RLR sir Note

Date
Page

Introduction

Telecommunication is derived from greek words: 'tele' meaning at distance and 'communication' meaning exchange of information.

What is information?

Anything which can be changed into electrical signal with the help of sensor or transducer is called information.

For example sound or voice ← microphone
picture or video ← light sensor
temperature or heat ← heat sensor
mechanical displacement ← metal sensor

Defination

Any transmission, emission or reception of information by metalled wire, optical fiber or radio waves.

OR.

Exchange of information from one point or points to another point or points by metalled wire or optical fiber or radio waves.

History

1. Fires (smoke signals) } Asia, Africa and America
2. Drum beatings }
3. Semaphore systems, emerged in Europe in the 1790's.

4. Postal service
5. Evolution of telecommunication and concept of telephone exchange or switching system.

Evolution of Telecommunication.

1. Telegraph → message in coded form of electrical sign
   → 1837 – Great Britain
   → 1845 – France
2. Telephone → Voice in electrical signal form
   → USA, 1876, Alaxander Graham Bell.

1. Telegraph

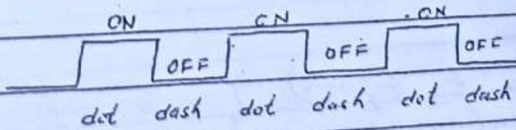
Historically, transmission of telegraphic signal over wires was first technological development in the field of modern telecommunication.

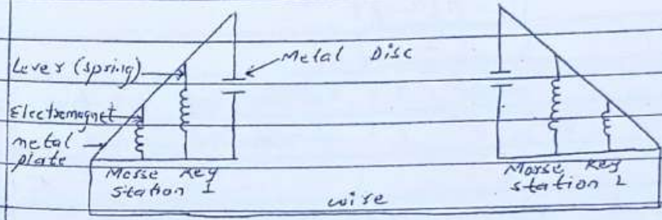
Telegraph was introduced in Great Britain in 1837 and in 1845 in France.

Construction and Operation.

Morse Code

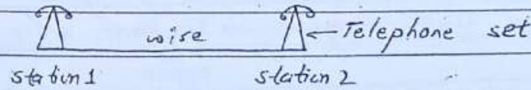
Example:



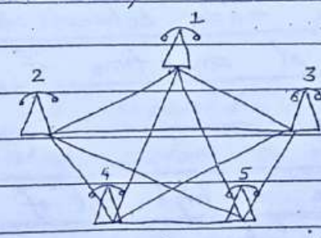


2. Telephone

In March 1876, Alexander Graham Bell of USA demonstrated his telephone set and the possibility of telephony i.e, long distance voice transmission. His demonstration laid the foundation for telephony.



Concept of Telephone exchange or Switching System



Fully connected network

$$\boxed{L = \frac{n(n-1)}{2}}$$

where, n = number of telephone set
L = number of wired links.

Example, For 5 subscribers or telephone set

$$L = \frac{5(5-1)}{2} = 10$$

For 100 subscribers

$$L = \frac{100\,(100-1)}{2} = 4950$$

Terms used.

Asia and Africa → Telephone Exchange
Europe and Australia → Switching System
North and South America → Telephone office or switching office.

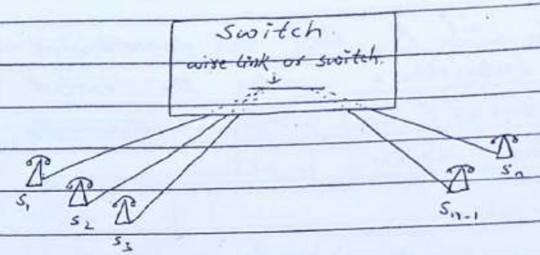
→ Technological progress and more economical.

Advantage

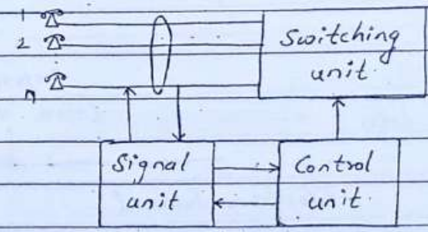
→ connection can be made between two telephone sets or subscribers at any time if they are free.

Disadvantage.

→ very expensive due to high cost of wire link.
→ complications in maintenance.



Modified Basic Block diagram of a Telephone Exchange.



Three Basic blocks.

1. Switching unit
2. Signalling unit
3. Control unit

Signalling is required to draw the attention of the switching system to establish or break a connection

The function of control unit is to connect or release the switch in switching unit with the help of signalling unit.

