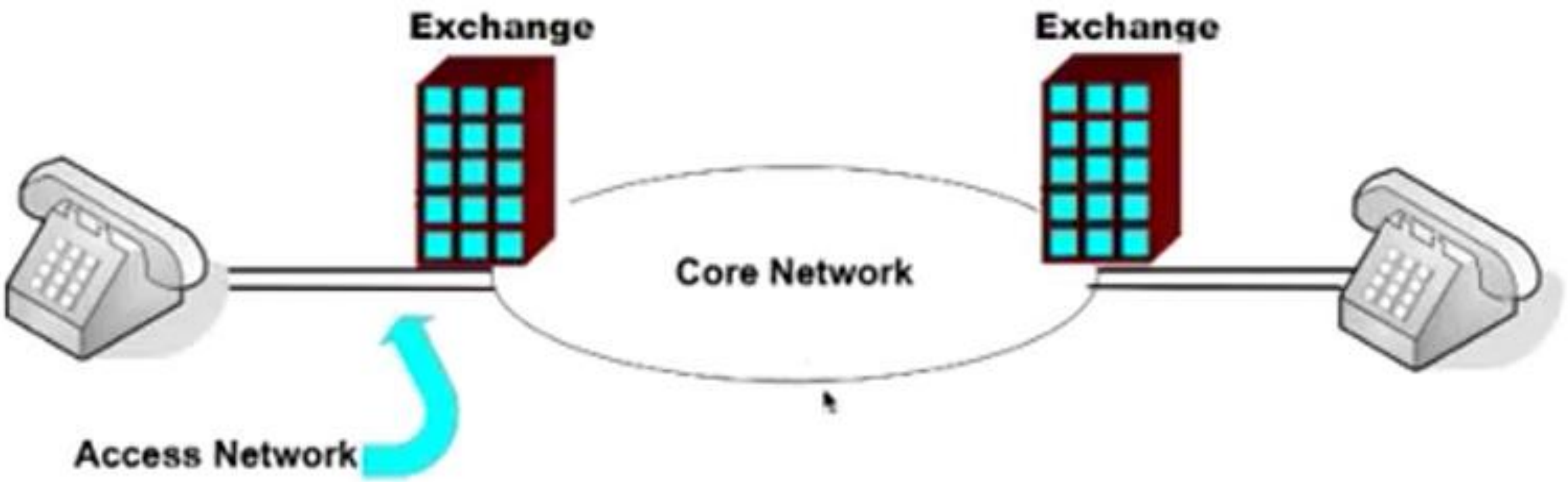
- Switching Offices
- Transmission facilities
- Customer Premise Equipment (CPE)



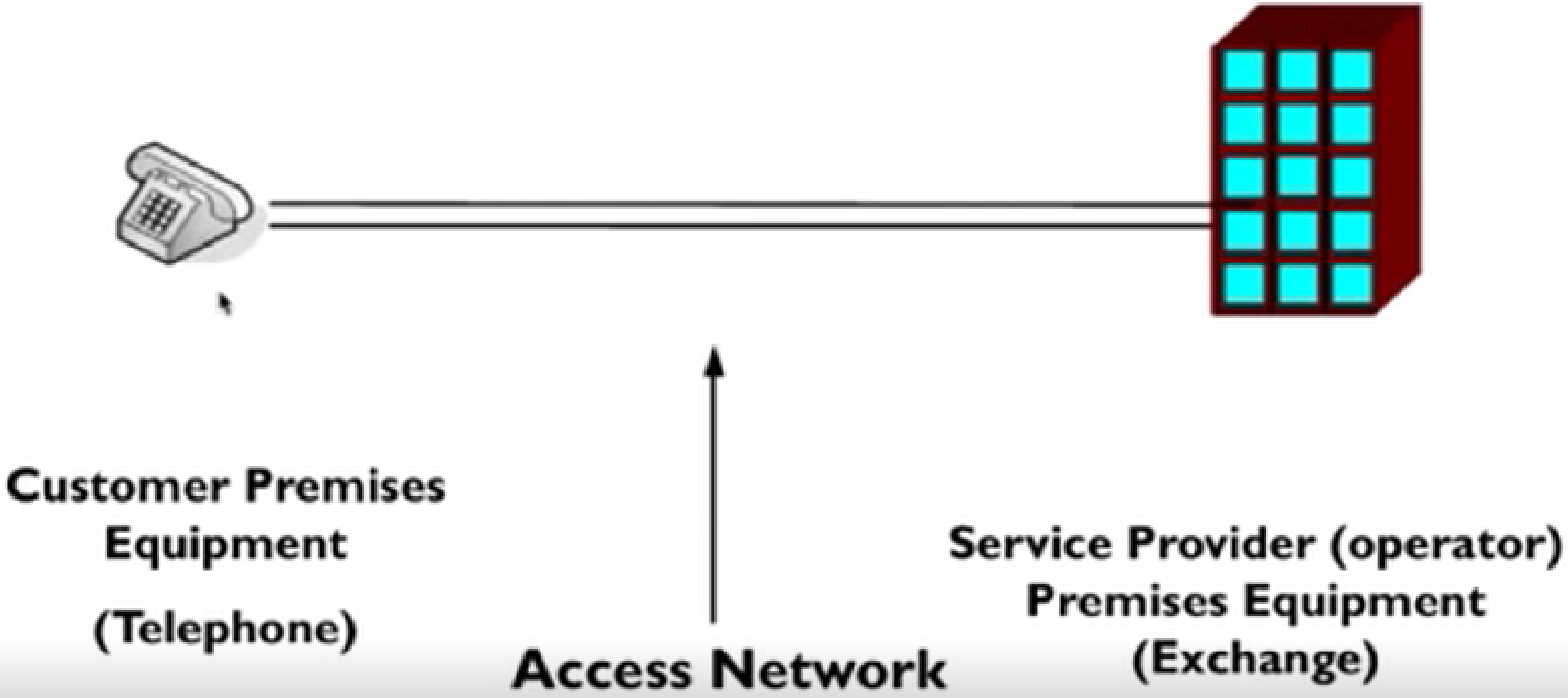
Public Switched Telephone Network (PSTN)



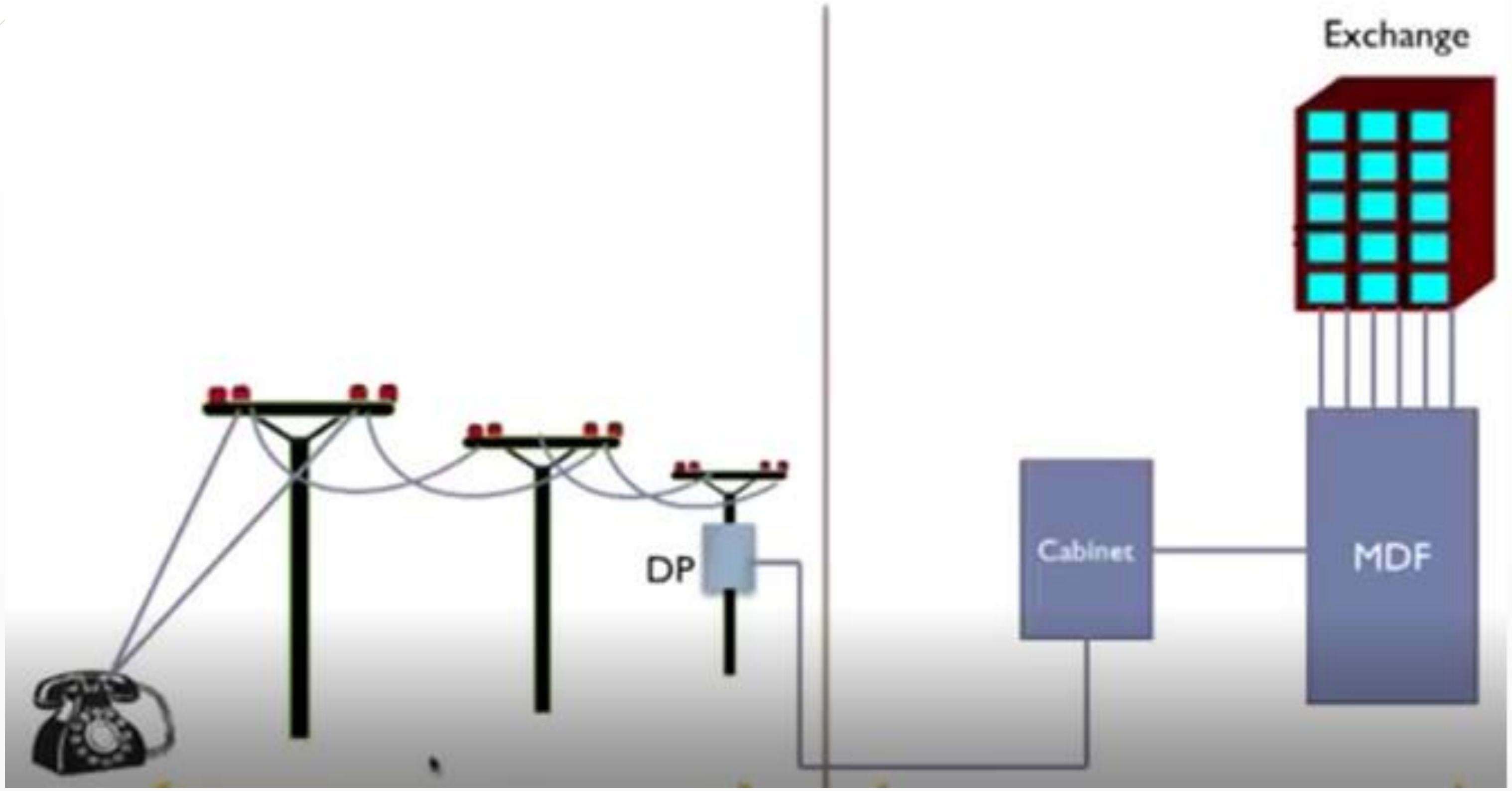
PSTN



Access Network



PSTN NETWORK



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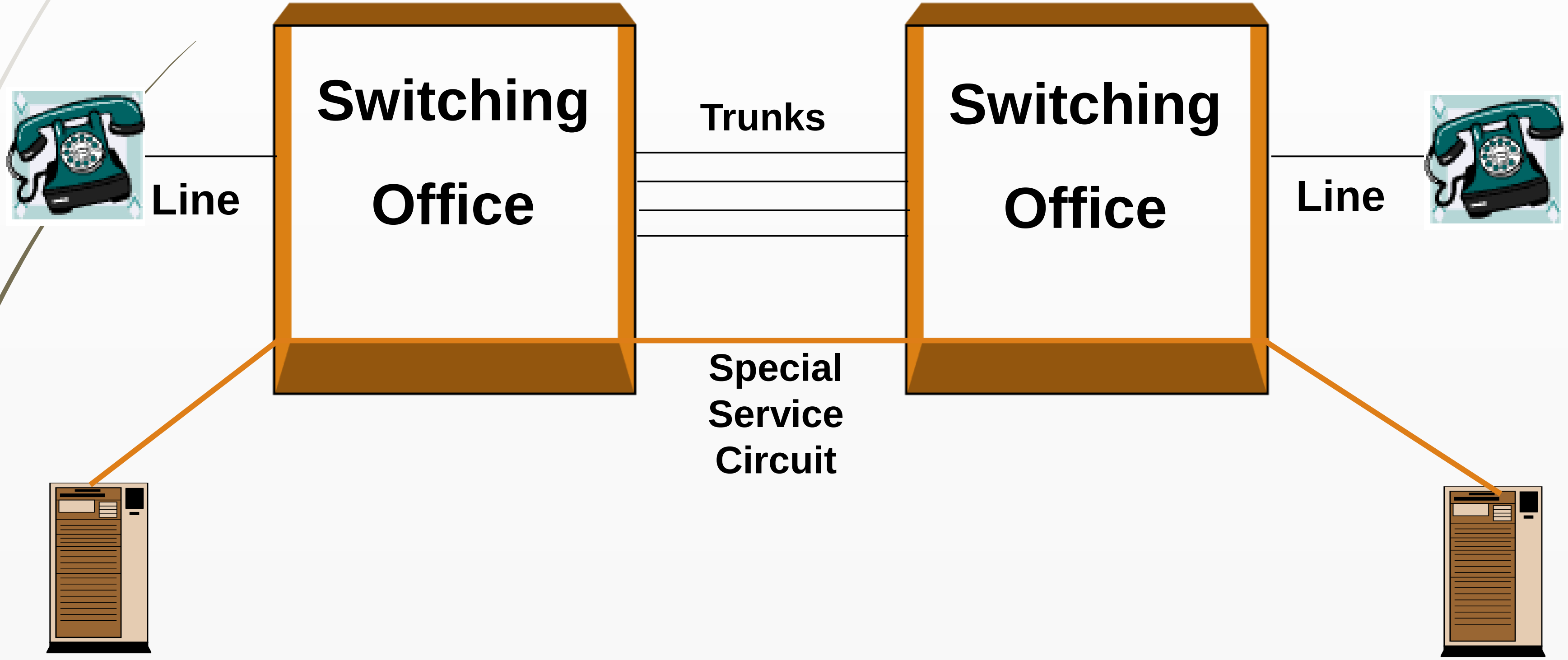
Switching
Office
Trunks
Switching
Office
Line
Line
Special
Service
Circuit

Introduction to channels on network

- Signaling information sent to the mobile is on the **Forward Control Channel (FCC)**.
- Signaling information sent to the cell site is on the **Reverse Control Channel (RCC)**.
- There is also a **Forward Voice Channel (FVC)** for voice communication and a **Reverse Voice Channel (RVC)**.

CONTROL CHANNEL FUNCTIONS

- Each carrier (non-wireline, wireline) have several channels allocated for control purposes. One of these channels is assigned to each cell site. A control channel is on the air 24 hours a day, 365 days a year.
- The control channel actually combines three functions into one channel:
  - **Control Information (Signaling)**
  - **Paging**
  - **Access**
- This is done to reduce the number of channel needed for control purposes to a minimum.

Forward Control Channel (FOCC)

- ❖ The information that is continuously transmitted is on the Forward Control Channel (FOCC).
- ❖ Continuous transmission is needed here because the mobile phone monitors the level of the received FOCC to determine when they approach the edge of the cell and needs to be passed to an adjacent cell base station.
- ❖ There are three types of messages that are multiplexed in this channel:

Con…

Overhead Messages	These are messages that are intended for all mobiles. Information in this message includes:- ➢ The cellular system's SID(System Identification Number), ➢ Combined paging and access channel information, ➢ Subscriber registration information.
Mobile Control Messages	These messages contain information for a specific mobile. ➢ This includes paging data, ➢ Voice channel assignment, ➢ Power level, or other commands that cause a mobile to respond.
Control Filler Messages	➢ These are messages that are transmitted when there are no other messages to be transmitted on the FOCC. ➢ This maintains the mobile's synchronization to the Overhead Message Train. ➢ This message can also be used to tell the mobiles what power level to use when accessing the system on the Reverse Control Channel.

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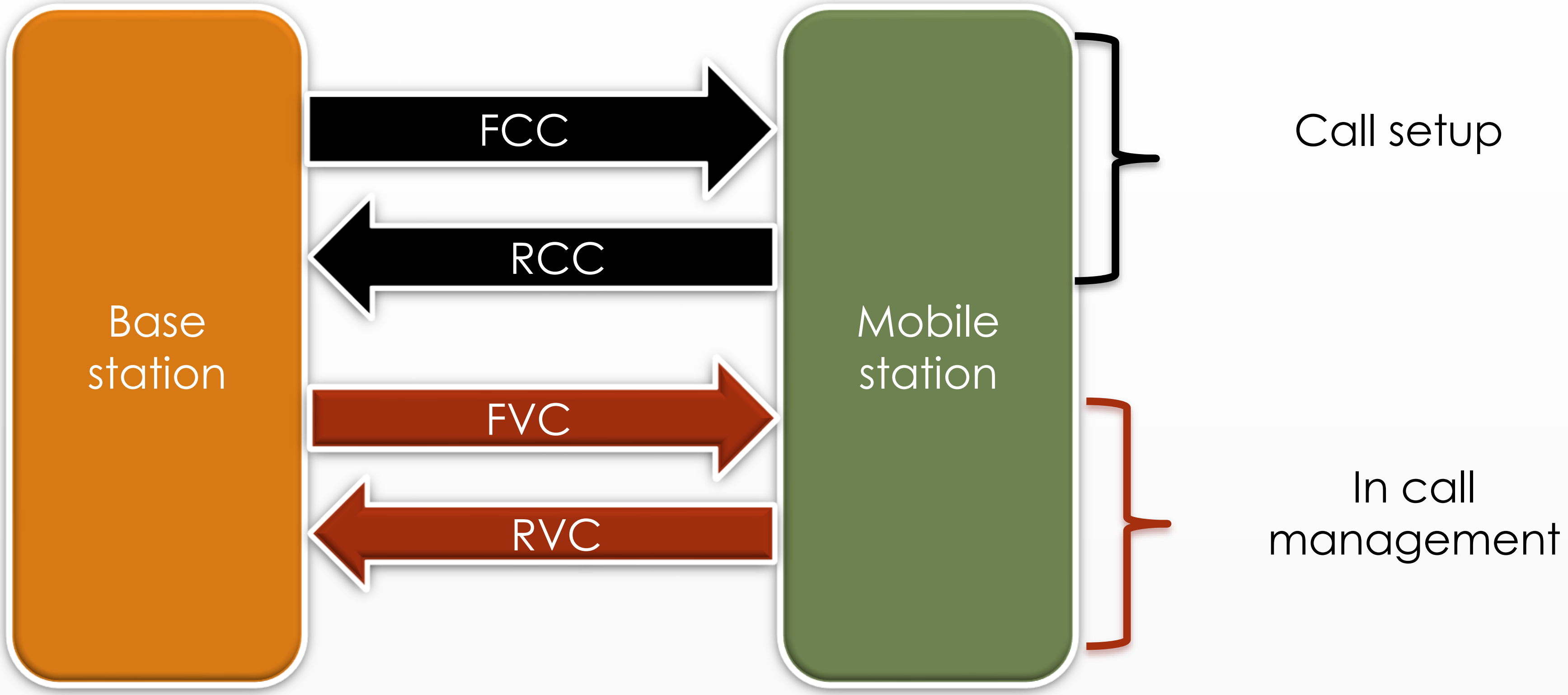
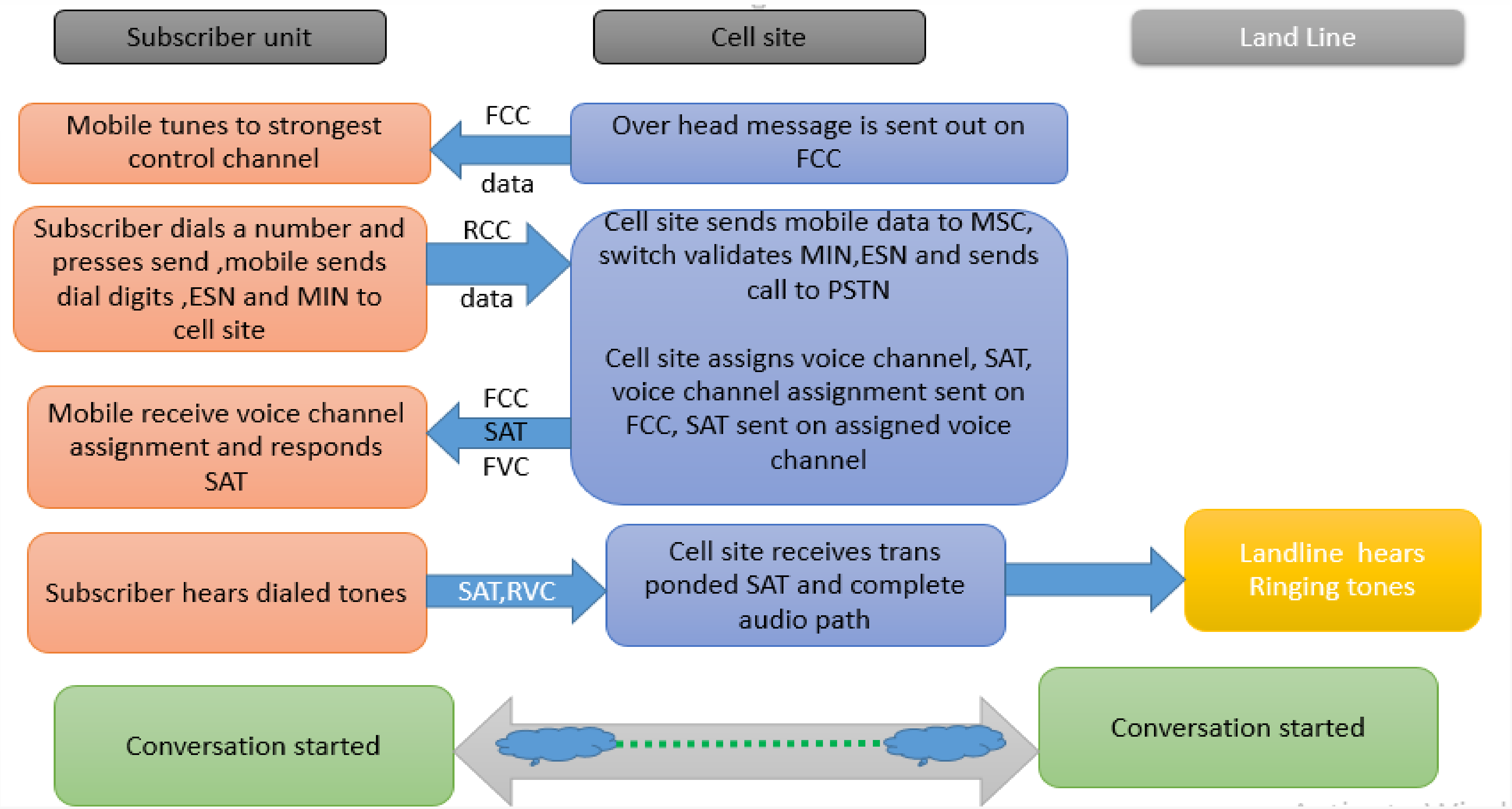

Base station
FCC
RCC
FVC
RVC
Mobile station
Call setup
In call management

Mobile to land line communication



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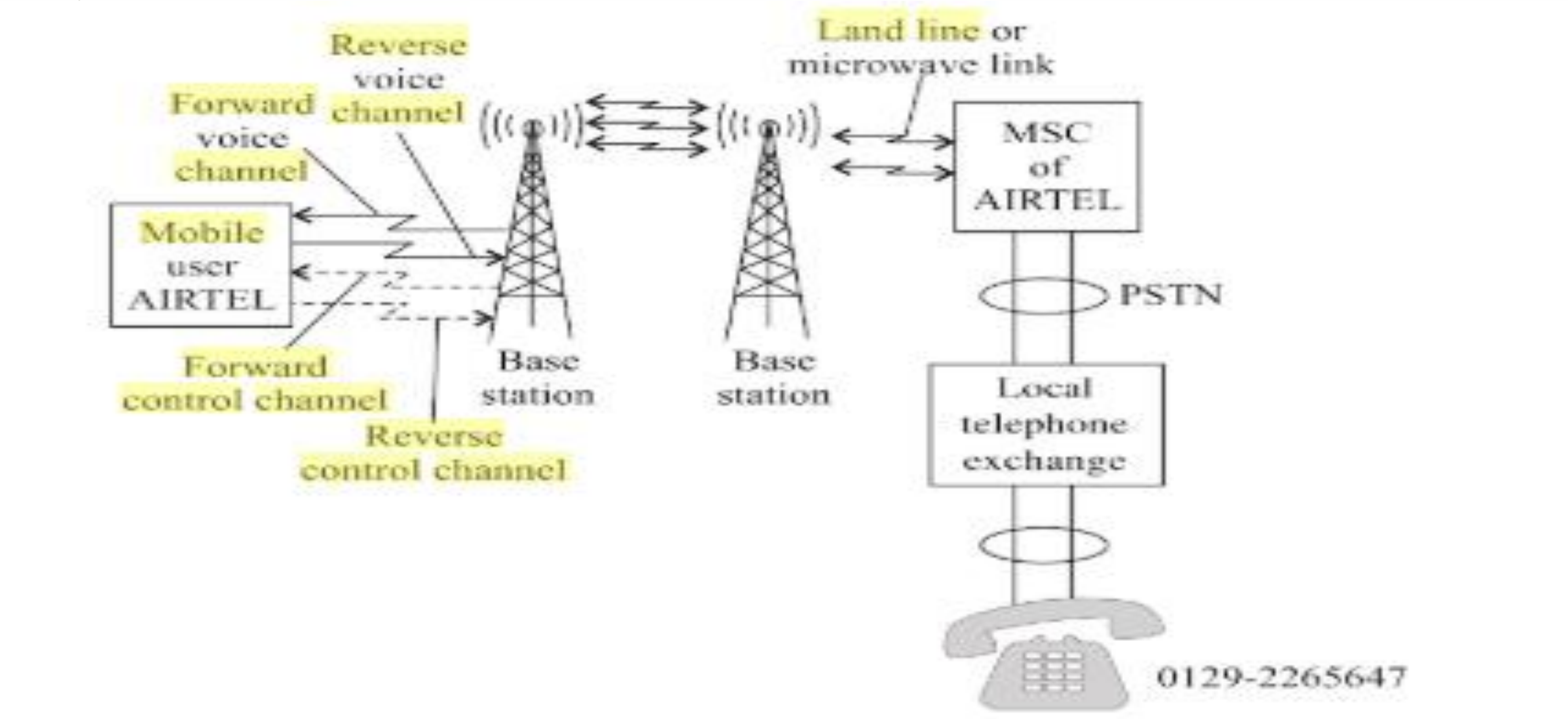
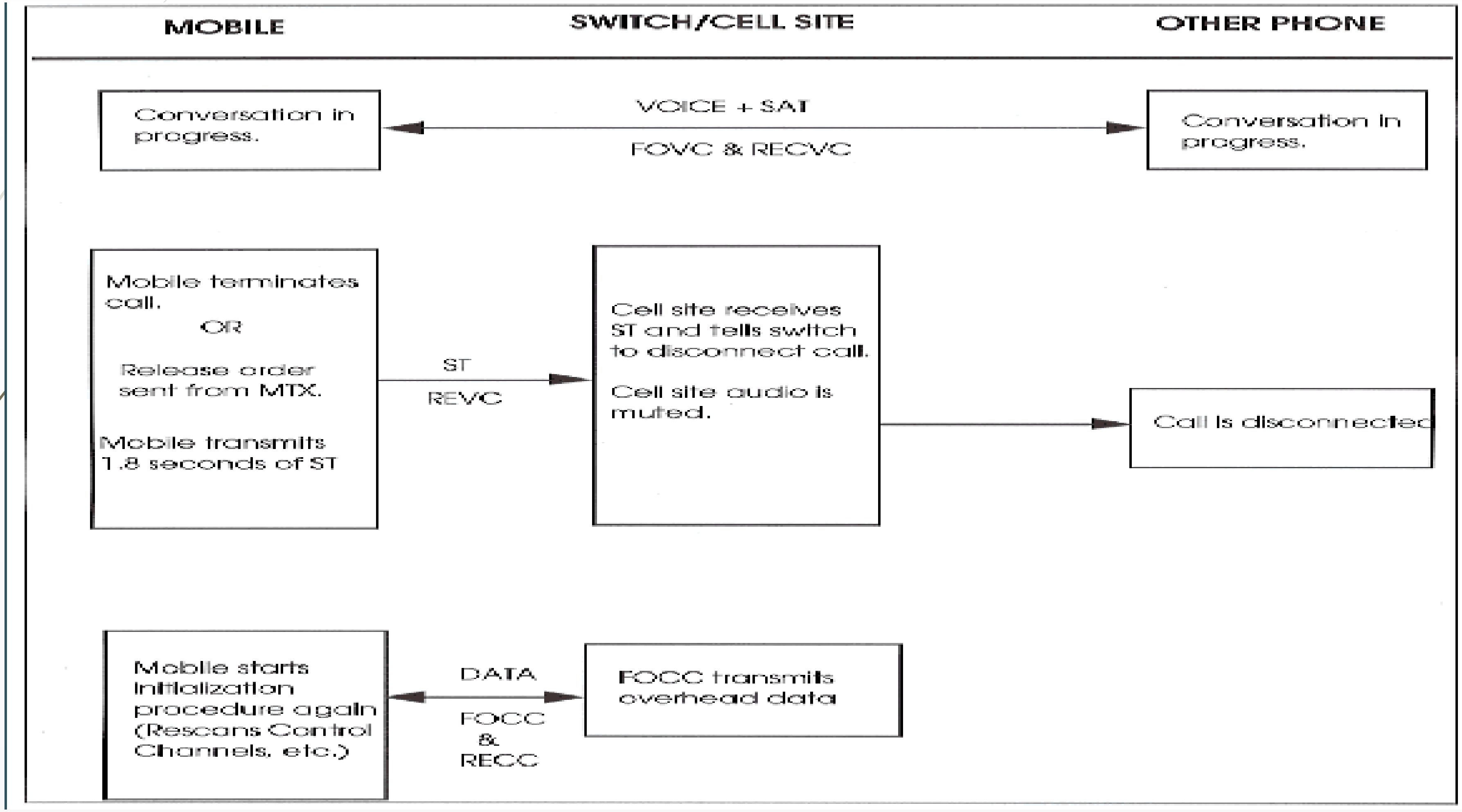
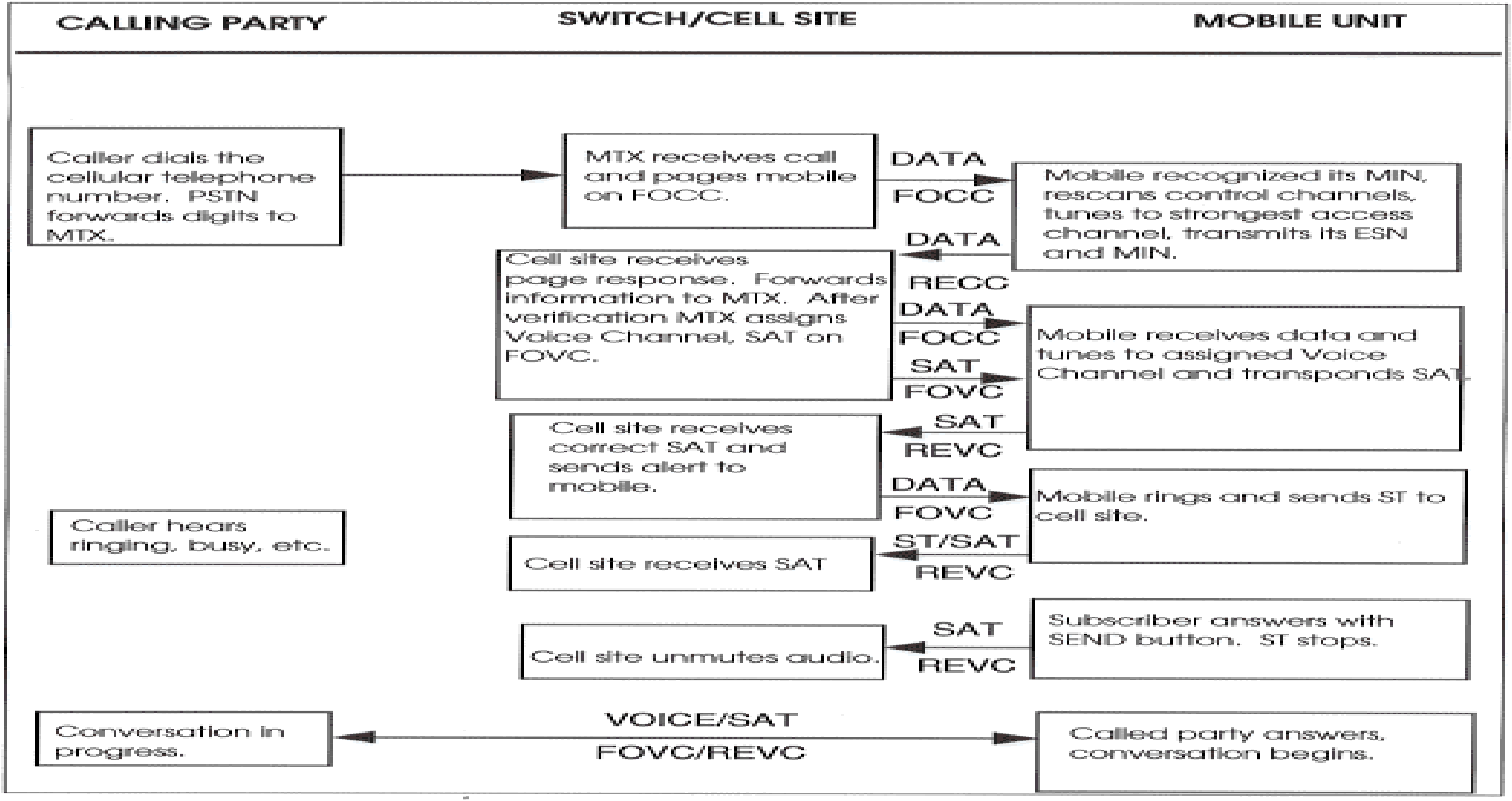

Reverse voice channel
Forward voice channel
Land line or microwave link
MSC of AIRTEL
Mobile user AIRTEL
PSTN
Forward control channel
Base station
Base station
Local telephone exchange
Reverse control channel
0129-2265647

Call Termination Flow Chart



Land to Mobile Call



Theory of traffic Engineering

- How to accommodate a large number of users in a limited radio spectrum?
- Sharing a fixed and small number of channels among a large and random user community.
- Each user demands access from a pool of channel infrequently & at random times
  - ✓ A channel is allocated on a per call basis and a channel is returned to the pool up on termination of a call
  - ✓ So a dedicated channel for each user is not required
  - ✓ If "U" be number of users and "C" be number of channels, for any c < u, possibility of more requests than channels
- Trunking exploits statistical behavior of users so that a fixed number of channels accommodate a large, random user.

TRUNKING

- When a user request service and if all channels occupied

1. The user is blocked or denied access to the system
2. In some systems a queue may be used to hold the requesting users until a channel becomes available.

Up on termination of a call the previously occupied channel immediately returned to the pool.

To handle a given capacity at a specific grade of service requires Trunking and queuing theory .

Trunking – Definition of Terms . . .

- Setup time: The time required to allocate a radio channel to a requesting user
  - Users request may be blocked or have to wait
- Blocked Call: A call that cannot be completed at the time of request due to congestion
  - Also called lost call => lost revenue, e.g., pick hours, holidays, …
- Holding time: Average call duration in seconds, denoted H
  - Depends on users and operator's tariff
- Request (or call) rate: Average number of calls per unit time, denoted λ $seconds^{-1}$
  - Typically taken to be at the busiest time of day
  - Depends on type of users community: Office, residential, call center

- Traffic Intensity: A measure of channel time utilization
  - Is the average channel occupancy measured in Erlang, denoted by A
- Load: Traffic intensity across the entire trunked radio system
  - Measured in Erlang
- Erlang: A “unit” of measure of usage or traffic intensity
  - A channel kept busy for one hour is defined as having a load of one Erlang
- Grade of Service (GoS): Measure of congestion (or ability of a user to access a trunked system) during the busiest hour
  - Typically given as likelihood that a call is blocked, called **Erlang B** or
  - The likelihood of a call experiencing a delay greater than a certain amount of time, called **Erlang C**

- Average arrival rate, λ:
  - Average number of MSs requesting service (call request/time)
- Average hold-time, H
  - Average duration of a call (or time for which MS requires service)
- An average traffic intensity offered (generated) by each user

$$\boxed{A_u = \lambda H \qquad (Erlangs)}$$

- Example 1: If a user makes on average two calls per hour, and that a call lasts an average of 3 minutes

$$\boxed{A_u = \frac{2}{60\,\text{min}} 3\,\text{min} = 0.1 Erlang}$$

- The total offered traffic intensity for U users $\boxed{A = UA_u}$
  - Note: A is not necessarily the traffic carried by the trunked system
- In a C channel trunked system, if traffic is distributed equally among channels, then traffic intensity per channel

$$\boxed{A_C = \frac{UA_u}{C} = \frac{A}{C}}$$

- In Example 1, assume that there are 100 users and 20 channels
  - Then A = 100(0.1)= 10 and A_c = 10/20 = 0.5
- Note: A_c is a measure of the efficiency of channels utilization

Modeling of Traffic Flows

- In telecommunications, calls being handled by a switching system are referred to as *traffic*.
  - Analogous to road way traffic. Must know how many lanes are needed, but with little waste.

- The job of a traffic engineer is to balance the trade-off between cost and service.
  - Must be able to handle traffic sufficiently during busy hours.
  - Designs based on historic data.
  - how many trunks are enough?

Principles Governing Network Design

- ***Efficiency*** is defined as the percent of time the server is working (carrying traffic) as opposed to waiting for a call.
  - This is also known as ***occupancy***.

- For a given load, the more the servers, the less blocked or held calls but costy.

- The rate at which traffic arrives is seldom(not often) uniform.

Principles of Network Design

- *More servers* provide good service, but at an excessive cost.
- Too few servers , both service and cost deteriorate, resulting in loss of customers and *decreased agent productivity.*
- The analyst's job is to reach an optimum balance between costs and service.

Classification of Networks

- Networks can be classified as ***Loss Systems***, or ***Delay Systems.***
  - Loss systems :-calls are blocked or rejected without being served.
    - Service is measured by the % of calls blocked.
  - Delay systems :-are queued calls
    - Service is measured by % of calls that are delayed within a certain time interval.

Classification of Networks

- To administer a telecommunications system, 3 variables are required:
  - **Grade of Service (GoS)**
  - **Traffic load**
  - **Quantity of servers**

Classification of Networks

- Grade of Service (GoS) refers to the % of calls that encounter blockage.
- As a business objective, must determine amount of blockage that can be tolerated.
    - If the GoS is too high (zero blockage), many circuits will be underutilized.
    - Too low means too many busy signals encountered.
    - Most business situations call for between 1% and 5% blockage.

Classification of Networks

- ***Traffic Load*** is the amount of traffic during busy hour.
- ***Busy Hour*** –
  - The Average Busy Hour (ABH) is the average of the busy hour over a several day period.
  - Used to determine the capacity needs.
  - If the busy hour of each day varies, the average is called the Average Bouncing Busy Hour (ABBH).

Classification of Networks

- Busy Hour
  - At the busy hour, more blockage will occur,
    - It is not feasible to design circuits to handle the absolute peaks.
    - Therefore, some blockage is expected.
  - If some seasons have busier times than others, the study should be taken then.
    - Between the holidays or registration week at a college.
  - A common definition for seasonal busy hour is the average amount of traffic during the 10 highest days of the year.

Classification of Networks

- Traffic Load can be distinguished between ***Offered load* vs *Carried load.***
  - ❖ **Carried Load** is what is shown on Billing Reports
    - ➢ ***Talk time.***
  - ❖ **Offered Load** also includes setup times.
    - ➢ Talk Time + Setup Time
    - ➢ Call setup time is the time used for signaling and ringing, as the circuit is occupied during that time, but no billing is recorded.

Classification of Networks

- The Traffic Load is the amount of traffic during busy hour.
- Once engineers know the Traffic Load and the required service level (amount of acceptable blockage, also called GoS), the number of circuits required could be found by looking at a traffic design table.
  - There are several type of traffic design tables, for specific characteristics.

Traffic Design Tables

- 3 types of tables are used, depending on the given situation. Each table uses a different distribution theorem
  - ***Erlang A table*** and assumes Block Calls *Held* (BCH), which assumes that callers immediately redial upon receiving a busy.
  - ***Erlang B table*** and assumes Block calls *Cleared* (BCC), which assumes the caller either waits longer to redial or does not redial.
  - ***Erlang C table*** and assumes Block called *Delayed* (BCD), in which the calls are queued until a circuit becomes available.

Trunking – Erlang B Formula

- The probability of an arriving call being blocked is:

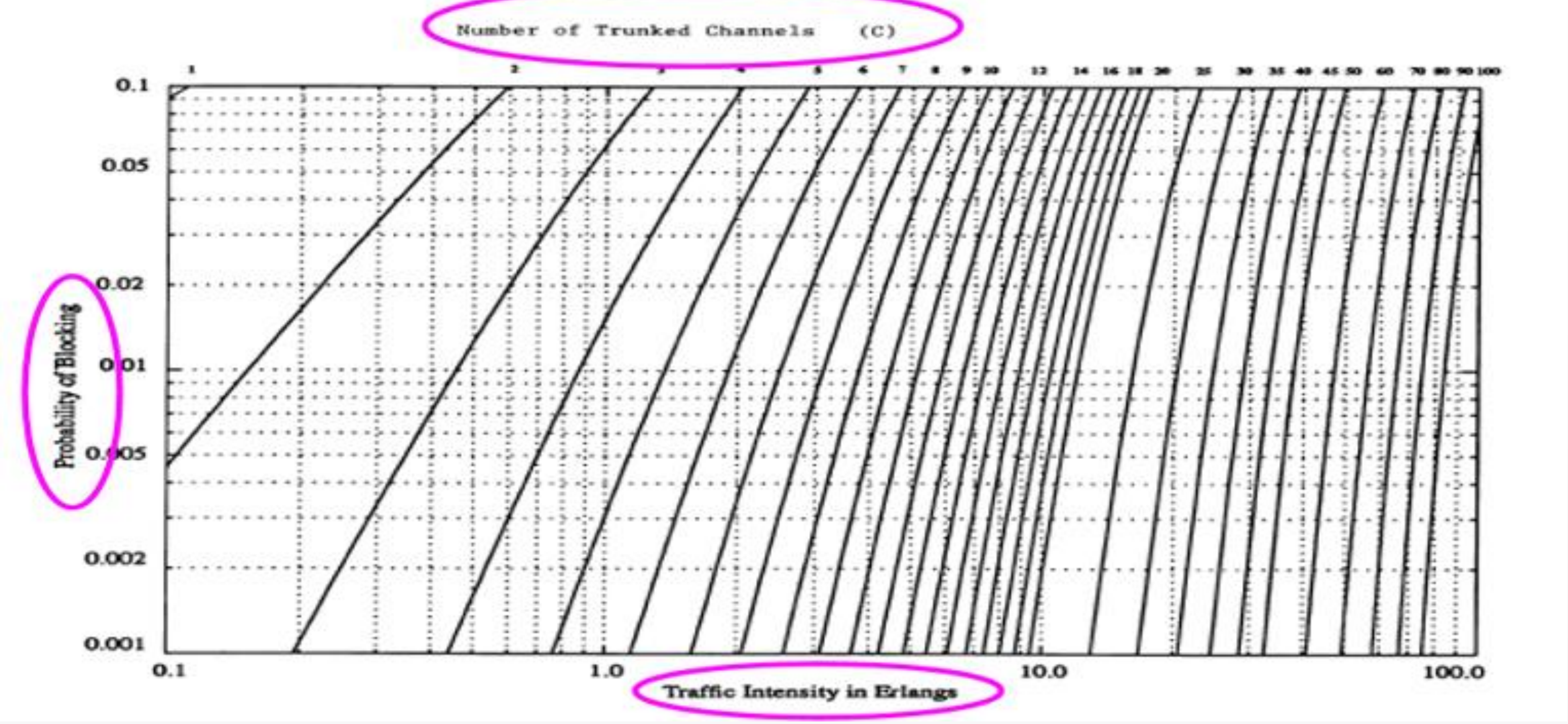
$$\boxed{P_r[blocking] = \frac{\frac{A^c}{C!}}{\sum_{k=0}^{C} \frac{A^k}{k!}} = \mathrm{GOS}}$$

For k event

  - Where C: number of trunked channels and A: total offered traffic
- Erlang B is a measure of the GOS for a trunked system which provides no queuing for blocked calls
- Setting the desired GOS, one can derive
  - Number of channels needed
  - The maximum number of users we can support as $A = UA_U$ or
  - The maximum A_U we can support (and set the number of minutes on our calling plans accordingly)
- Since C is very high, it is easier to use table or graph

Con…

Erlang B Formula - Graphical Form



- Traffic Usage is measured in CCS or Erlang.
  - Erlang equals 1 hour of traffic usage.
    - Can be converted to minutes by multiplying by 60.
  - CCS (Centi* Call Seconds) is used for smaller increments of measurement.
    - 1 CCS is worth 100 seconds.
    - C is the Roman Numeral for 100
- 1 Erlang=60 minutes=3600 seconds=36 CCS, 1 CCS=1.67 minutes.
- Knowing desired GoS and traffic load, the number of circuits is found using traffic tables.

Example: using the first type of table, we state that 1% blocking is acceptable, and 4 Erlangs of traffic are measured during the ABBH, 10 trunks are needed. If 5% blocking is acceptable, only 8 trunks are needed.

	$P = 0.01$		$P = 0.05$	
Trunks	Erlangs	CCS	Erlangs	CCS
1	0.01	0.4	0.05	1.8
2	0.15	5.4	0.38	13.7
3	0.46	16.6	0.90	32.4
4	0.87	31.3	1.52	54.7
5	1.36	49	2.22	79.9
6	1.91	68.8	2.97	107
7	2.50	90	3.75	135
8	3.14	113	4.53	163
9	3.78	136	5.36	193
10	4.47	161	6.22	224
11	5.17	186	7.08	255
12	5.89	212	7.94	286

Trunking – Blocked Calls Delayed

- Instead of clearing a call, put it in a queue and have it wait until a channel is available
  - First-in, first-out line: Calls will be processed in the order received
- There are two things to determine here
  1. The probability a call will be delayed (enter the queue), and
  2. The probability that the delay will be longer than t seconds
- The first is no longer the same as Erlang B
  - It goes up, because blocked calls aren't cleared, they "stick around" and wait for the first open channel
- Meaning of GOS
  - The probability that a call will be forced into the queue AND it will wait longer than t seconds before being served (for some given t)

Trunking - Blocked Calls Delayed …

- Additional assumptions:
  1. The queue is infinitely long: Translates to infinite memory
  2. No one who is queued gives up/hangs up (rather than wait)
- The probability of an arriving call not having an immediate access to a channel (or being delayed) is given by Erlang C Formula

$$\boxed{P_r[delay > 0] = \frac{A^c}{A^c + C!\left(1 - \frac{A}{C}\right)\sum_{k=0}^{C-1} \frac{A^k}{k!}}}$$

- It is typically easiest to find a result from a chart

Trunking - Calls Delayed …

- Once it enters the queue, the probability that the delay is greater than *t* (for *t* > 0) is given as

$$\boxed{P_r[delay > t \mid delay > 0] = \exp\left(-\frac{C-A}{H}t\right)}$$

- GOS: The marginal (overall) probability that a call will be delayed AND experience a delay greater than t is then

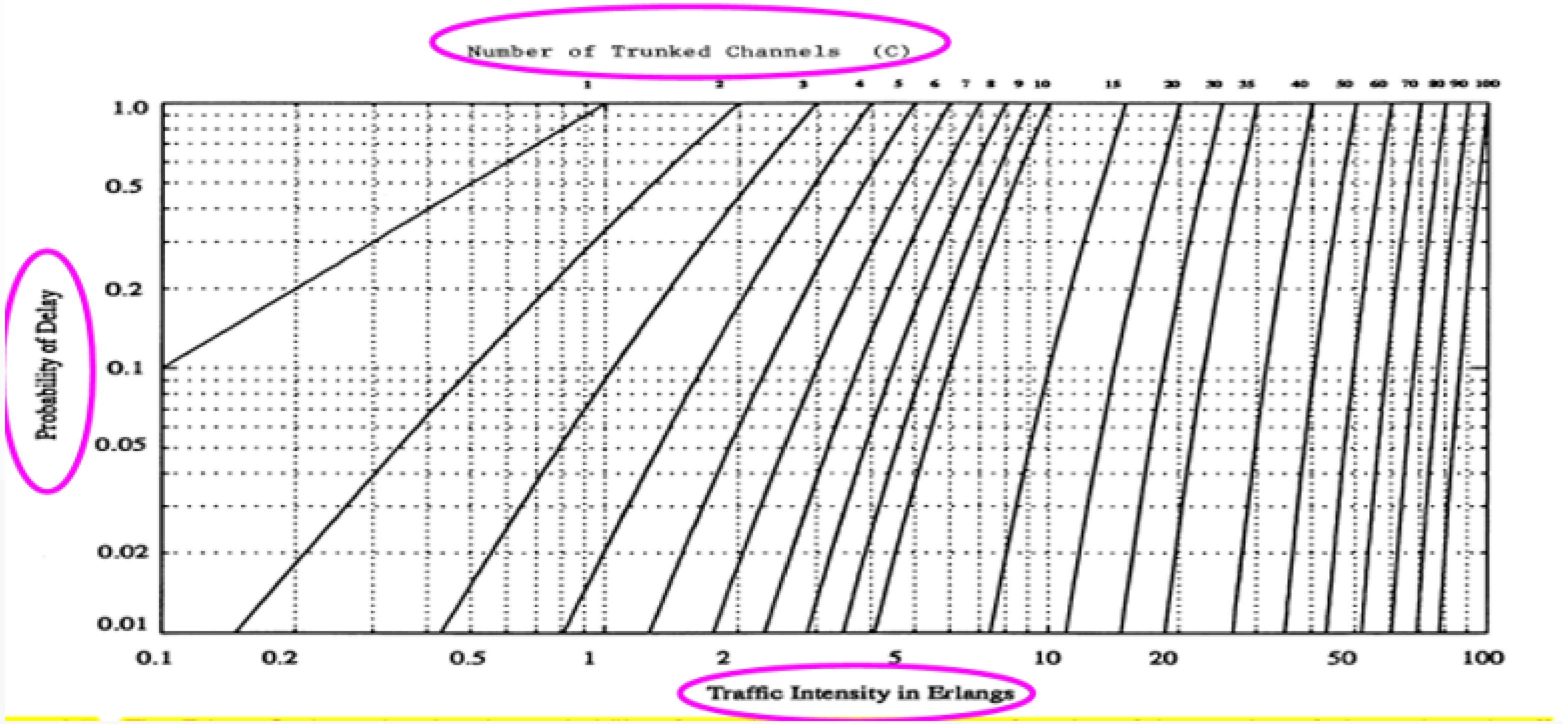
$$\boxed{\begin{aligned} P_r[delay > t] &= P_r[delay > 0]P_r[delay > t \mid delay > 0] \\ &= P_r[delay > 0]\exp\left(-\frac{C-A}{H}t\right) \end{aligned}}$$

- The average delay for all calls in a queued system

$$\boxed{D = P_r[delay > 0]\frac{H}{C-A}}$$

Con…

Erlang C Formula - Graphical Form



Sources of traffic usage information

- Many PBX systems provide traffic usage data.
- Most LEC(local exchange carrier)s can perform a *busy study* on selected trunk groups.
  - shows the number of items that callers attempted and encountered a busy.
  - If no busies are logged, too many trunks have been assigned.
- Call Accounting System – a PC connected to a SMDR (Station Messaging Detail Record) port of the switch provides info.
  - No call setup time info is provided in this report.

Uses of Traffic Modeling

- There are two main uses for Traffic Modeling:
  - Performance Analysis
    - Concerned with questions such as delay, throughput, packet loss.
  - Network Engineering and Management:
    - Concerned with questions such as capacity planning, traffic engineering, anomaly detection.

QUESTION PLEASE ???

Chapter -IV

Introduction to digital telecom signaling and applications

overview of SDH, PDH, SONET and ATM

Transmission mode

- Data transmissions may be either asynchronous or synchronous.
- In an asynchronous transmission, only one character is transmitted or received at a time.
- During transmission, there is start bit and followed by a stop bit that lets the receiving device know where a character begins and ends.
- Asynchronous transmission is inherently inefficient due to the additional overhead required for start and stop bits, and the idle time between transmissions.
- It is generally used only for relatively low-speed data transmission.
- In synchronous transmission, a group of characters is sent over a communications link in a continuous bit stream, while data transfer is controlled by a timing signal initiated by the sending device.
- The sender and receiver must be in perfect synchronization to avoid the loss or gain of bits.
- Synchronous transmission is generally used for transmitting large volumes of data at high speeds.

Plesiochronous Digital Hierarchy (PDH)

- Greek: Plesio- Close, chronous-time

=> almost synchronous but not completely synchronous.

- PDH is a technology for transporting voice or data between multiple devices and which are working with clock sources with accepted tolerance levels for synchronization.
- PDH was built for Digital transmission of signals.
- Pulse code Modulation(PCM) is the technique used in PDH networks which is based on the Time Division Multiplexing(TDM).
- Main mode of transmission was Twisted pair, co-axial cable and Microwave

Plesiochronous Digital Hierarchy (PDH)

- ITU-T Recommendation G.702 defines a time-multiplex structure based on 64 Kbit/s channels for the basic bit rates of 2.048 Mbit/s in E1 and 1.544 Mbit/s in T1.
- The conversion of voice signals into digital code was always performed at a sampling rate of 8 kHz.
- The analog signal is sampled at intervals of 125 μs, which according to Nyquist is sufficient to digitize all the information contained in a 4 kHz voice channel.
- Because every measured value is coded in 8 bits, the voice channel is transmitted at 64 Kbit/s.

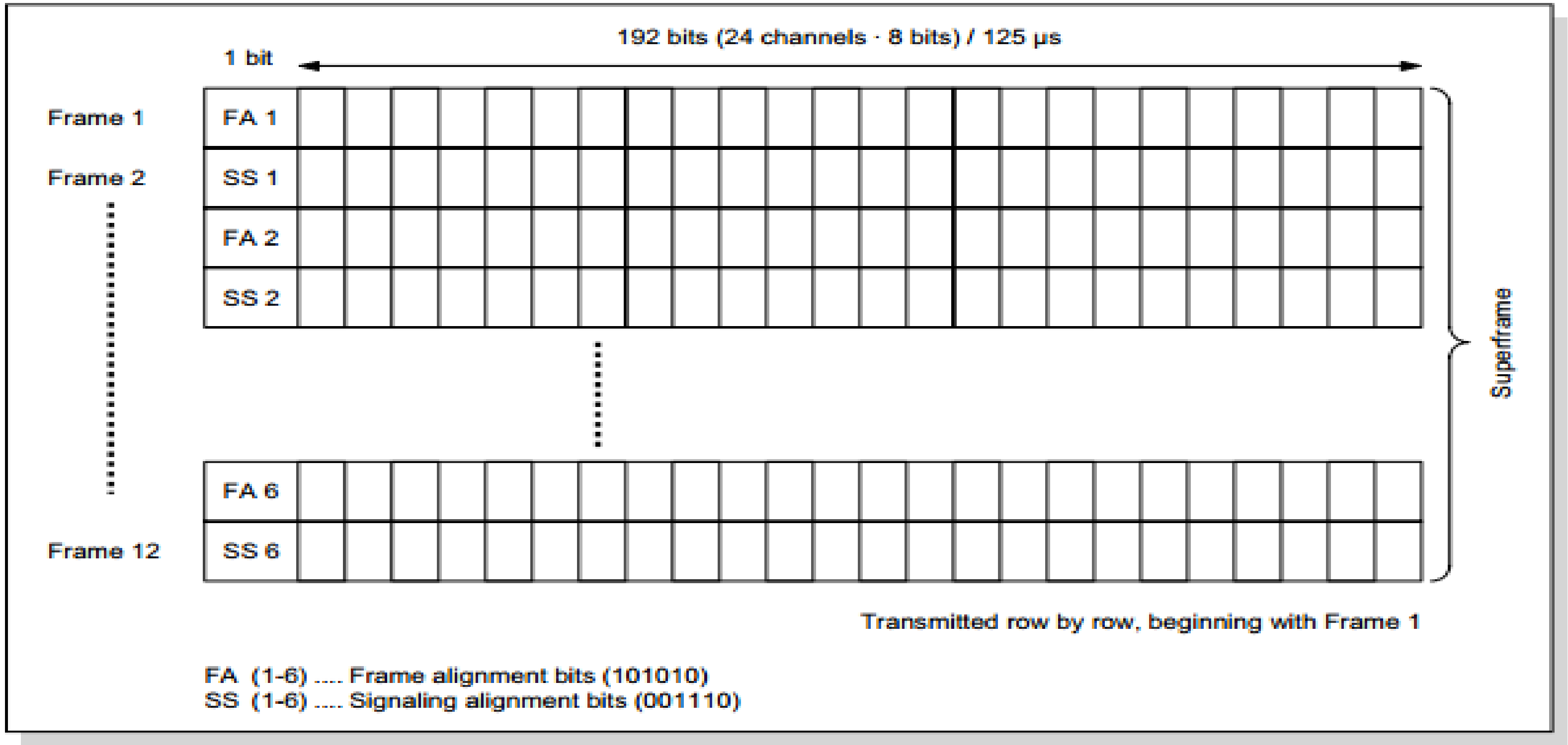
The T1 Interface (Carrying DS1 Signals)

- The North American standard defines a primary rate of 1.544 Mbit/s called T1.
- This provides for the transmission of 24 channels at 64 Kbit/s per channel.
- Note that "**T1" (Transmission level 1**) describes the **electrical signal, **independent of the **frame structure.**
- "**DS1" (Digital Signal level 1**) defines the frame structure carried within T1.
- In practice, the terms tend to be used interchangeably, although strictly speaking the physical interface should be called "T1".
- DS1 signals from T1 interfaces can be multiplexed to higher rate signals (DS2, DS3) etc.

Con…

- Each **DS1 frame is 192 bits long (24 x 8 bits).**
- The addition of 1 bit for frame alignment yields a total of **1.544 Mbit/s (193 bits x 8 kHz).**
- The pattern for frame alignment consists of 6 bits (101010), which are spread out over six frames because each frame carries only one alignment bit.
- The alignment bit is also used to identify the frames containing signaling bits, by means of another 6-bit pattern (001110).
- The alignment bit changes between framing and signal framing, so that each of the two patterns is completed once in every 12 frames.
- A multi-frame sequence of 2,316 bits (12 frames of 193 bits) containing both complete alignment patterns is also referred to as a super-frame.

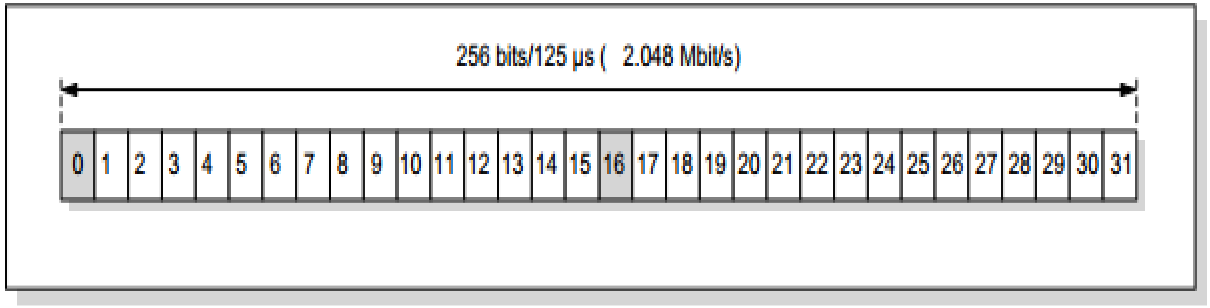
Con…



DS1 super frame structure

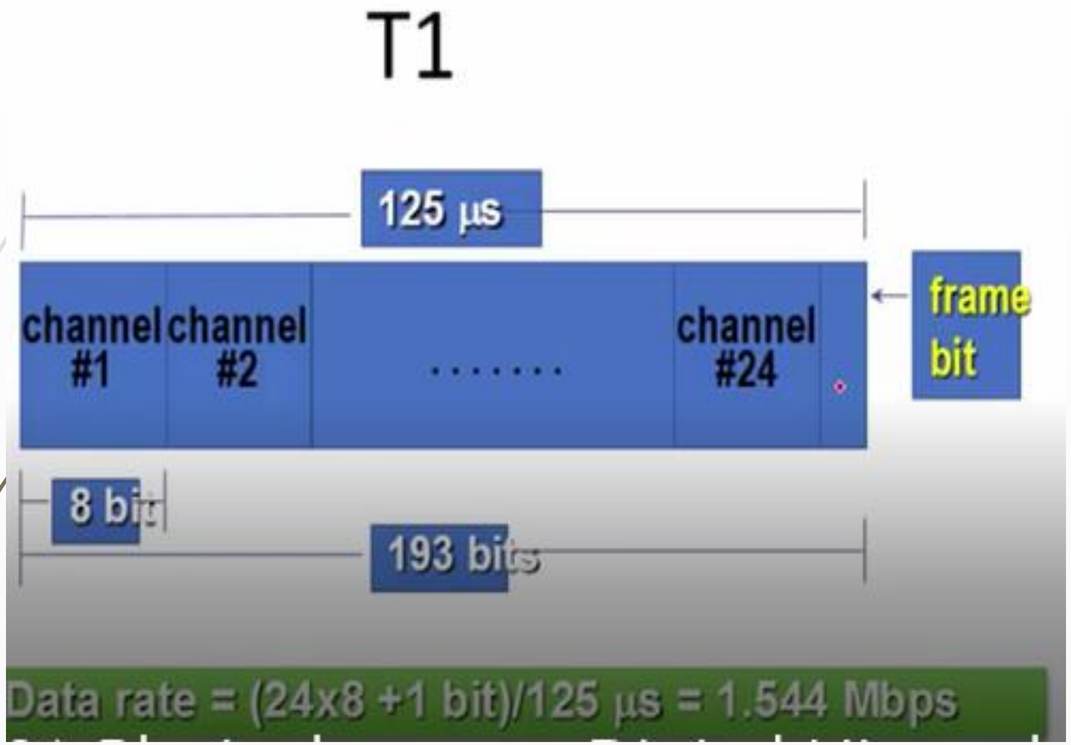
The E1 Interface

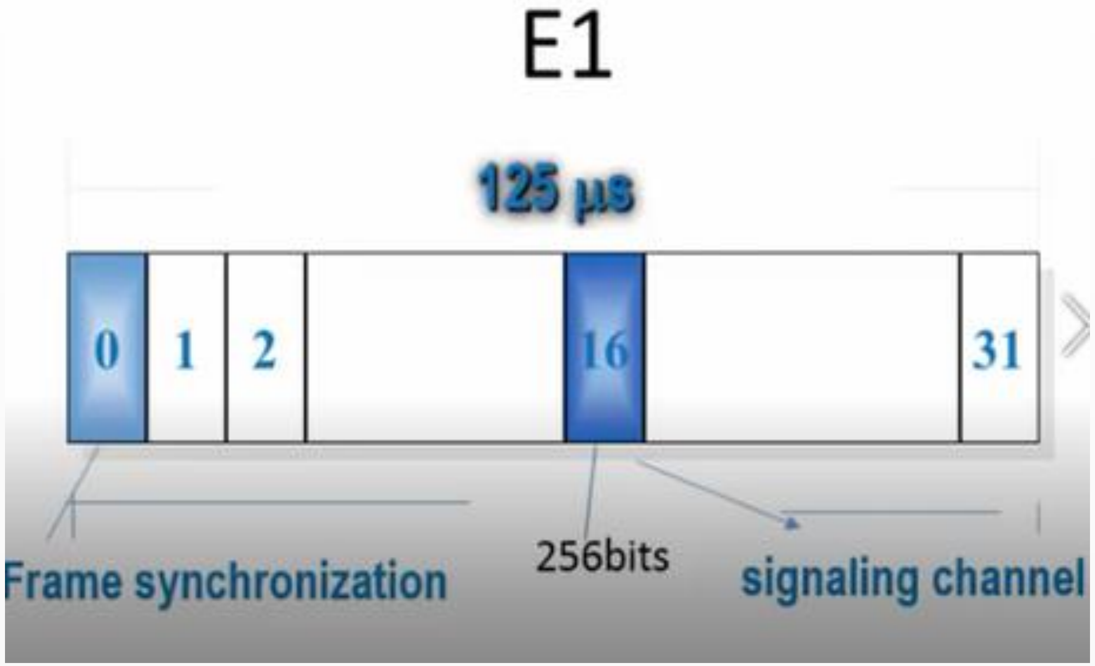
- The **E1** system is based on a **frame structure of 32 x 8 bit "timeslots"** (that is, a total of **256 bits**); the timeslots are numbered 0 to 31.
- Like the **DS1** frame, the **E1** frame repeats every 125 µs; this creates a signal of **2.048 Mbit/s (256 bits x 8 kHz).**
- Because each 8-bit timeslot is repeated at a rate of 8 kHz, it is able to carry a 64 Kbit/s channel.



E1 frame

DS-1 frame in T1 and E1





Con…

- **Timeslot 0** alternates a **frame alignment signal (FAS),** containing an alignment bit pattern, with a "**Not Frame Alignment**" signal (NFAS), **containing error management information.**
- **Timeslot 16 was originally designed to carry signaling information, such as telephone numbers dialed.**
- This leaves 30 payload timeslots (1 to 15, 17 to 31) available in the so-called PCM-30 system. In a PCM-30 system, Timeslot 16 of each frame carries signaling information for two payload channels (4 bits each)

The SONET OC-1 Interface

- The first hierarchical level in SONET is the **Synchronous Transport Signal 1 (STS-1).**
- This is an 810-byte frame that is transmitted at 51.84 Mbit/s and, when transmitted over an optical interface, the resulting signal is known as **Optical Carrier 1 (OC-1).**
- **STS-1** can also exist as an electrical interface, which is called **Electrical Carrier 1 (EC-1),** although this term is rarely used.
- The transmission time of a **STS-1** frame corresponds to the **125 µs** pulse code modulation (PCM) sampling interval;
- Each byte in the SONET signal thus represents a bandwidth of **64 Kbit/s**.
- The frame is divided into nine sub-frames of 90 bytes each

Con…

Section Overhead	Framing A1	Framing A2	Ident J0/Z0	Trace J0	Path Overhead
	BIB-8 B1	Orderwire E1	User F1	BIP-8 B3	
	Datacom D1	Datacom D2	Datacom D3	Signal Label C2	
Line Overhead	Pointer H1	Pointer H2	Pointer Action H3	Path Status G1	
	BIP-8 B2	APS K1	APS K2	User Channel F2	
	Datacom D4	Datacom D5	Datacom D6	Indicator H4	
	Datacom D7	Datacom D8	Datacom D9	Growth Z3	
	Datacom D10	Datacom D11	Datacom D12	Growth Z4	
	SyncStat S1	REI-L M0	Orderwire E2	Tandem Connection Z5	

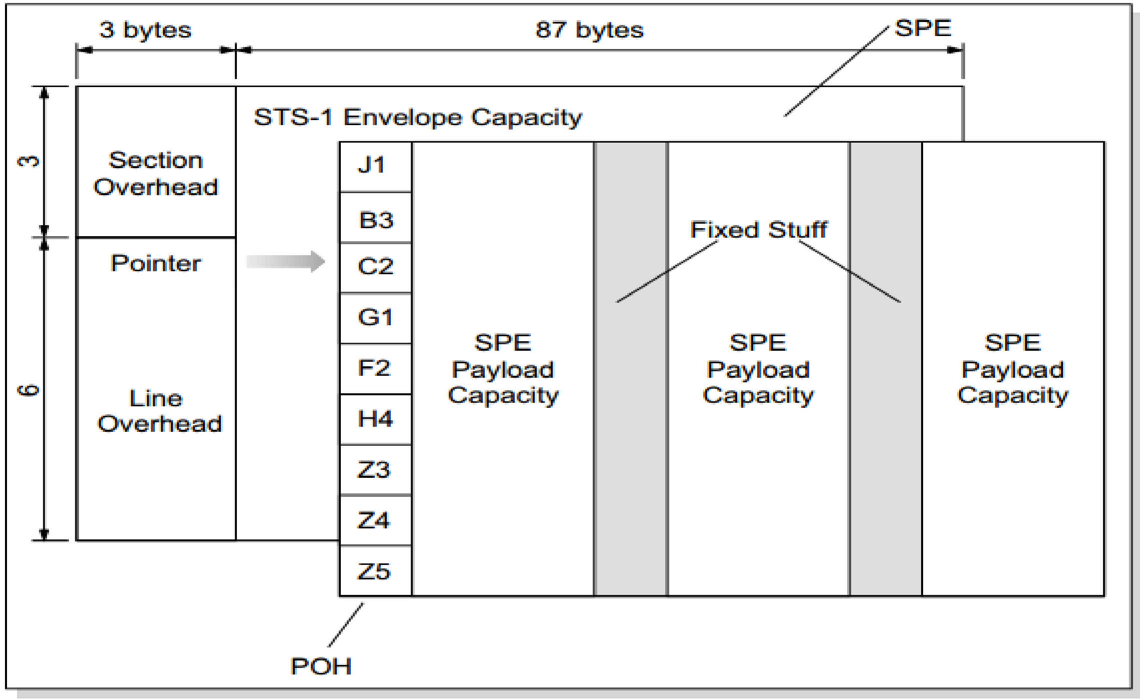
STS-1 Transport Overhead (TOH) and Path Overhead (POH)

Con…

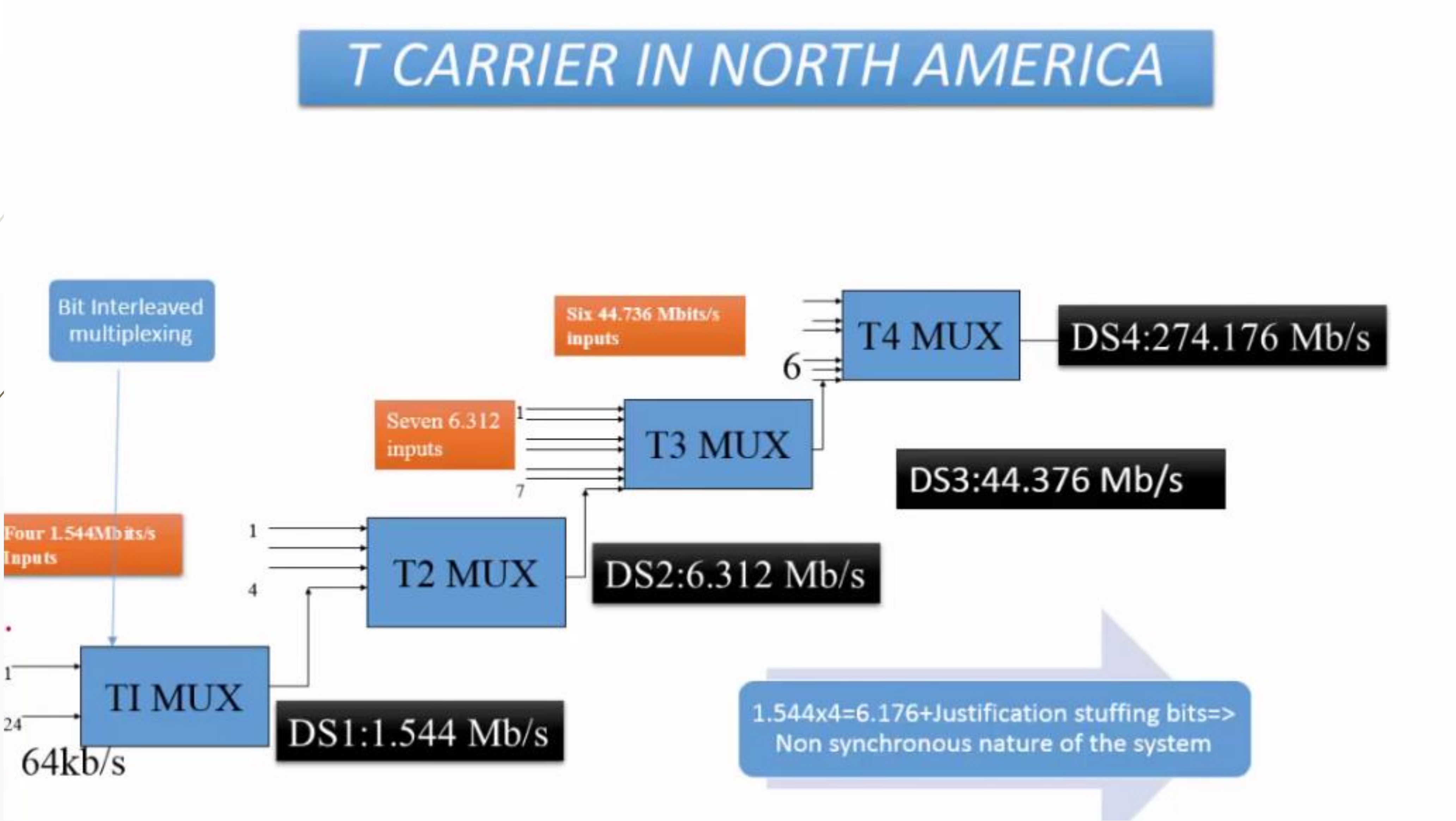
- The first **3 bytes** of each sub-frame comprise 3 bytes of the 27 (that is, 9 x 3) byte **Transport Overhead (TOH).**
- The remaining **87 bytes** of each sub-frame are occupied by 87 bytes of the 783 (that is, 9 x 87) byte **Synchronous Payload Envelope (SPE).**
- The **27 TOH** bytes **control** the transport of user data between neighboring network nodes, and contain information required for the **transport section.**
- The TOH is divided into two parts, the **Section Overhead and the Line Overhead.**
- The TOH bytes A1, A2, J0/Z0, B1, E1, F1 and D1 through D3 comprise the **Section Overhead**,
- And bytes H1, H2, H3, B2, K1, K2, D4 through D12, S1/Z1, M0, E2 form the **Line Overhead**.

Con…

- The **SPE**, also a structure of 783 bytes, is located in the 9 x 87 byte Envelope Capacity (frame payload area). The first column of this is occupied by the **POH** and a further two columns (30 and 59) are reserved for "**fixed stuff**". This leaves **84 columns of "Payload Capacity" for carrying user traffic**.



STS-1 frame


T CARRIER IN NORTH AMERICA
Bit Interleaved multiplexing
Six 44.736 Mbits/s inputs
T4 MUX
DS4:274.176 Mb/s
6
Seven 6.312 inputs
1
T3 MUX
7
DS3:44.376 Mb/s
Four 1.544Mbits/s Inputs
1
T2 MUX
4
DS2:6.312 Mb/s
1
TI MUX
24
64kb/s
DS1:1.544 Mb/s
1.544x4=6.176+Justification stuffing bits=>
Non synchronous nature of the system

PDH BIT RATES

(European standard)

- E1- 2048 Kbps (2Mb) [30 Voice Channel]
- E2- 8448 Kbps (8Mb) [120 Voice Channel]
- E3- 34368 Kbps (34Mb) [480 Voice Channel]
- E4- 139264 Kbps (140Mb) [1920 Voice Channel]

PDH HIERARCHY(E CARRIER) IN EUROPE

PCM 30 Mux (D1 Level)

32 * 64 KHz = 2.048 Mb/s

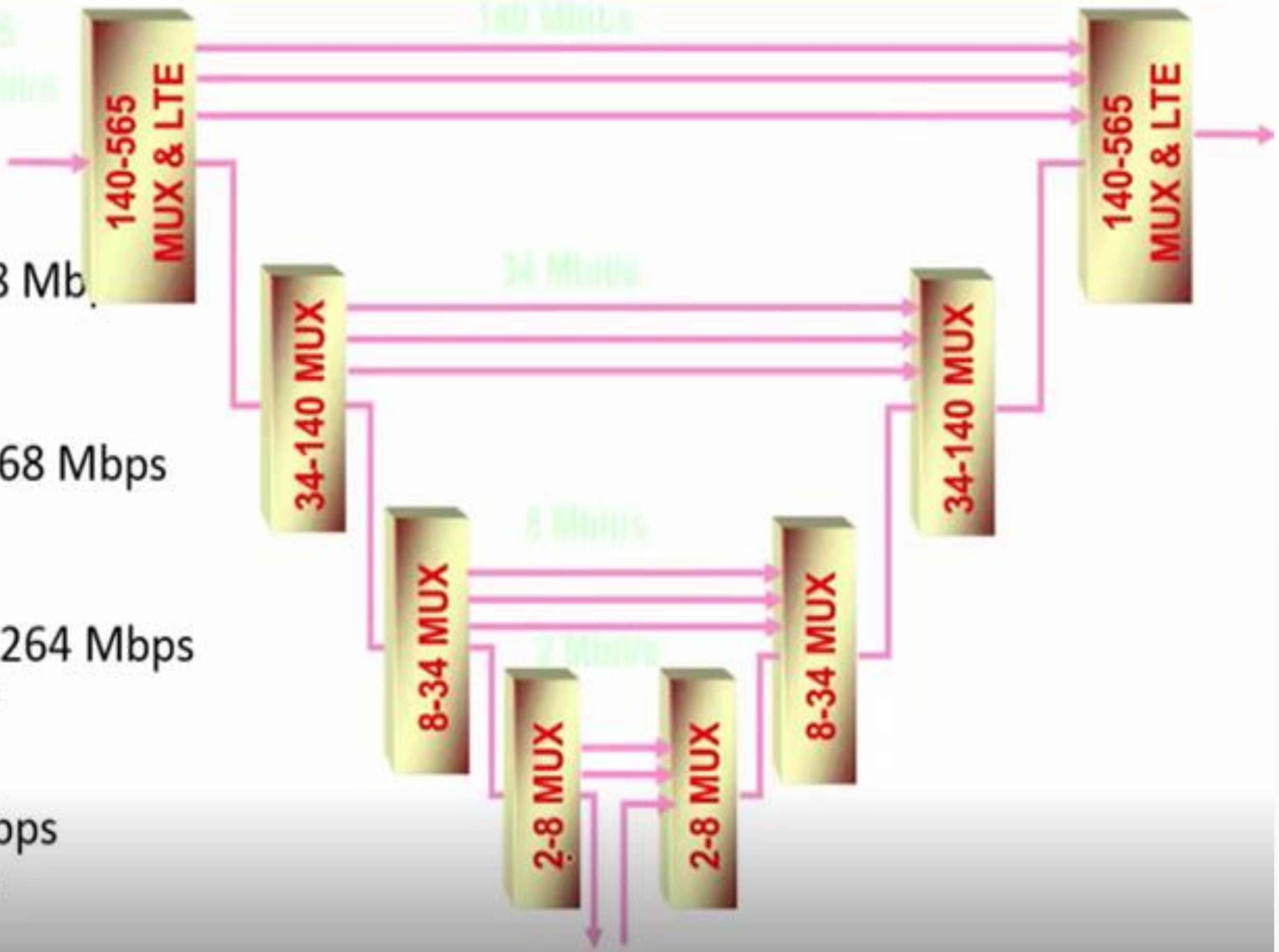
Capacity = 30 Base Channels

PDH (D2 Level)
4 * 2.048 +stuffing bits = 8.448 Mb
Capacity = 120 Base Channels

PDH (D3 Level)
4 * 8.448 + stuffing bits = 34.368 Mbps
Capacity = 480 Base Channels

PDH (D4 Level)
4 * 34.368 +stuffing bits = 139.264 Mbps
Capacity = 1920 Base Channels

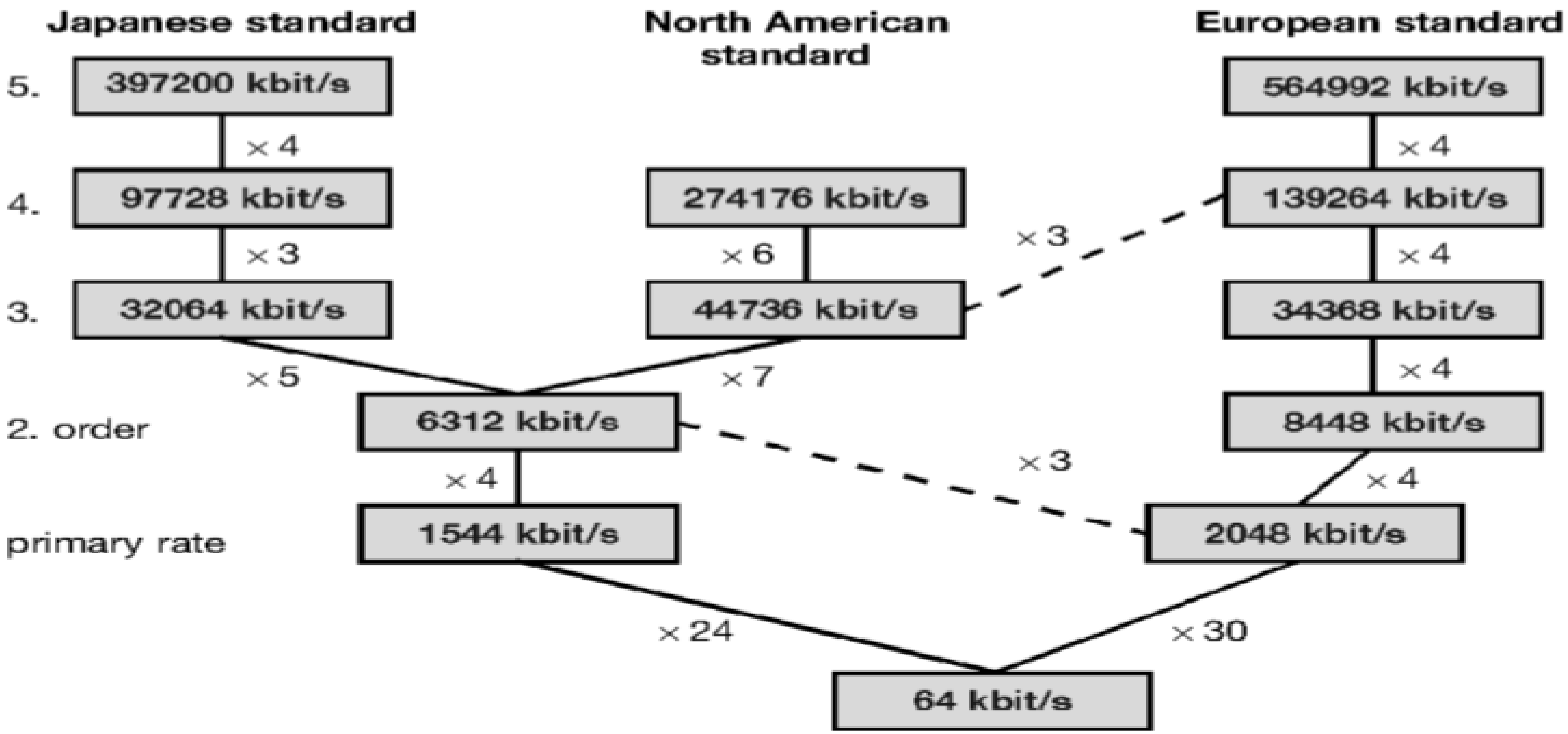
PDH (D5 Level)
4 * 139 + stuffing bits = 565 Mbps
Capacity = 7680 Base Channels



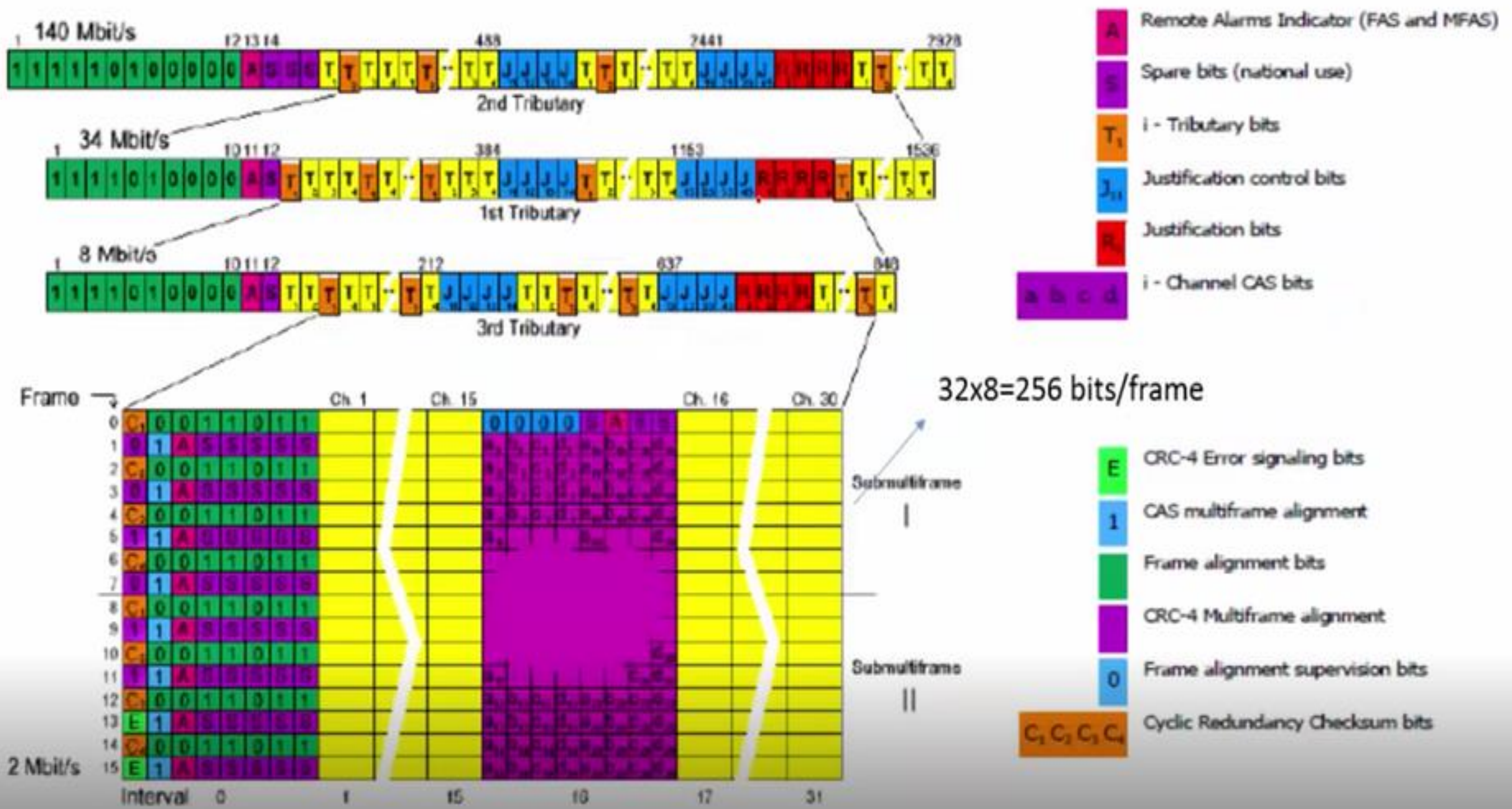
DIGITAL MUX LEVELS IN North America, Europe, Japan

Digital MUX Level	No. of 64Kb/s Channels	North America Mbits/s	Europe Mbits/s	Japan Mbits/s
0	1	0.064	0.064	0.064
1	24	1.544		1.544
	30		2.048	
	48	3.152		3.152
2	96	6.312		6.312
	120		8.448	
3	480		34.368	32.064
	672	44.376		
	1344	91.053		
	1440			97.728
4	1920		139.264	
	4032	274.176		
	5760			397.200

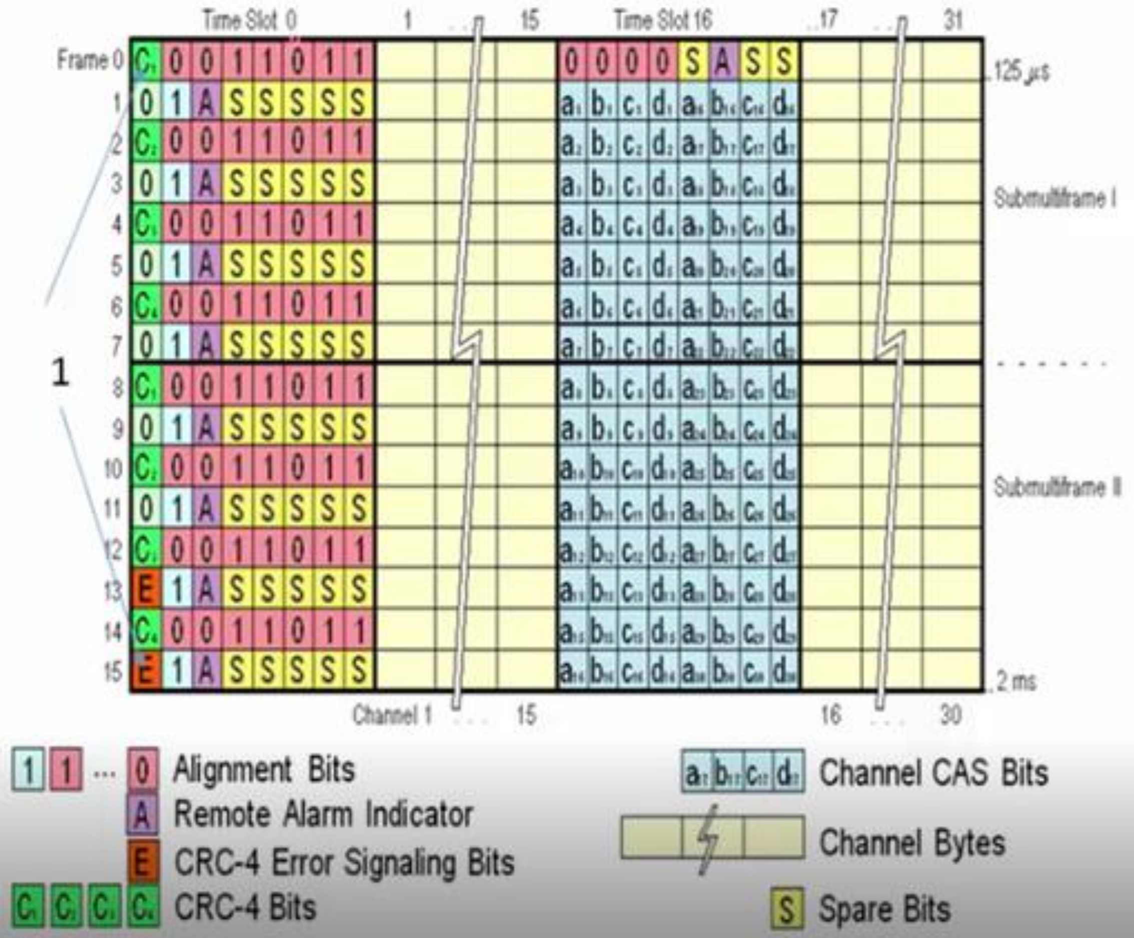
PDH HIERARCHY



The PDH hierarchy

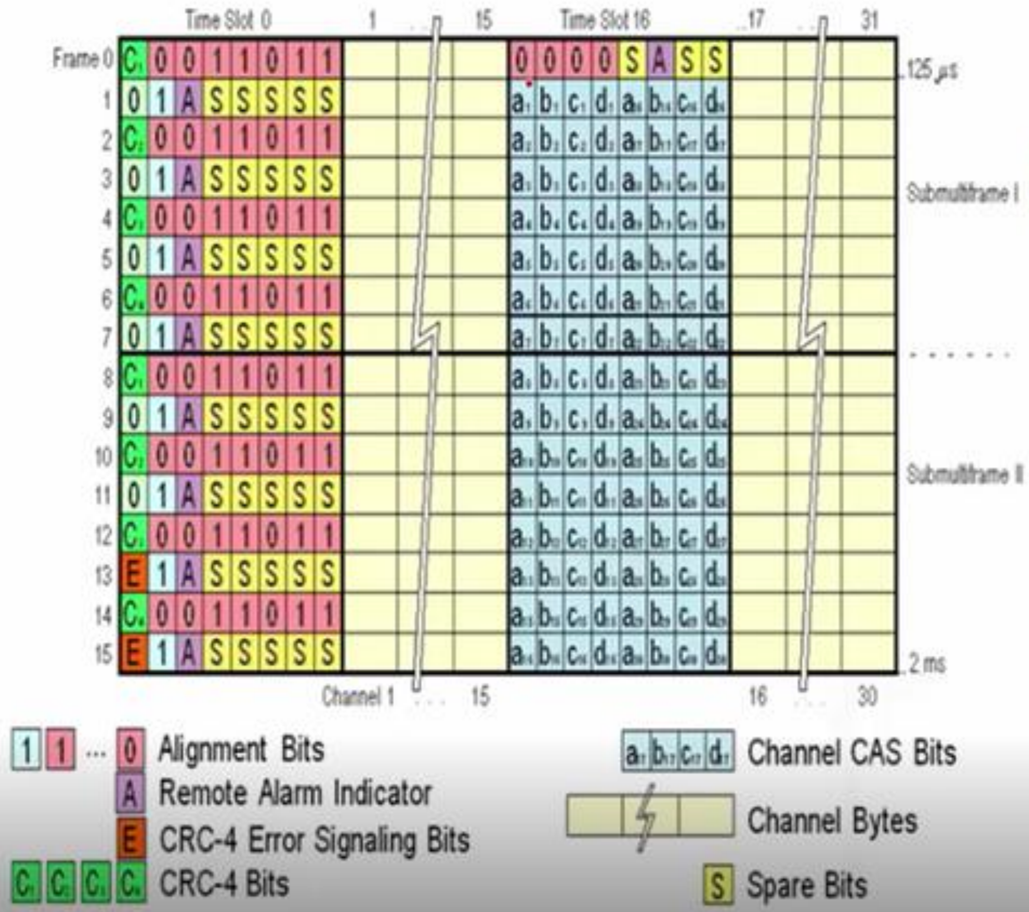


E1 carrier Frame Structure Details-TS0



- Time Slot 0 is for Synchronization and Transmission management.
- The 1st bit in TS0(FAS frame) is used for international use. If not used it should be set to 1.
- Bit 2 to 8 in TS0 is used for Frame alignment signal. A fixed FAS "0011011" is transmitted which enables the frame alignment of each frame between transmitter and Receiver. This is transmitted in alternative frames. This enables receiver to synchronize with transmitter.
- The 1st bit in TS0(NFAS) is used for international use and by default the value should be 1.
- The 2nd bit in TS0(NFAS) is fixed at 1 to assist in avoiding simulations of the frame alignment signal. Can you imagine what would happen if this value becomes 0 and rest of the bits sequence between b3 and b8 becomes equivalent to FAS frame pattern of 0011011?
- The 3rd bit (A Bit) in TS0(NFAS) is used for Remote Alarm indication, if alarm, set to 1 else 0.
- The bits S4 to S8 in TS0(NFAS) are spare bits for national use. Could be used for point to point applications, synchronization status messaging, if not used, then must be set to 1

E1 carrier Frame Structure Details-TS16



- Used for Signalling of information between Transmitter and receiver channels during the course of the call, origination ph number, destination ph number, engage, on-hook and for intended connection signalling between caller and called party.

  Bit 1 to bit 4 in TS16 have fixed value "0000" which is used as MFAS(used to synchronize CAS)

  Bit 5,7 and 8 are spare bits used for Application, maintenance or monitoring of performance.

  Bit 6 is used for alarm monitoring. A=1 in alarmed condition. Power fault, Loss of Signal, Loss of Multi-frame alignment. When A=1, CAS bits=1.

  Channel Associated bits (a,b,c,d):Channel numbers refer to telephone channel numbers. 64 kbit/s channel time slots 1 to 15 and 17 to 31 are assigned to telephone channels numbered from 1 to 30.

  This bit allocation provides four 500 bit/s signalling channels designated a, b, c and d for each channel for telephone and other services. With this arrangement, the signalling distortion of each signalling channel introduced by the PCM transmission system, will not exceed ± 2 ms

  When bits b, c or d are not used they should have the values: b = 1, c = 0, d = 1.

- It is recommended that the combination 0000 of bits a, b, c and d should not be used for signalling purposes for channels 1 to 15

Limitations of PDH

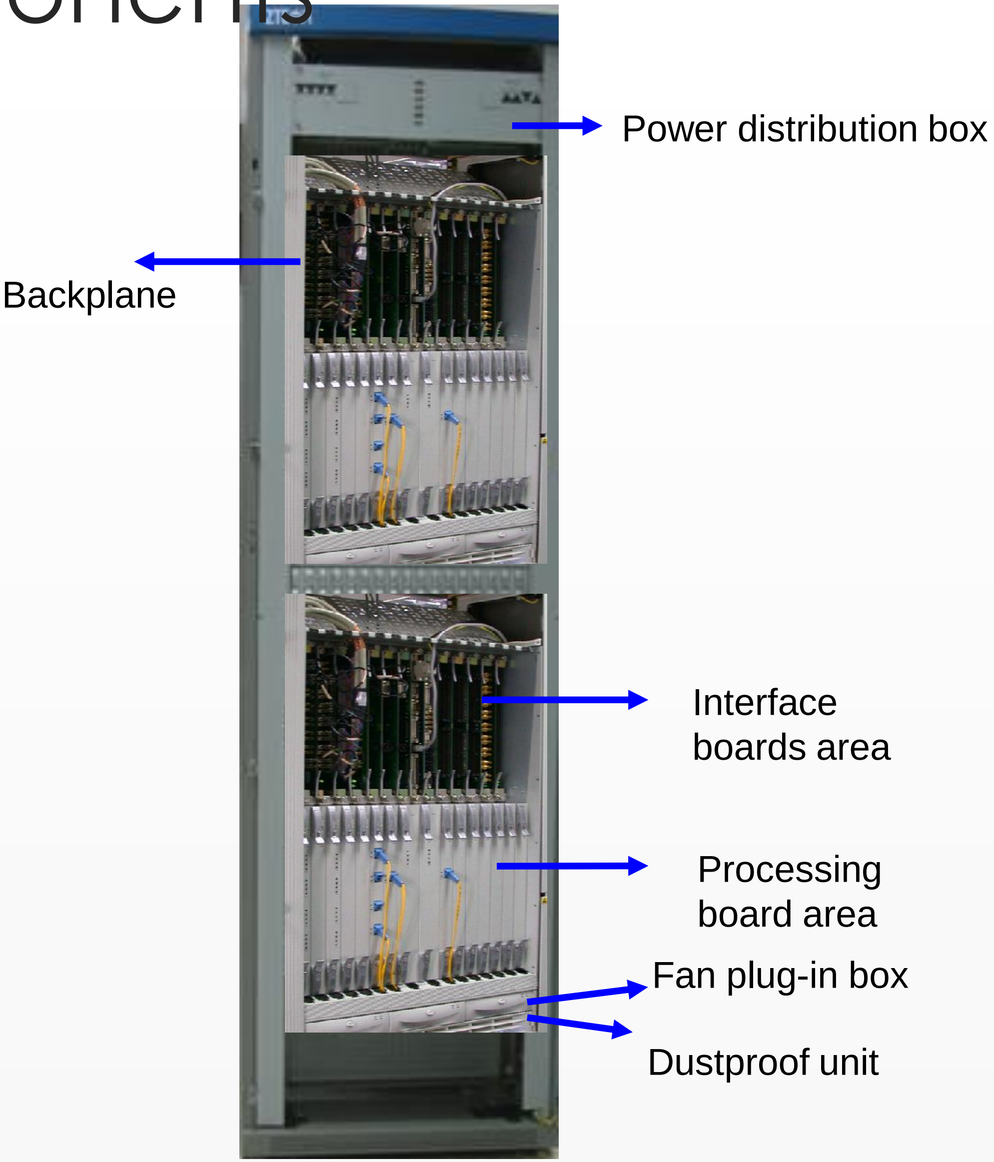
- Specialized equipment required for interwork two hierarchy
- Inability to identify individual channels in a higher order bit stream.
- Insufficient capacity for network management
- Higher bit rates are difficult to achieve
- Supports only linear topology
- no common standards among vendors.

SDH-Synchronous Digital Hierarchy

- SDH is an ITU-T standard for a high capacity telecom network.
- SDH is a synchronous digital transport system, aim to provide a simple, economical and flexible telecom infrastructure.
- This is the information structure used to support information payload and overhead information organized in a block frame structure which repeats every 125 micro seconds
- The basis of Synchronous Digital Hierarchy (SDH) is synchronous multiplexing - data from multiple tributary sources is byte interleaved.

Equipment Components

- Equipment's Components include：
  - Power distribution box
  - Sub rack
    - Board area
    - Back plane
    - Fan plug-in box
    - Dustproof unit



Requirements for SDH/SONET

- Need for extensive network management capability within the hierarchy. Flexibility in OAM through overhead provisions.
- Standard interfaces between equipment and support multiplexing formats for previous interfaces/data rate supported by PDH.
- Facilities to add or drop tributaries directly from a high speed signal=> should have ability to identify sub-streams in any high data rate frame.
- Need for inter-working between north American and European systems.(SDH with SONET and support for both E and T carrier)
- Provide synchronization between NE's and making all the NE's to take reference from PRC.
- Should support both electrical and optical interfaces.
- Should support Self healing mechanism- Ring architecture? And fast recovery.
- Ability to support and transport all the services of PDH.
- Ability to provide better interleaving mechanism compared to bit interleaving during multiplexing(byte interleaving?)
- Decrease stuffing bits during higher order multiplexing.

Building Blocks in SDH

- Regenerator (Reg.)
- Terminal Multiplexer (TM)
- Add/Drop Multiplexer (ADM)
- Digital Cross Connect (DXC)

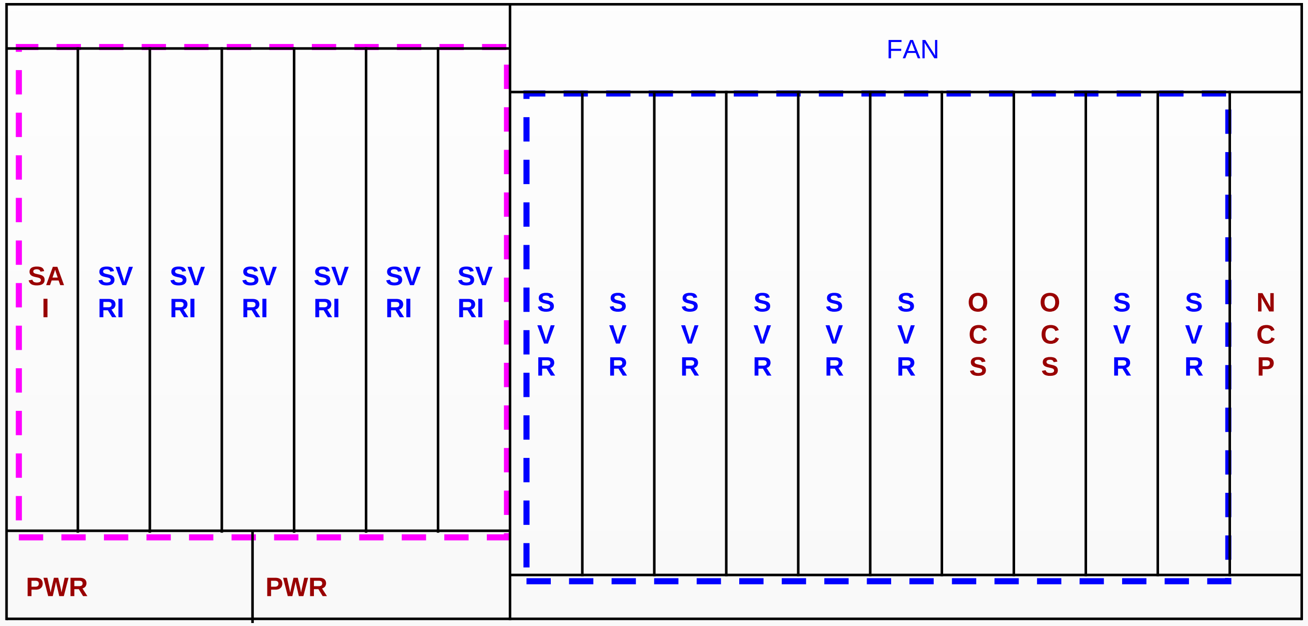
Comparing SDH and SONET

- The main difference between SDH and SONET is that SONET generally uses the VC-3 virtual container for data transmission, while SDH transports user data for the most part in VC-4 containers.
- This is because the existing North American PDH hierarchy, especially the third hierarchical layer, DS3 (44.736 Mbit/s), is better suited for transport in a VC-3 than in a VC-4.
- Furthermore, SONET has the extra STS-1 level with a bit rate of 51.84 Mbit/s that can transport exactly one VC-3 and is thus ideal for transporting DS3 streams.

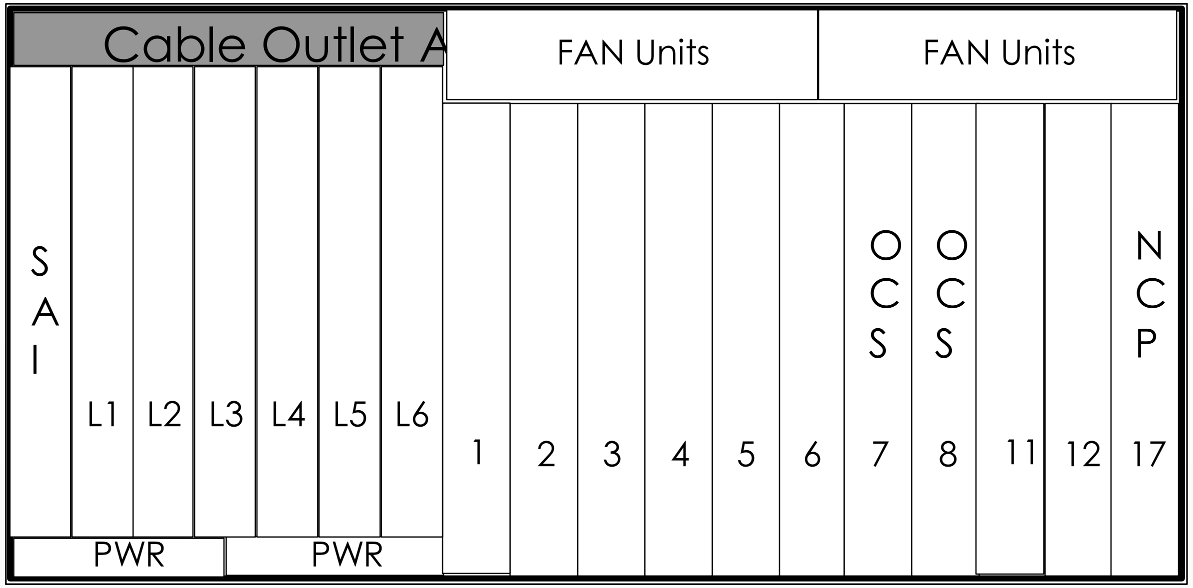

ZTE中兴
ZXMP
S325
FAN
FAN

Con…

SVRI: Interface/bridging card
SVR: Service Processing card



Introduction of Slots



Interface card area: from L1 to L6;

Processing card area: 1, 2, 3, 4, 5, 6, 11, 12

MERITS OF SDH

- Simplified multiplexing / demultiplexing techniques
- Direct access to lower speed tributaries
- Enhance Operation , Administration & Maintenance
- Easy growth to higher bit rates in steps with evolution of transmission technology
- Capable of transporting existing PDH
- Capable of transporting future ATM
- Capable of operating multi vendor and multi –operator environment

SDH Rates

- SDH is a transport hierarchy based on multiples of 155.52 Mbit/s.
- The basic unit of SDH is STM-1
- STM-Synchronous Transport Module
- Higher rate is an exact multiple of the lower rate therefore the hierarchy is synchronous.

STM-N

Where N =1,4,16 ,64   n is a multiples of four

SDH BIT RATES

SDH Levels	Bit rates in Mbps
STM-1	155.520
STM-4	622.080
STM-16	2488.320
STM-64	9953.28

STM-1 Frame

- Synchronous Transport Module –
- A frame with a bit rate of 155.52 Mbit/s is defined in ITU-T Recommendation G.707
- It is made up from a byte matrix of 9 rows and 270 columns.

SONET/SDH Designations and bandwidths				
SONET Optical Carrier level	SONET frame format	SDH level and frame format	Payload bandwidth(kbit/s)	Line rate (kbit/s)
OC-1	STS-1	STM-0	50,112	51,840
OC-3	STS-3	STM-1	150,336	155,520
OC-12	STS-12	STM-4	601,344	622,080
OC-24	STS-24	–	1,202,688	1,244,160
OC-48	STS-48	STM-16	2,405,376	2,488,320
OC-192	STS-192	STM-64	9,621,504	9,953,280
OC-768	STS-768	STM-256	38,486,016	39,813,120

CHAPTER -V

Introduction to Digital Subscriber Line

DSL, ADSL. HDSL, SDSL and VDSL

Digital Subscriber Line(DSL)

- DSL (digital subscriber line) is a technology for bringing high bandwidth information to home and small businesses over ordinary copper telephone lines.
- Digital subscriber line (DSL) technology transforms an ordinary telephone line into a broadband communications link, much like adding express LAN'S to an existing highway.
- DSL refers to different variations of DSL like ADSL,SDSL,VDSL…
- The term is a general term applied to a variety of different technologies used to achieve 'broadband' or high speed digital transmission.
- All DSL technology can be subdivided into one of two types:
  - ✓ SDSL (symmetric digital subscriber line) and
  - ✓ ADSL (asymmetric digital subscriber line)

Con…

- The prime difference between SDSL and ADSL is the speed of transmission in the downstream and upstream direction.
- In SDSL the transmission rate in downstream and upstream directions is the same (i.e. symmetric).
- In ADSL, the downstream rate of transmission is greater than the upstream bitrate (i.e. asymmetric).

Technologies of XDSL

- SDSL – Symmetrical DSL
  - Unlike ADSL, the bandwidth partitioning is symmetrical
  - More suitable for businesses
- HDSL – High-bit-rate DSL
  - HDSL reaches data rates up to 2Mbps without repeaters over 3.6km distance
  - 2 TP wires for full-duplex communication
- VDSL – Very-high-bit-rate DSL
  - Similar to ADSL, but uses coaxial, fiber-optic, or TP for higher rates over short distances (300-1800m)
  - 50-55Mbps downstream, 1.5-2.5 Mbps upstream

What Does ADSL mean?

- **A**symmetric - The data can flow faster in one direction than the other. Data transmission has faster downstream to the subscriber than upstream.
- **D**igital - No type of communication is transferred in an analog method. All data is purely digital, and only at the end, modulated to be carried over the line.
- **S**ubscriber **L**ine - The data is carried over a single twisted pair copper loop to the subscriber premises .

Con…

- **ADSL** is a form of DSL, a data communications technology that enables faster data transmission over copper telephone lines

- ADSL is capable of providing up to 50 Mbps, and supports voice, video and data.

- ADSL is the #1 Broadband Choice in the World with over 60% market share

- ADSL is now available in every region of the world

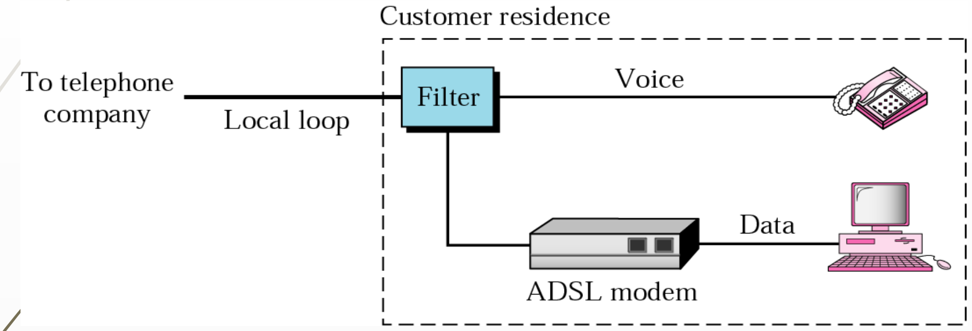
ADSL

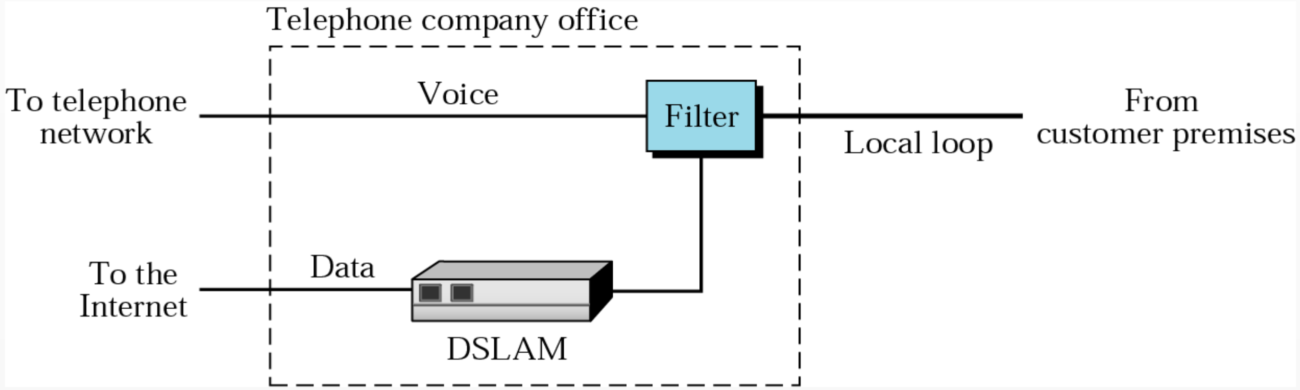
- In ADSL the total available capacity of the high speed digital subscriber line is split asymmetrically between downstream and upstream directions of transmission.
- There is a much higher bitrate made available for downstream transmission – at the expense of the upstream transmission rate.
- The advantage of this is that to get more information in downstream (e.g. delivery of webpages, software downloads etc.).
- ADSL is a technology used to provide a digital high-speed internet access line over a normal 2-wire telephone line.
- ADSL internet access line sharing the same 2-wire connection to the public network.

con…

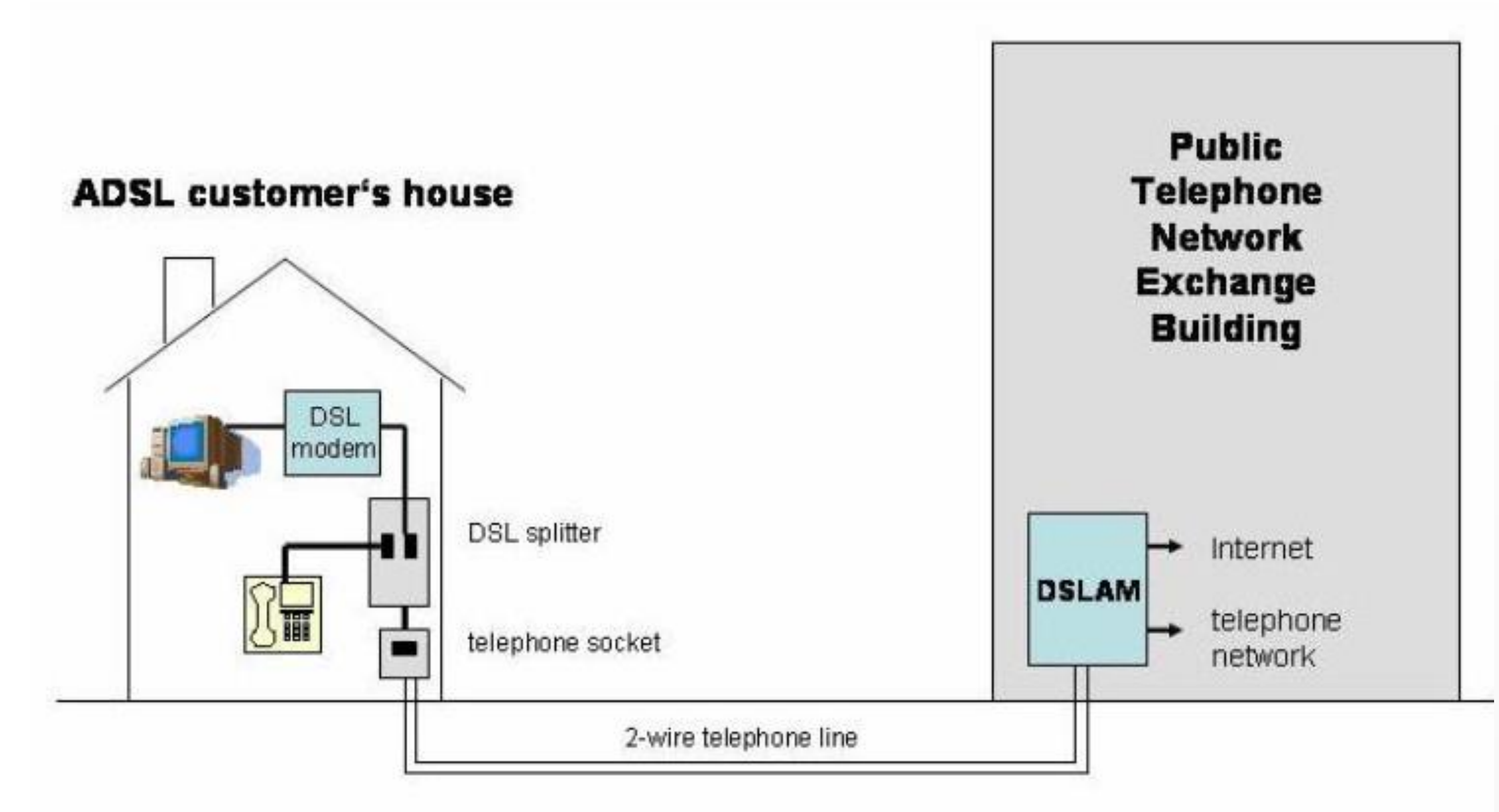
- **Narrower bandwidth for upstream transmission.**
- **Near-end crosstalk is reduced by partial or full separation of the upstream and downstream frequency bands.**
- **Simultaneous transport of POTS and data is achieved by transmitting data in a frequency band above voice telephony.**
- **Use of advanced transmission techniques (trellis coding, Reed-Solomon codes with interleaving, and DMT modulation).**

ADSL Equipment





Operation of ADSL



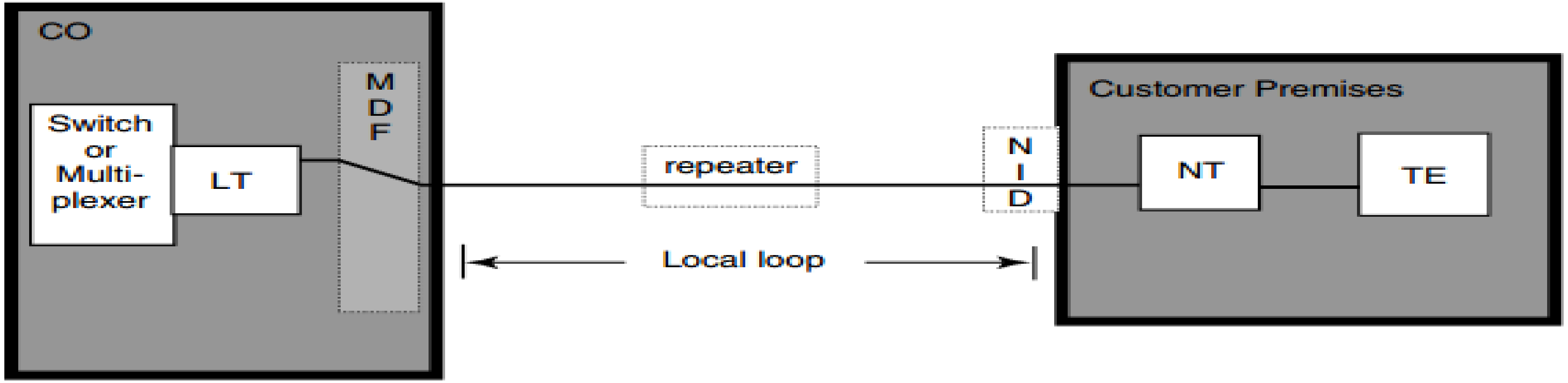
Con…

- To keep the telephone/ISDN and ADSL services apart, high frequency filters or splitters are used.
- At the customer’s premises, the splitting device is called simply a DSL splitter or DSL filter.

DSL splitter

DSL filter

DSL REFERENCE MODEL



CO: Central Office—A building where local loops connect to transmission and switching equipment.

LT: Line Termination—DSL modem at the network end of the local loop.

MDF: Main Distributing Frame—A wire cross-connecting field used to connect any loop to any CO equipment.

Repeater: A signal regeneration device located near the mid-point of a cable. This enables transmission on long loops. Repeaters are not needed for the majority of loops.

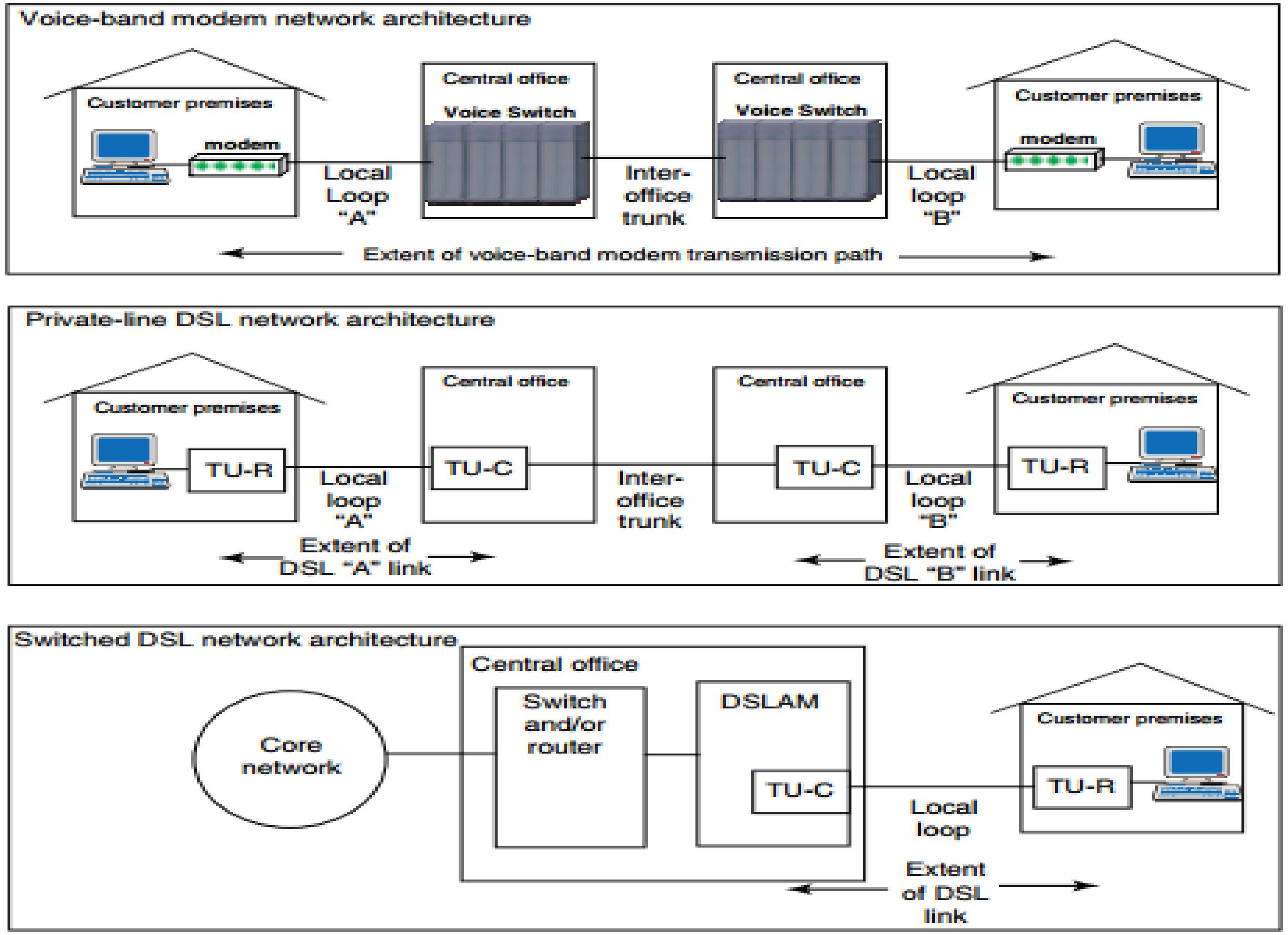
Local Loop: The telephone wire connecting the CO to the customer premises.

NID: Network Interface Device—The point of demarcation between the customer installation and the telephone company outside wire.

NT: Network Termination—DSL modem at the customer end of the local loop.

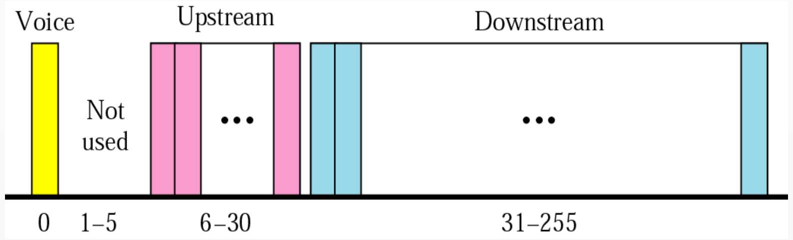
TE: Terminal Equipment—End-user equipment such as a personal computer or a telephone.

DSLAM based

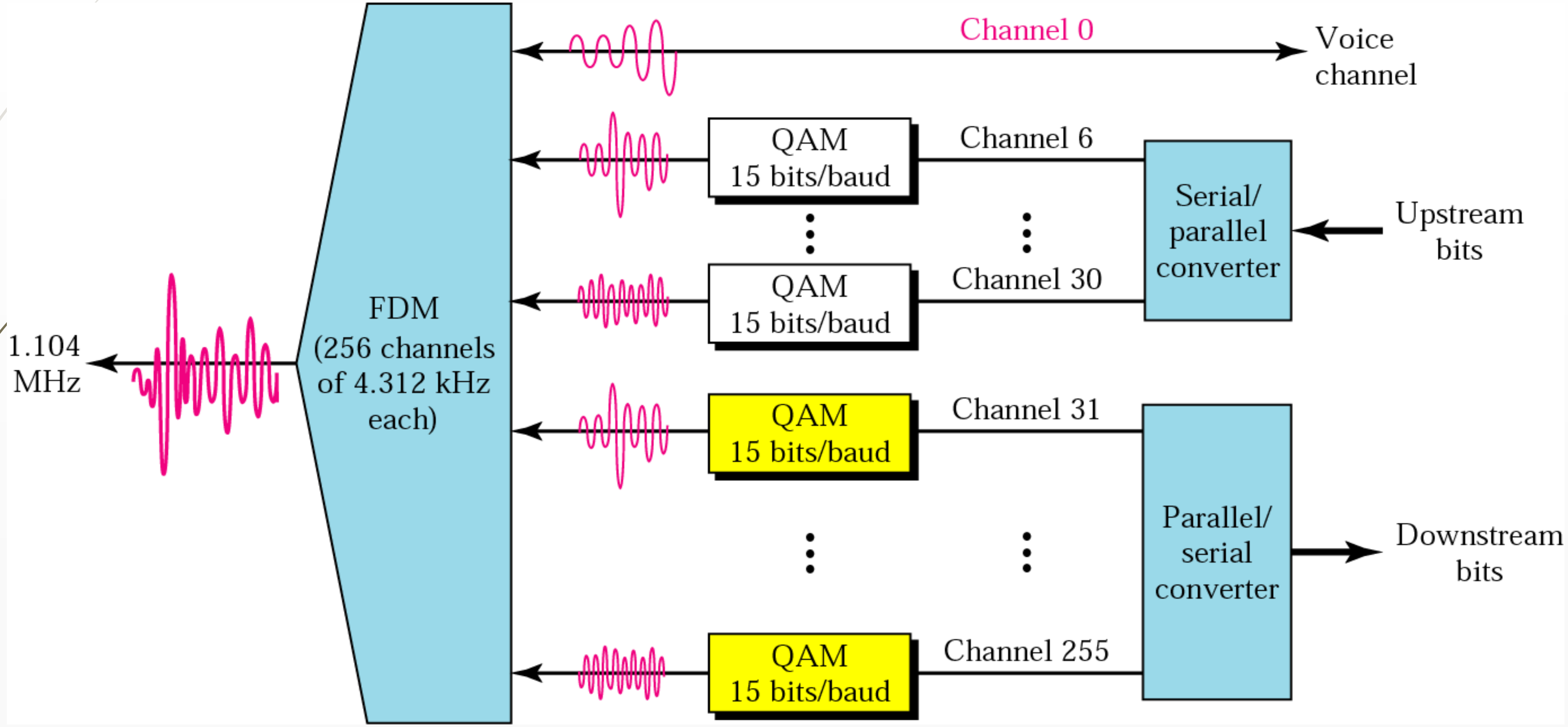

Voice-band modem network architecture
Customer premises
modem
Local Loop "A"
Central office
Voice Switch
Inter-office trunk
Central office
Voice Switch
Local loop "B"
Customer premises
modem
Extent of voice-band modem transmission path
Private-line DSL network architecture
Customer premises
TU-R
Local loop "A"
Central office
TU-C
Inter-office trunk
Central office
TU-C
Local loop "B"
Customer premises
TU-R
Extent of DSL "A" link
Extent of DSL "B" link
Switched DSL network architecture
Core network
Central office
Switch and/or router
DSLAM
TU-C
Local loop
Customer premises
TU-R
Extent of DSL link

Possible bandwidth division

- Channel 0: Voice
- Channels 1-5: Idle to separate voice and data
- Channels 6-30: 1 control and 24 data
  - 24 channels using 4KHz each with QAM
    24 x 4000 x 15 = 1.44Mbps max upstream
- Channels 31-255: 1 control and 224 data
  - 224 channels using 4KHz each with QAM
    224 x 4000 x 15 = 13.4Mbps max downstream
- Actual rates:
  - Upstream: 64Kbps-1Mbps
  - Downstream: 500Kbps-8Mbps



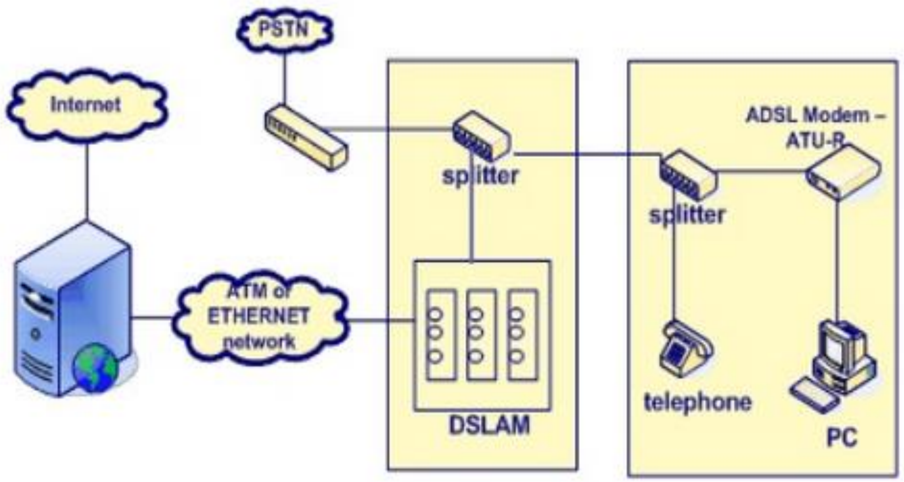
Digital division technique



ADSL Speed Factors:

- The distance from the local exchange
- The type and thickness of wires used
- The number and type of joins in the wire
- The proximity of the wire to other wires carrying ADSL, ISDN and other non-voice signals
- The proximity of the wires to radio transmitters.

Architecture:



Advantages of ADSL

- Faster downloads compared to dial-up or ISDN.
- No need for a second phone line - by allowing voice and data transfer at the same time (you can use the phone as normal while connected to the internet).
- Because ADSL transfers data digitally it doesn't need to convert the data from digital to analogue and back again.
- ADSL connections are Always on, which makes the usual long wait to connect a thing of the past.

Limitations of ADSL

- ADSL connections are not available to everyone, you need to be within 3 miles of an ADSL enabled exchange.
  The hardware costs can be quite significant as you will need a special ADSL modem and ADSL filters to use the service, most ISPs allow you to hire these items which can reduce the initial cost.
  Because ADSL connections are Always on you will need a firewall to protect your PC.
- The limit for ADSL service is **18,000 feet** (5,460 meters)
- At the extremes of the distance limits, ADSL customers may see speeds far below the promised maximums.

Factor for consideration choosing your ADSL supplier

- **Downstream bitrate**
- **Upstream bitrate**
- **Fast Path**
- **Geographical service coverage**
- **Monthly charges for connection**
- **Data volume charges – flat rate?**
- **Quality of Service (QOS)**
- **Forced release every 24 hours?**
- **DSL modem choice and price**
- **etc**

SDSL

- SDSL stands for 'symmetric digital subscriber line'.
- An SDSL line provides for transport of digital data simultaneously in both directions across the line.
- the same bitrate being available in both directions (thus 'symmetric').
- SDSL connections typically allow transmission of up to 6 Mbit/s in both directions, but usually require a 4-wire connection (equivalent to two standard telephone lines).
- SDSL service is typically more expensive than ADSL.

Chapter -VI

Packet Switched Networks OSI and IP models:

INTRODUCTION

- There are three types of switching used in PSTN network.
- Those are circuit switching , message and packet switching
- Circuit switching was designed for voice communication.
- Circuit switching creates dedicated links that are well suited to this type of connection.
- The circuit switching also limits the flexibility and not suitable for connecting variety of digital devices.

Con…

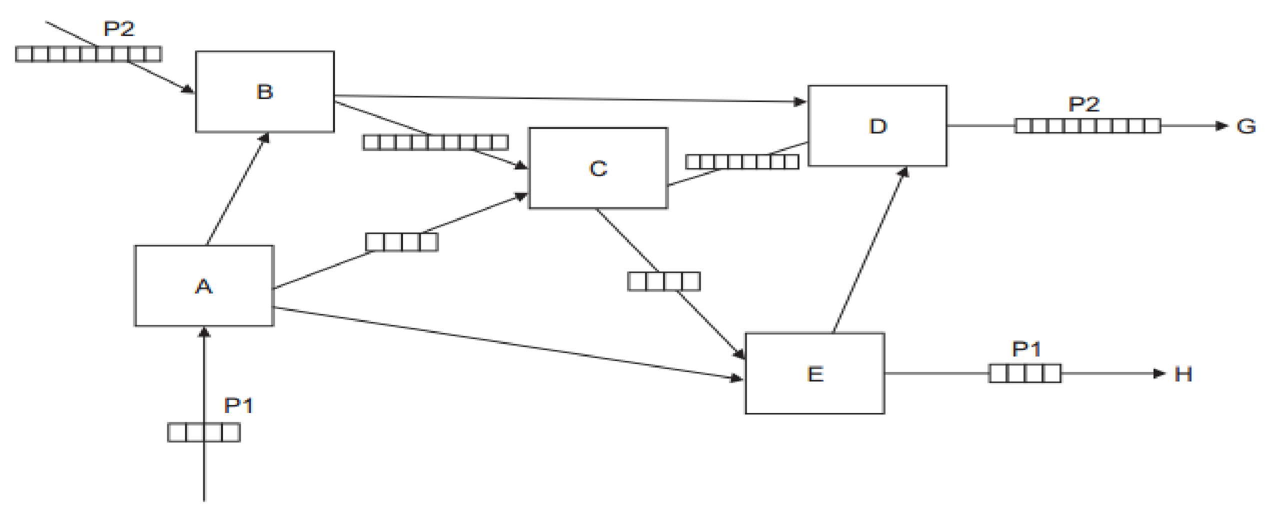
- For More efficient utilization of the network greater control channel bandwidth and high call processing capacities required.
- But the circuit switching not providing these capabilities.
- Message switching overcomes the limitations of circuits switching.
- This switching stores the incoming messages into a computer memory and forward it to the destination when available, This causes delay in switching.
- The packet switching overcomes all the limitations of message and circuit switching.
- Thus it is highly suitable for the data communication

Packet Switching Principles

- The DataStream originating at the source is divided into **packets of fixed or variable size.**
- The data communication system typically have **burst traffic.** Thus, the time interval between consecutive packets may vary, depending on the burstiness of the data stream.
- A typical upper bound on packet length is 512 octets (bytes). Each packet contains a portion of the user's data plus some control information.
- As the bits in a packet arrive at a switch or router, they are **read into a buffer.**
- When the entire packet is **stored, the switch routes the packet over one of its out going links.**

Con…

- The packet remains **queued** in its buffer until the outgoing link becomes idle.
- This technique is called **store and forward technique**
- Depends on the path availability the packet P1 choses the path as station A–C–E– H.
- Similarly the packet P2 choses the path as station B–C–D–G. In each station, the packets are stored in a buffer and forwarded to the next station after the availability.



Routing Control

- **In Packet switching messages are broken into packets and sends one at a time to the network.**
- **Routing control decides how the network will handle the stream of packets as it attempts to route them through the network and deliver them to the intended destination.**
- **The routing decision is determined in one of two ways. They are**

  i. **Datagram and**
  ii. **Virtual circuit.**

DATAGRAM

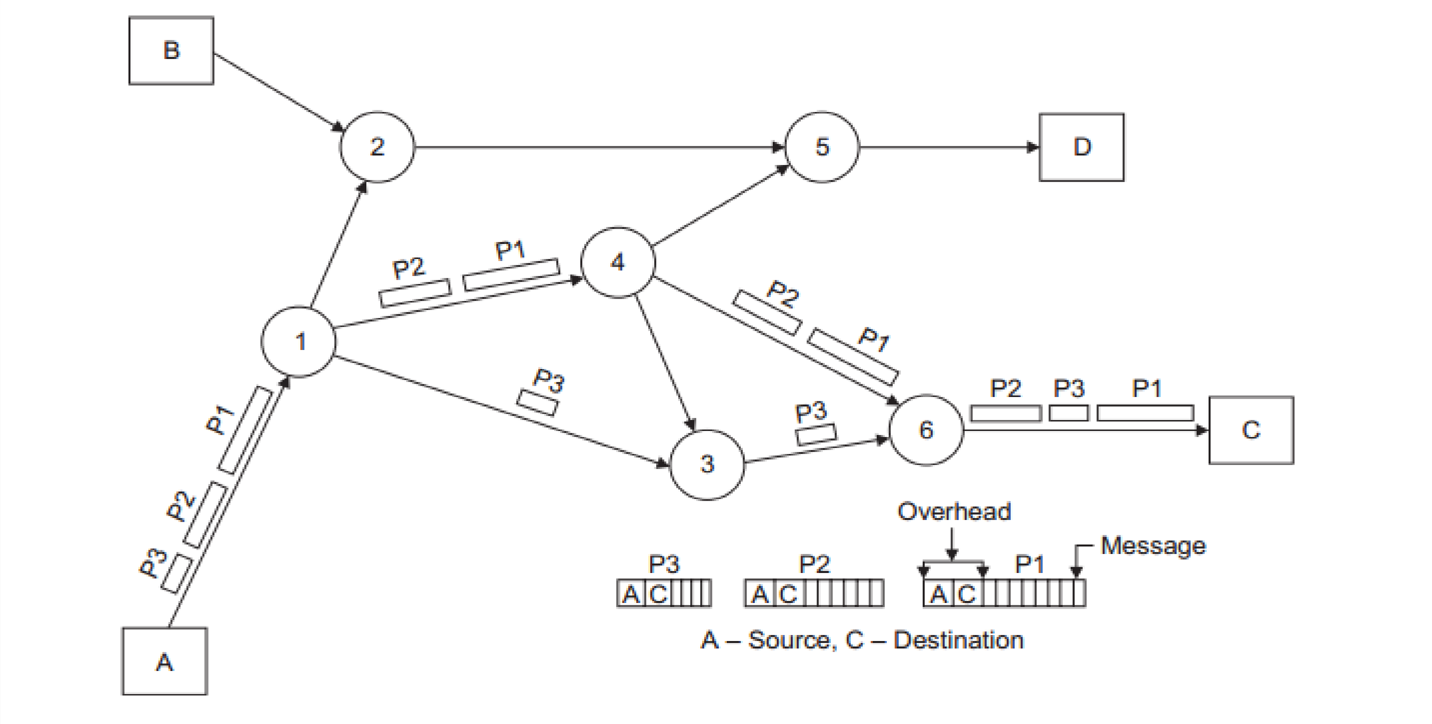
- In **datagram**, each packet within a stream is independently routed.
- A routing table stored in the router (switch) specifies the outgoing link for each destination.
- The table may be static or it may be periodically updated.
- In the second case, the routing depends on the router's estimate of the shortest path to the destination.
- Since the estimate may change with time, consecutive packets may be routed over different links.
- Therefore each packet must contain bits denoting the source and destination. Thus may be a significant overhead.

Con…

- The circled one are called the switching nodes whose purpose is to provide a switching facility that will move the data from node to node until they reach the destination.
- The squared one are called the stations. The stations may be computers, terminals, telephones or other.
- Station A is assumed to send three packets of message namely P1, P2 and P3 .
- At first, A transmits these packets to node 1. Node 1 makes decision on routing of these packets.

Con…

- Node 1 finds node 4 as shortest compared to node 3.
- Thus it passes p1 and p2 to node 4.
- Accidently, if node 4 is not accessible, node 1 finds node 3 as shortest and sends packet P3 to node 3.



Con…

- Node 3 and 4 sends its received messages to the destination C through node 6.
- It is shown that the order of the packet is changed due to the different routing of the packets.
- Thus in datagram, it is the responsibility of destination station to reorder the packets in proper sequence.
- Also if a packet crashes in a switching node, the destination C may not receive, all packets.
- In such a case also, it is the responsibility of station C to recover the lost packet.

Advantage of datagram switching over circuit switching

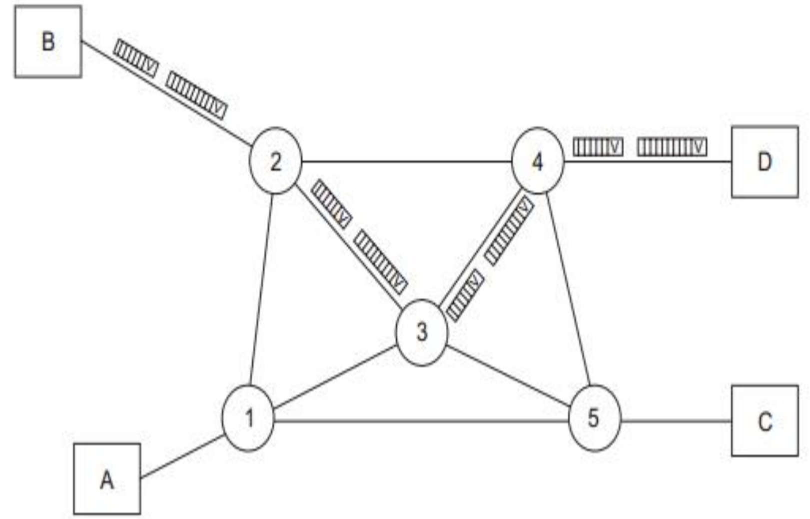
- It achieves higher link utilization when the traffic is burst.
- No dedicated path required

Virtual circuit

- In virtual circuit, a fixed route is selected before any data is transmitted in a call setup phase similar to circuit switched network.
- All packets belonging to the same data stream follow this fixed route called a virtual circuit.
- Packet must now contain a virtual circuit identifier.
- This bit string is usually shorter than the source and destination address identifiers needed for datagram.
- Once the virtual circuit is established, the message is transmitted in packets.

Con…

- **suppose that station B has two messages to send to the destination D.**
- First B sends a control packet referred as call-request packet to node 2, requesting logical connection to D.
- Node 2 decides to route the request and the subsequent message packets through node 3 and 4 to destination D.

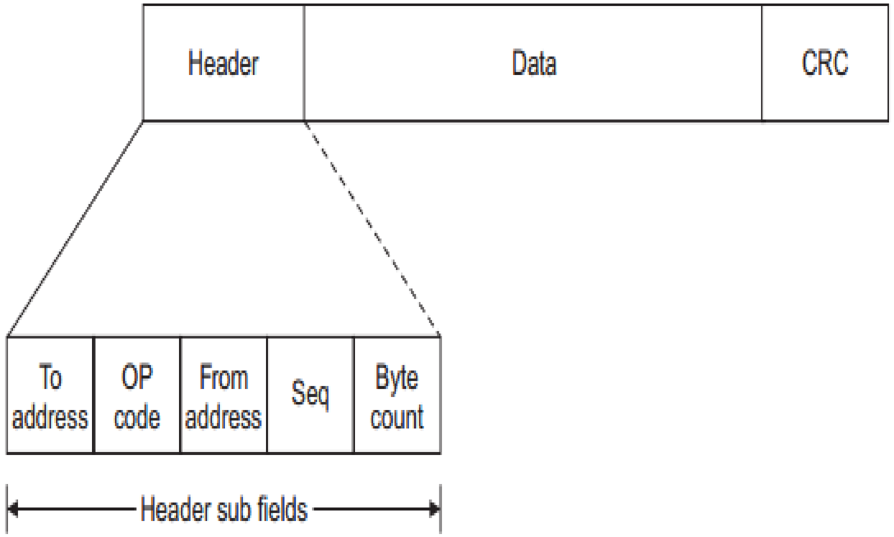


- **If D prepared to accept the connection, it sends a call-accept packet to node 4.**
- **Node 4 sends the call-accept packet to B through node 3 and 2.**
- **Because the route is fixed for the duration of the logical connection, it is somewhat similar to a circuit switching network and is referred to as a virtual circuit.**

Advantage of virtual circuit

- All packets follow the same route
- They reach the destination in the same order
- There is no need of re ordering work for destination station
- Error control
- Packets transit/traverse rapidly
- Source and destination identifiers are short bits.

Packet format



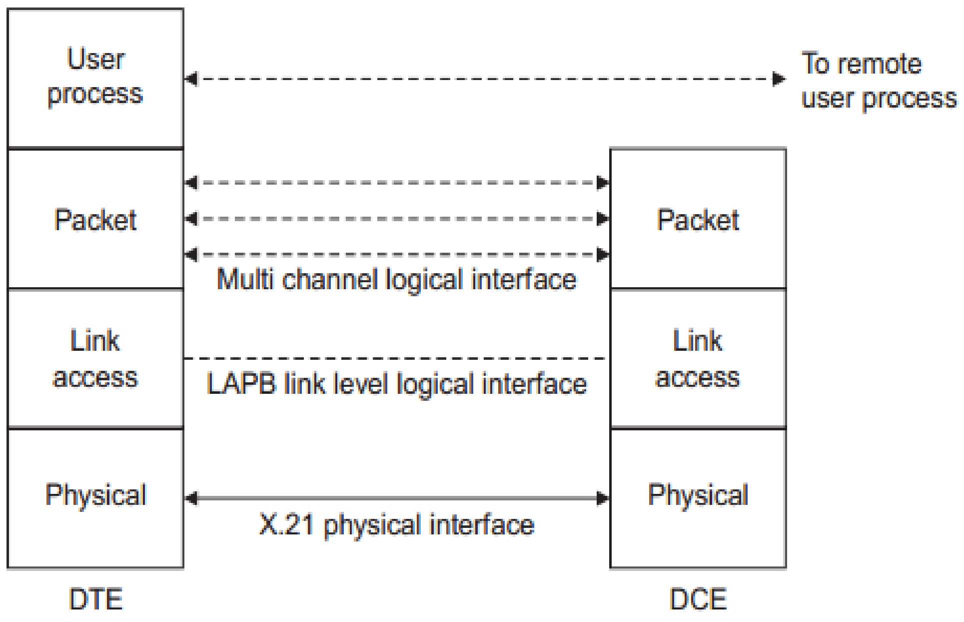
- The format of a packet switching may vary rom network to network
- Generally packets have header includes all related control information.

1. **Header.** It contains sub fields in addition to the necessary address fields.
   a. **Op code.** It designates/show whether the packet is a message (text) packet or control packet.
   b. **A sequence number (Seq)** to reassemble messages at the destination node, detect faults and facilitates recovery procedures.
   c. **Byte count.** Used to indicate the length of a packet.
2. **Data**
3. **CRC The cyclic redundancy checks (CRC)**

X-25

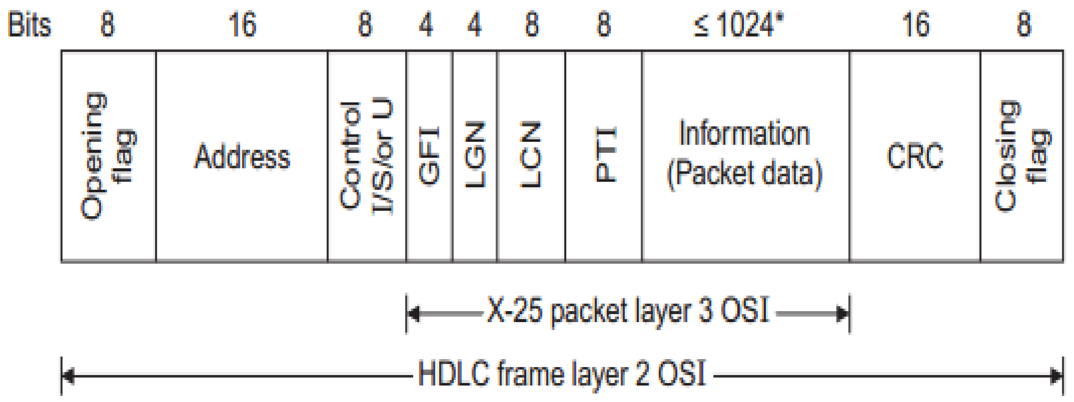
- **X-25 is an ITU standard, well known and most widely used protocol established in 1976.**
- **The standard specifies an interface between a host system and a packet switching network.**
- **X-25 standard for packet switching is a lower three layer equivalent of the OSI model, physical layer, a link layer, and packet layer.**
- **The data link layer of X-25 is link access procedure balanced (LAPB) using high level data link control (HDLC).**
- **HDLC is a bit oriented protocol based on the synchronous data link protocol (SDLC) established by IBM for Synchronous Network Architecture (SNA) networks**

X-25 interface



- The physical layer deals with the physical interface station and switching node with X-25
- At packet layer, data are transmitted as packets over virtual circuits.
- The link layer provides for the reliable transfer of data across the physical link by transmitting the data as a sequence of frames
- The link layer standard is known as LAPB

HDLC FORMAT



- The opening flag and closing flag are made up of 8 bit information.
- Packets are delimited by a starting and an ending flag (01111110).
- The address field is typically 8 bits long, but can be extended in increments of 8 bits.
- ***Control I/S/ or U*** **:** Control information consists of 8 bits of data describing the **type of HDLC frame.**
- ***Information (I).*** Used to transfer data across the link at a rate determined by the receiver and with error detection and correction.
- ***Supervisory (S).*** Used to determine the ready state of the devices receiver is ready (RR), receiver is not ready (RNR) or reject (REJ).
- ***Unnumbered (U).*** Used to dictate parameters, such as set modes, disconnect and so on.

Con…

Packet information : The packet information consists of GFI, LGN and LCN.

- **General format identifier (GFI).** 4 bit of information that describes how the data in the packet is being used, from/to an end user, from/to a device controlling the end user device and so on.
- **Logical channel group number (LGN).** 4 bits of information that describe the grouping of channels.
- **Logical channel number (LCN).** 8 bit of information of the actual channel being used. Theoretical number of logical channels available is 2048.
- **Packet type identifier (PTI).** It is an 8 bit sequence that describers the type of packet being sent across the network. six types of packets are used in X-25 switching network.
- They are call request, call accept, clear request, interrupt request, reset request and restart request.
- **Cyclic Redundancy Check (CRC).** The 16 bit sequence is used for error detection and/or correction.

OSI MODEL

- **The ISO developed OSI for networking.**
- **An open system is a set of protocols that allows two computers to communicate with each other regardless of their design, manufacturer or CPU type.**
- **The concept of an open system approach to networking allows any device or system operating with any protocol to communicate with another device or system using its own protocol.**

OSI layer specifications

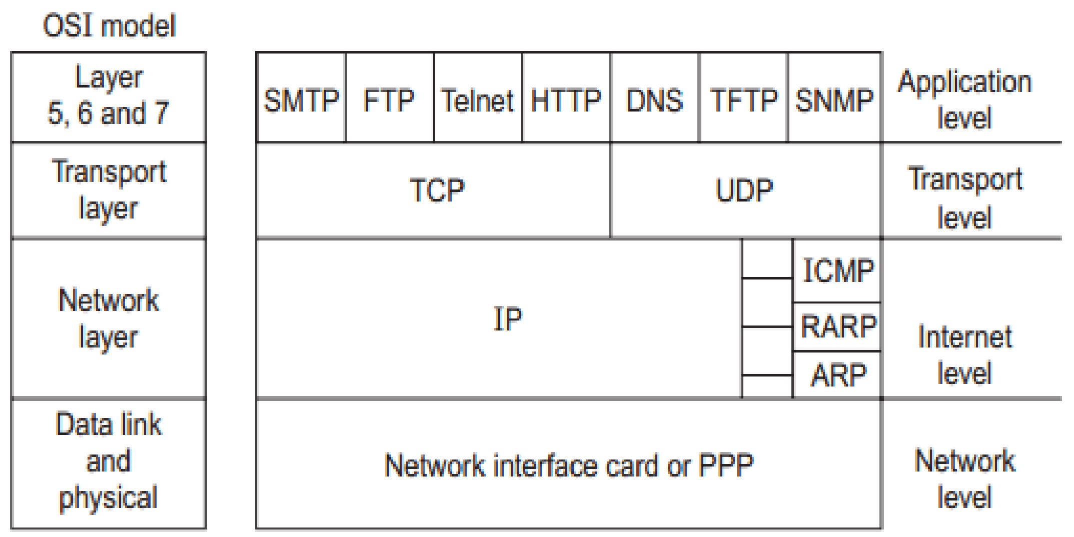
LAYER	SPECIFICATIONS
7	**APPLICATION LAYER:** Performs information processing such as file transfer, e-mail and teletext. Detailed and application specific information about data being exchanged.
6	**PRESENTATION LAYER:** Defines the format of data to be sent : ASCII, data encryption, data compression and EBCDIC.
5	**SESSION LAYER:** Management of connections between programs. Sets up a session between two applications by determining the type of communication such as duplex, half duplex, synchronization etc.
4	**Transport layer:** Delivery of sequence of packets. Ensures data gets to destination. Manages error control, flow control and quality of service.
3	**NETWORK LAYER:** Format of individual data packets. Sets up connection, disconnects connection, provides routing and multiplexing.
2	**DATA LINK LAYER:** Manages framing, error detection, and retransmission of message. Access to and control of transmission medium.
1	**PHYSICAL LAYER:** Medium and signal formed of raw bit information. Electrical interface (type of signal), Mechanical interface (type of connector), converts electrical signal to bits, transmits and receives electrical signals.

TCP/IP

- Transmission Control Protocol and Internet Protocol (TCP/IP) was developed to create LANs and also for internetworking multiple LAN's.
- Data sharing and broadcasting are prominent features of LAN technology.
- An organization may create different LANs with different protocols.
- These networks are connected together by an internal gateway, in turn connected to the external gateway of the internet.
- TCP/IP protocol is used to communicate among nodes.
- Today these protocols are the primary building blocks for the Internet.

TCP/IP Reference Model

- **TCP/IP** architecture is a **four layer** stack that deals with the equivalent **seven layer architecture** of **OSI.**



Application level : Some of the internet applications are SMTP, FTP, Telnet, HTTP, DNS, TFTP, SNMP.

Simple Mail Transfer Protocol (SMTP) is used for E-mail. It is used for transferring messages between two hosts.

Telnet :-It enables one computer to establish a connection to another computer.

File transfer protocol (FTP) is an internet standard for file transfer.

Hypertext Transfer Protocol (HTTP) is an advanced file retrieval program that can access distributed and linked documents on the web.

DNS. Domain Name System (DNS) is used to identify and locate computers connected to the internet.

SNMP. Simple Network Management Protocol (SNMP) is used by the network administrator to detect problem in the network such as router and gateway.

Con…

Transport Level Protocols : This layer consists of

- **User Datagram Protocol (UDP)** and
- **Transmission Control Protocol (TCP).**

Datagram Protocol (UDP) It provides unreliable service between hosts.

UDP accepts information from the application layer and adds a source port, destination port, UDP length and UDP checksum.

Con…

16 bits	16 bits
Source port	Destination port
UDP length defines number of bits in UDP header and data	Checksum used for error detection of UDP header and data
Data	

- The **IP protocol** adds its **header** to the packet received from UDP and passes to LLC.
- The LLC generates 802.2 Frame and passes to MAC layer.
- MAC adds its own header and transfers the frame to the physical layer for transmission.

Transmission Control Protocol (TCP)

- The transmission control protocol (TCP) is a transport layer that carries application layer packets and services between two users.
- In TCP, connection between users is established before transmitting information.
- TCP assigns a sequence number to each packet.
- The receiving end checks the sequence number of all packets to ensure that they are received.
- When the receiving end gets a packet, it sends acknowledgment.
- If the sending node does not receive an acknowledgement within a given period of time, if retransmits the previous packet.

Source	Destination	Sequence number	Acknowledgement number	Data offset	Unused

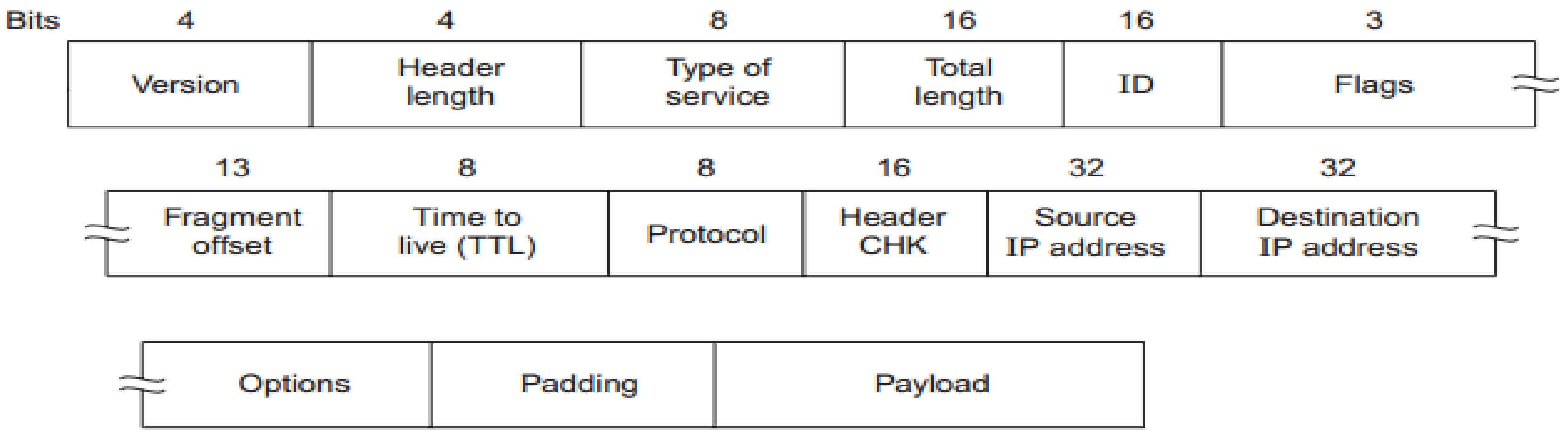
TCP header

URG	ACK	PSH	RST	SYN	FIN	WIN	CHK	UGP	OPT	Payload data

RST. It resets the connection.
SYN. used to establish a connection in combination with the acknowledge (ACK) bit.
SYN = 1, ACK = 0 is a connection request and
SYN = 1, ACK = 1 is a connection acknowledgement.
FIN. The FIN flag set to '1' indicates that the incoming packet is the last packet.
WIN. The window specifies the number of data bytes the sender is willing to accept.
CHK. checksum is 16 bit long. It is used for error detection in TCP header and data field.
OPT. It is called option field
URG = 0 = end of option list
1 = no operation
2 = maximum segment size

- **Source and destination.**
- **Sequence number.**
- **Acknowledgment number.**
- **Data offset.** it indicates how much the beginning of the data is offset by the header.
- **URG.** informs the receiver to find the urgent
data indicated by URG.
- **ACK.** Acknowledge flag bit is set to '1' to represent that the acknowledgement number is valid.
- **PSH.** This field set to '1' means that the receiver should push the data to an application as soon as possible.

IP PACKET FORMAT



- **The function of IP is a packet delivery with unreliable and connectionless service**
- **IP delivers message packets within the same network or within the interconnected networks .**
- **The IPV4 address size is 32 bits =4 billion users**
- **The later version IPV6 has the address size of 128 bits.**

In IP packet delivery

- There is no acknowledgement from the destination to source.
- There is no physical connection between the source and destination.
- IP datagrams can arrive at the destination out of order.

Ethernet and IEEE 802.3

- Ethernet was invented by Xerox Corporation in 1972.
- Ethernet is the most commonly used LAN technology today
- Ethernet still enjoys continued popularity and growth, Some of the reasons are :
  1. Least expensive
  2. It is fast enough for the vast majority of applications in use
  3. It continue to keep pace with other LAN technologies
  4. Its various standards supporting a wide variety of media
  5. Sufficiently defined standards
  6. Ethernet is considered to be the most user friendly.

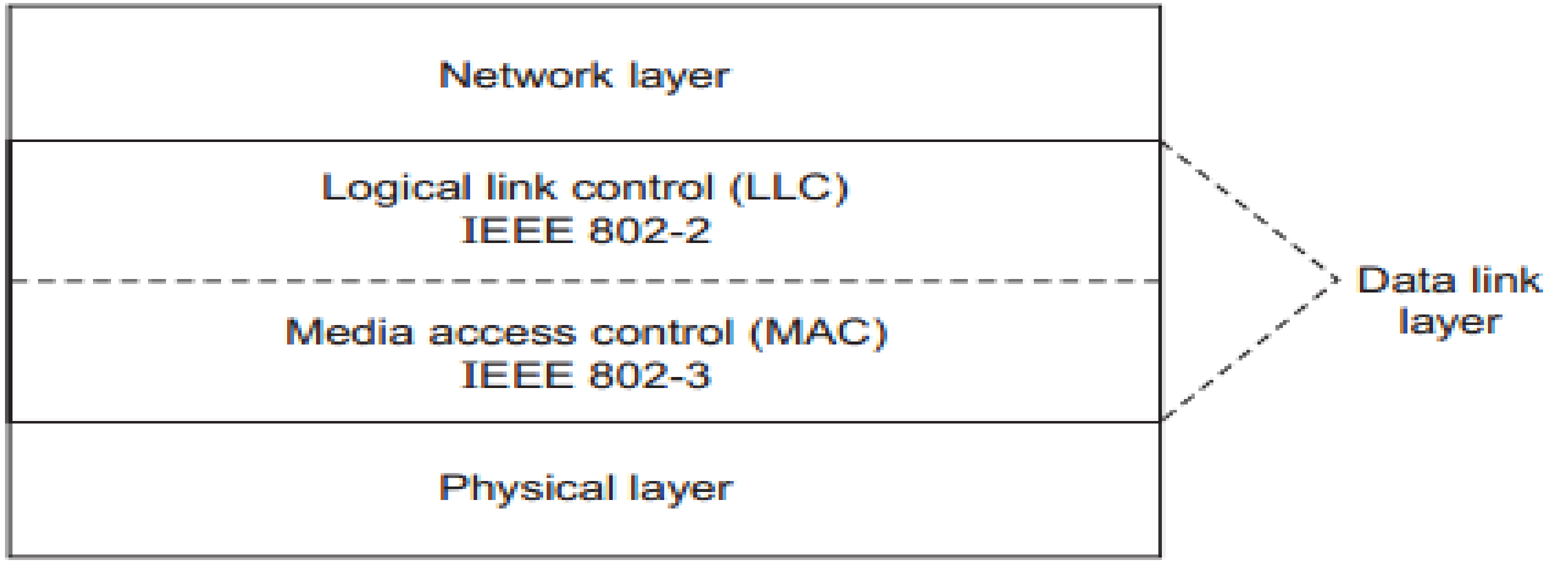
Ethernet Reference Model.

- The IEEE developed the standards for Ethernet in 1984.
- It is called IEEE 802.3 and uses bus topology.
- The Ethernet system consists of three basic elements

1. The physical media used to carry signals,
2. A set of rules, embedded in each Ethernet interface and
3. The Ethernet frame that consists of bits used to carry data, control and address information.

- ✓ Each station in the network consists of Network Interface Card (NIC).
- ✓ This card is connected to the Ethernet cable via a transceiver cable.
- ✓ The NIC is also called as Ethernet controller

Ethernet (IEEE 802.3) reference model.



The data link layer is divided into two sublayers. They are :

1. Logical link control (LLC) and

2. Media Access control (MAC)

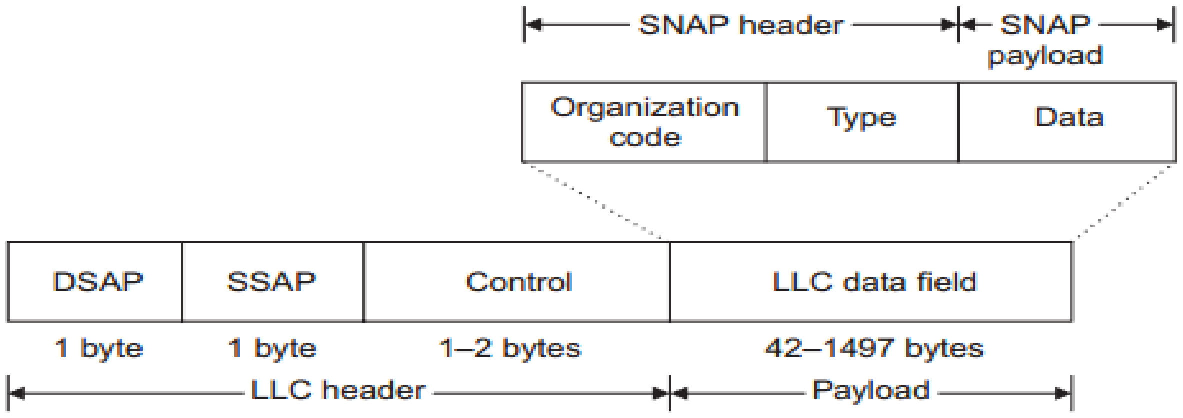
LLC layer

- The LLC layer is designed to establish a logical connection between source and destination.
- The IEEE standard for LLC is IEEE 802.2

responsibility of LLC layer

i. stablishing and terminating a communication link LLC1/LLC2
ii. Supplying a frame format for the pay load
iii. detecting and correcting errors and
iv. Maintaining control over the traffic flow

FRAME Format of LLC(IEEE 802.2)



- Destination Service Access Point (DSAP) and Source Service Access Point (SSAP) defines the location of the end points in communication link for either connection oriented or connectionless service
- The control field identifies the frame type
- If the LLC frame is used to encapsulate, a higher level protocol within the pay load field, 802.2 provides for a means, known as the sub network assess protocol (SNAP),to identify the protocol

MAC layer

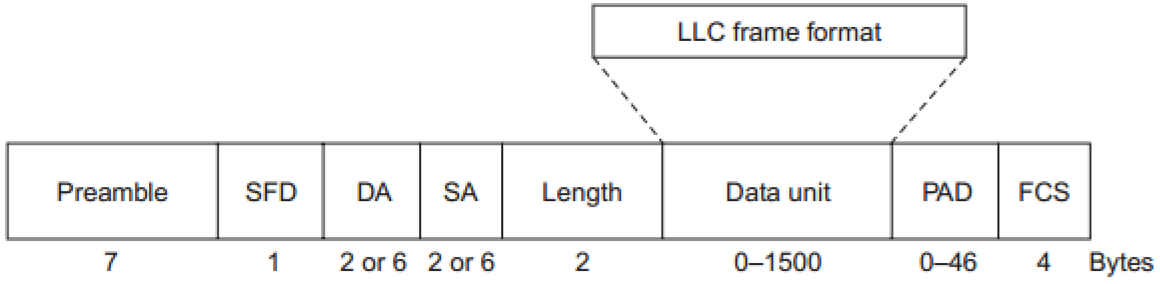
- The function of MAC is to access the network.
- It defines, how different stations can access the transmission medium.
- The MAC uses CSMA/CD protocol.
- In an Ethernet network, each station uses CSMA/CD protocol to access the network in order to transmit information.

Working principle of CSMA/CD:

1. If a station wants to transmit, the station senses the channel.
2. If the channel is busy, it continues the sensing of the channel. When the channel becomes idle, the station starts transmit data.
3. After sending data, the station senses for the collision, as there is a possibility that two station may send data at the same time.
4. If the collision detected, the station which detected first sends a jamming code (32bits) on the bus, in order to indicate the other station that there is a collision on the bus.
5. If no collision, transmission completed.
6. The two stations, which enveloped in collision, wait according to back-off algorithm.

Back off algorithm is a method used to generate waiting time for stations that were involved in collision.

FRAME Format of MAC



- The preamble provides signal synchronization.
- preamble and start Frame Delimiter (SFD) Synchronize the receiver
- Frame check sequence (FCS) is used to detect errors and corrupted information during transmission. IEEE uses CRC-32 for error detection.
-

Ethernet Media

- **The Ethernet network uses four different media.**
- **They are 10 Base 5,10 Base 2, 10 Base T and 10 Base-F.**
- **10 Base 5 (Thick Net).** It uses 10 Mbps Ethernet media with base band signaling with maximum segment lengths of 500 meters.
- **10 Base 2 (Thin Net).** It uses 10 Mbps Ethernet media with maximum segment lengths of 185 m, The maximum length of a network cable is 925 meters with four repeaters
- **10 Base T.** It uses 10 Mbps 22 to 26 AWG twisted pair cable (unshielded) instead of coaxial cable, A maximum segment length of 100 meter is supported.

  The transceiver is built into NIC.
- **10 Base F.** It uses 10 Mbps Ethernet media over fiber optic cable.

Token Bus and Token Ring Networking

The Ethernet system has two main disadvantages.

1. This system may not be suitable for a high traffic if excessive number of collisions are anticipated.
2. Due to number of retransmission, reduced throughput and delay increased.

The alternative for the Ethernet system are **Token Bus (IEEE 802.4) and Token Ring (802.5)**

The token bus combines features of Ethernet and

Token ring to provide a deterministic delay under heavy loads without causing collisions.

Token ring is a powerful LAN technology that is designed to handle heavy loads.

TOKEN BUS NETWORKING (IEEE 802.4)

- The Token bus system operates on a bus topology and is suitable for industry, factory automation and process control
- A token is a short message that specifies the *station currently using the network* and *the next station that access after the current station finish its work*.
- The token is passed from station to station in descending order and it is not necessary for all the stations on the bus to be active at all times.

The procedure for transmit a data by a computer are.

- **The computer has to wait for the token.**
- **Once it possess the token, it can add its traffic to the data stream.**
- **It then includes the successors address and passes the token.**
- **After passing the token, the sending computer monitors, whether the successor receives the token or not.**
- **If the successor had data, it sends in data stream and passes to the next successor.**
- **If it does not have any data to communicate, it simply modify the successor station address.**
- **Similarly the token passes to each station and reaches the original station which sends message.**

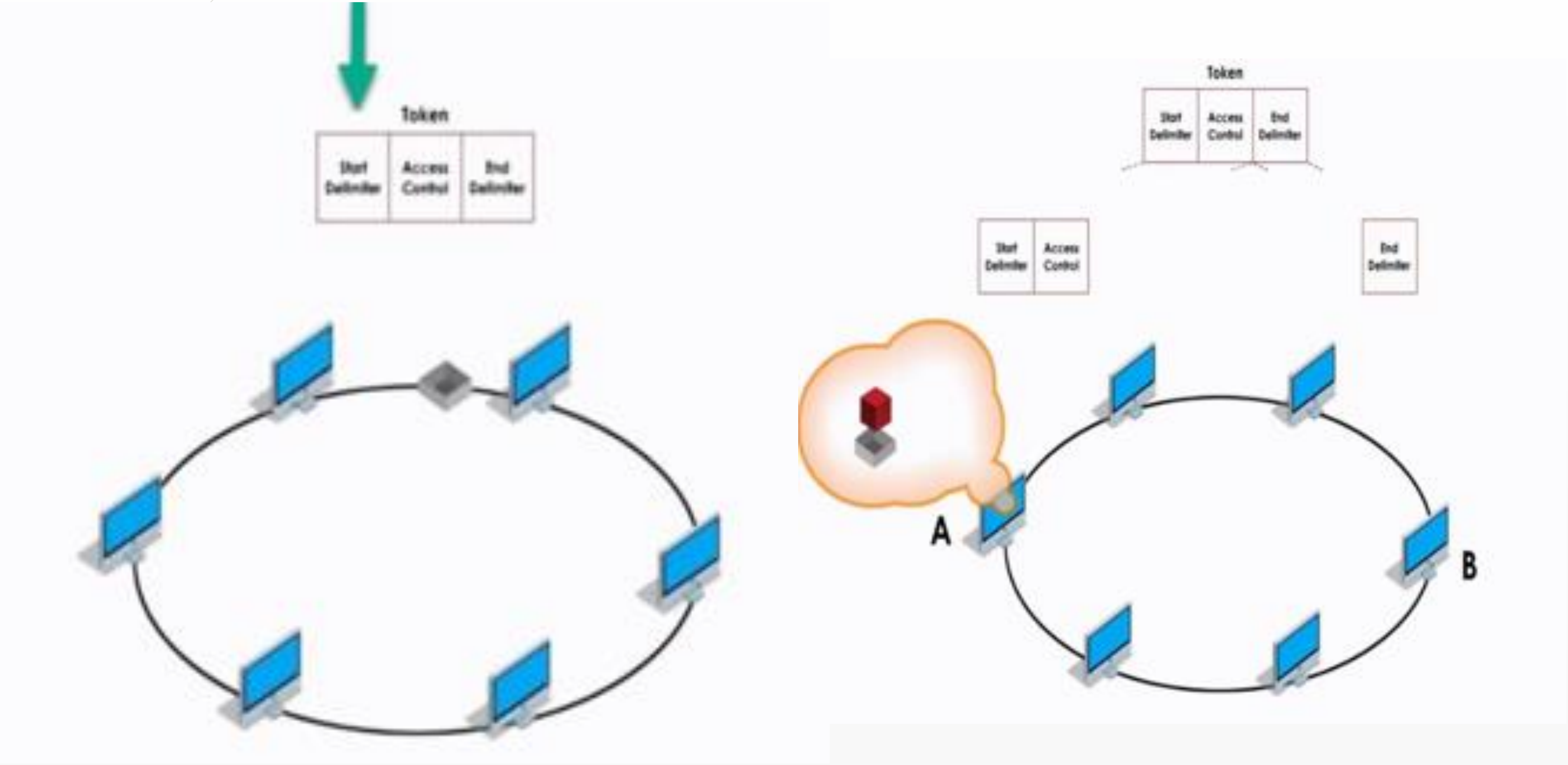
Con…

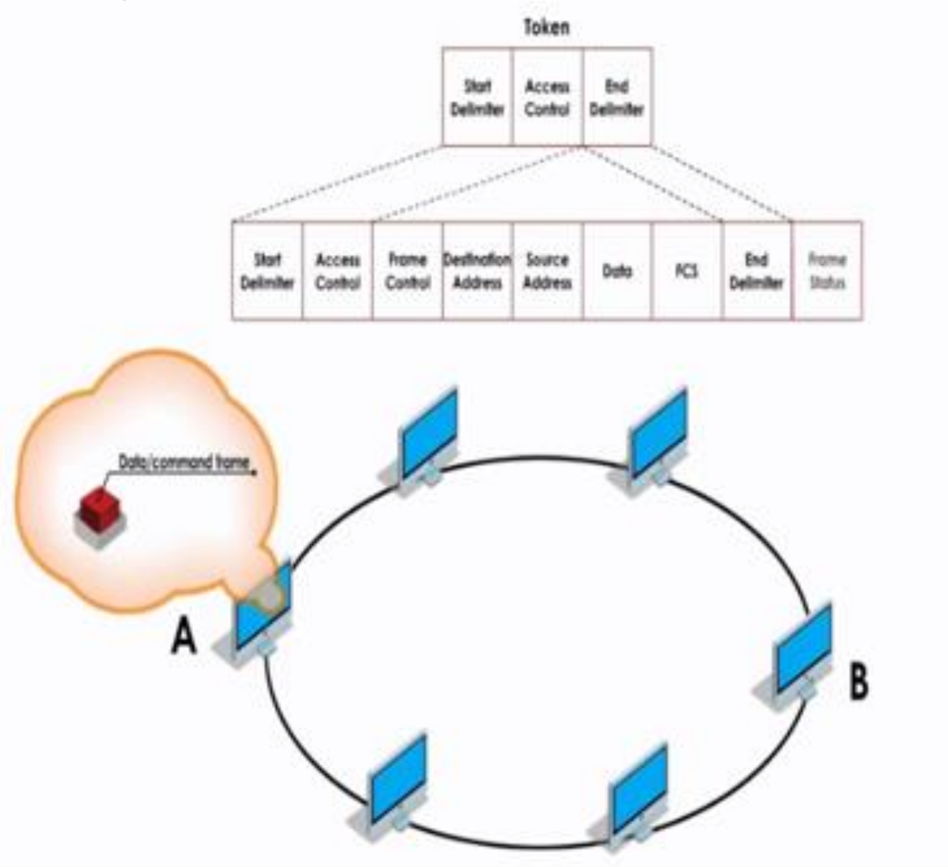
- The computer checks whether the data reached or not.
- Then if it has any data, it transfers to the destination, otherwise the token is passed to the successor address.
- As the designated system may not be active, the information sent to this computer may be kept in queue.
- After three continuous attempt, the source station removes the message meant for destination.
- This technology has not been widely used because of its delayed property

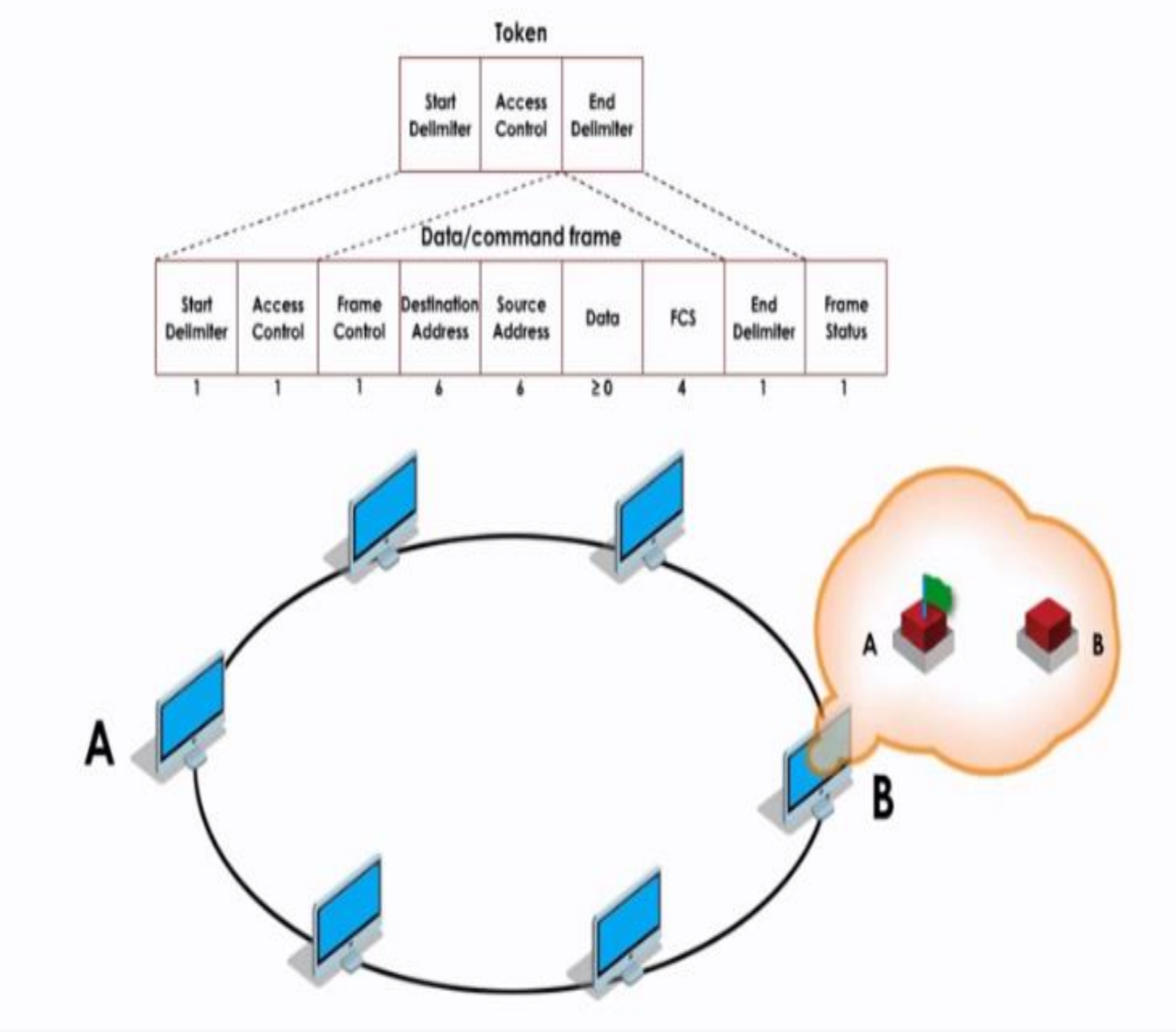
TOKEN RING NETWORKING (IEEE 802.5)

- Token ring network was introduced in 1985, at a data rate of 4 Mbs and 1989 with a data rate of 16 Mbps by IBM for local area network.
- The main difference of token ring over token bus is that tokens and messages are passed around the ring to each station in the ring in a fixed sequence.
- Token ring technology consists of a ring station and a transmission medium.
- A token ring network uses a wiring concentrator device called Multi-station Access Unit (MAU).
- Physically, token ring is a star topology and electrically it is a ring topology.

Con…


Token
Start Delimiter
Access Control
End Delimiter
Token
Start Delimiter
Access Control
End Delimiter
Start Delimiter
Access Control
End Delimiter
A
B


Token
Start Delimiter
Access Control
End Delimiter
Start Delimiter
Access Control
Frame Control
Destination Address
Source Address
Data
FCS
End Delimiter
Frame Status
Data/command frame
A
B


Token
Start Delimiter
Access Control
End Delimiter
Data/command frame
Start Delimiter
Access Control
Frame Control
Destination Address
Source Address
Data
FCS
End Delimiter
Frame Status
1
1
1
6
6
≥0
4
1
1
A
B
A
B

Any station wants to send messages follow the procedures

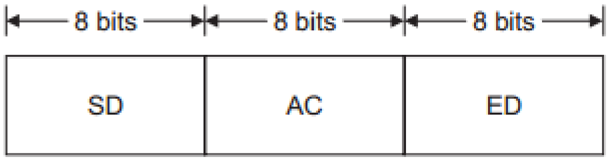
1. The station seize/hold the token. The token is a three byte frame circulating around the network.
2. Once the station possesses/holds the token, it inserts information into the token and transmits the frame on the ring.
3. The next station checks the destination address of the frame. If not matches, pass it to the next station. If the address matches, it performs the following function

   I. The destination station copies the message and sets the last two bits of the frame to inform the source that the frame was copied and the frame is retransmitted. Via fame status.

   II. The frame circulates on the ring until it reaches the source. Once it reaches the source, the source removes the frame from the ring.

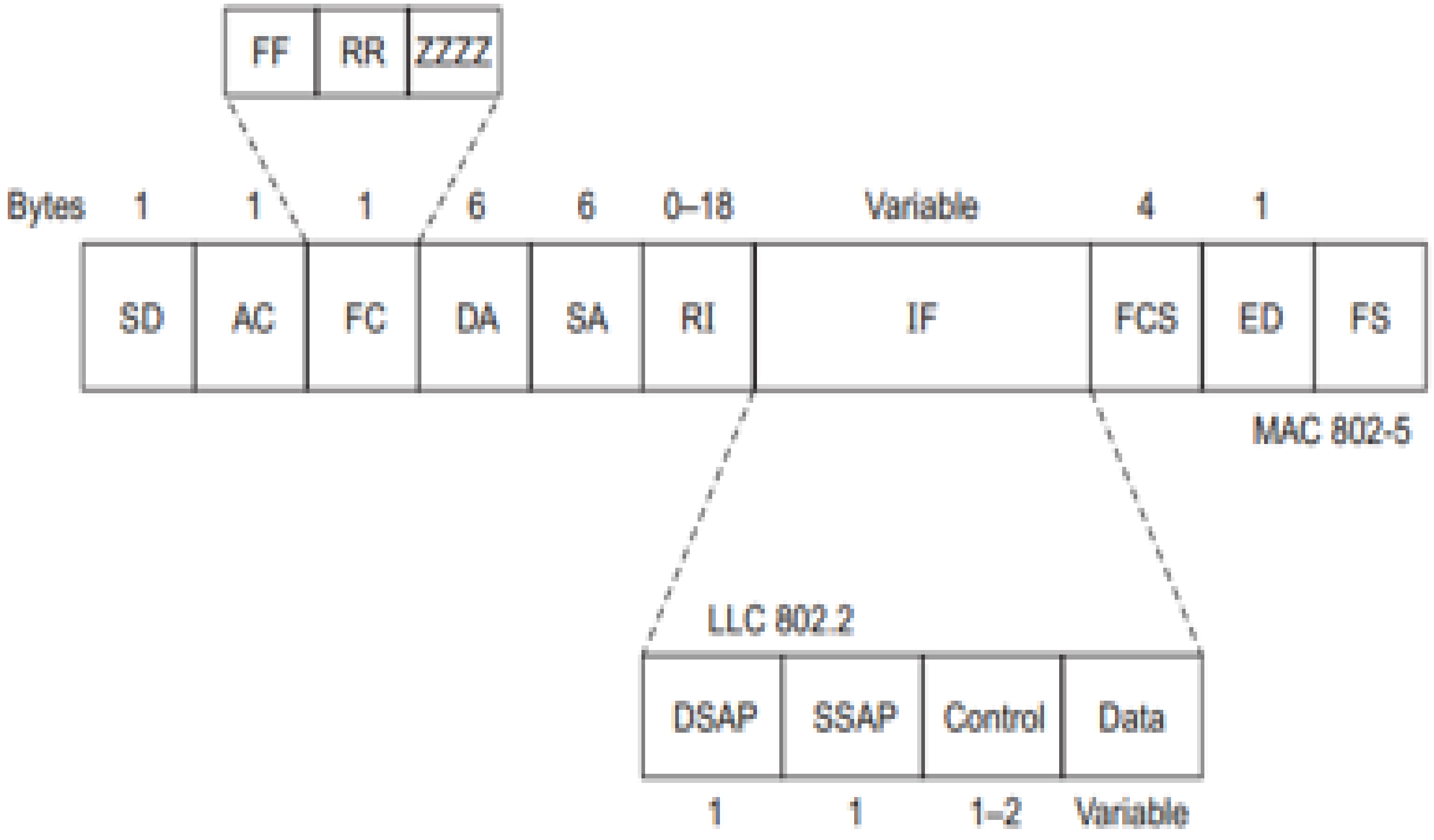
   III. The source releases the token by changing the Token Bit (T bit) to one.

Token Frame format. The token is a three byte frame.



- **Start Delimiter (SD).** It is set to JK0JK000.
- J and K are non-data bits. They are violated Differential Manchester coding.
- The purpose of this pattern is to keep the SD byte (or ED type) from repeating in the information field of token ring frame.
- **End of Delimiter (ED).** It is set to JK1JK10E. E is always set to zero,
- If any station detects an error, it will set E to 1.
- **Access control byte (AC).** It contains three priority bits (PPP or more), a token bit (T), monitor bit (M) and remaining reserved bits.
- The priority bits varies 000 through 111.
- If token bit T = 0, data is in token or if T = 1, no data in token. The Monitor Bit (M) is used to prevent frames from circulating onto the ring.

IEEE 802.5 Frame format



- SD, AC and ED fields are already described.
- FC field is the Frame control (FC). In FC field, FF bits indicate the *frame type*.
- FF = 00 indicates MAC Frame, FF = 01 indicates LLC frame, and FF = 10 and 11 meant for reserved.
- RR set to 00 and they are reserved bits. ZZZZ = 0000 means a normal buffer and ZZZZ = 0001 indicates express buffer.
- DA and SA are destination and source address fields respectively. DA uses the same address format of 802.3.
- Routing information (RI) is optional.
- information field (IF) contains a MAC frame, the frame is called the MAC protocol data unit.
- If IF field contains an LLC frame, the field is called LLC protocol unit (LPDU).
- FCS is frame check sequence for error detection and Frame Status field (FS) have the field of 1 byte.

ASSIGNMENT

- Write some of the link between access and dense level of SDH
- Write about ATM and compare it with SONET and SDH
- Write about DWDM and compare with CDWDM
- Write about CDWM and compare it with FDD
- Write about ad hoc and too write some algorithms
- Write about Li-Fi and camper it with Wi-Fi technology

THANK YOU HAVE A NICE SUMMER

NO ONE IS only A TEACHER NO ONE is only A STUDENT

WE ALL ARE STUDENTS

Having B+ ,not about grade rather
Holding and acting positive attitude.

"THE ROOTS OF EDUCATION ARE BITTER, BUT THE FRUIT IS SWEET. "