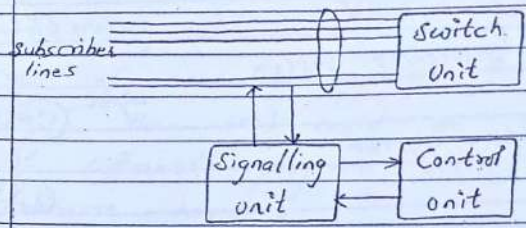
Switch unit : It acts to connect or release the two subscribers under the control of control unit. The control unit acts under the direction of signalling unit.

Classification of switching system.

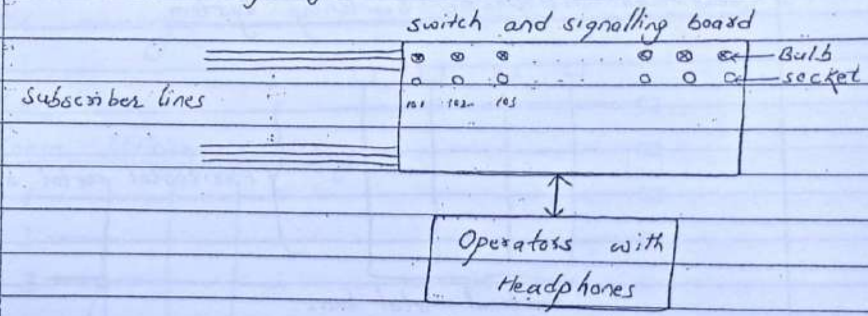
switching system.

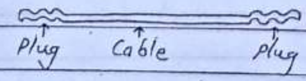
- Manual (less than 1000 subscriber lines)
- Automatic (Above 1000 subscriber lines)
  - Electromechanical
    1. step by step / strowger - Germany
    2. Cross-bar - Ericson Co. - Sweden.
  - Electronic (Stored Program Control)
    - Space Division switching
    - Time Division switching
      - Digital
        - Space Switch(S)
        - Time Switch(T)
        - Combined
      - Analog

Basic block diagram of switching system.



Manual Switching System.





→ Each subscriber is assigned with one bulb and one socket.
→ Every time one had to make a call, s/he needs to call the operator and tell with whom they wanted to make a call. Accordingly, the operator would perform.

Automatic Switching System.

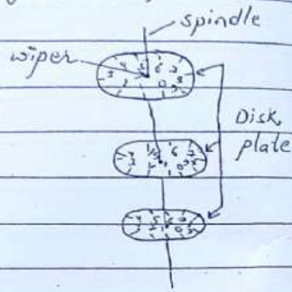
1. Electromechanical Automatic Switching System (Telephone Exchang

i. step-by-step switching system

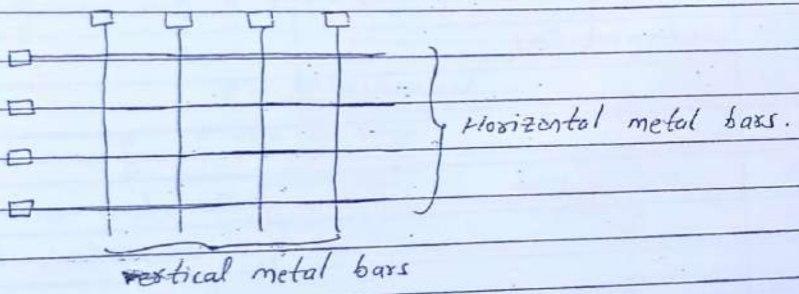
OR,

strowger switching system

– A. B. strowger



ii. Cross-bar Automatic Switching System.



2. Electronic Switching System.

i. With the development of logic circuit, all electro-mechanical switch are replaced by digital switch.

ii. Memory unit is added in control unit. All hardware operations being done by microprocessor which is the main part of control unit. Microprocessor has acted as the interface between hardwares and softwares.

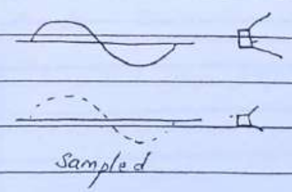
iii. Lot of hardware operation for difficult facilities (more than 100) could be done with the help of different programs stored in memory. No extra hardware required.

Even the operation of the system is being done with the help of operating program present in the memory.

Types

a. Space Division Switching

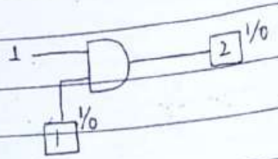
b. Time Division Switching



Digital Telephone Exchange.

i. Space Switch (S)

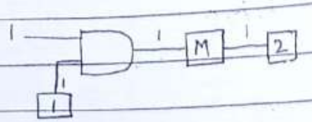
→ Digital switch - with no memory.



ii. Time Switch (T)

→ Digital switch with memory

→ the output can be controlled.



Electronic Telephone Exchange OR,
(Stored Program Control) OR,
Digital Telephone Exchange

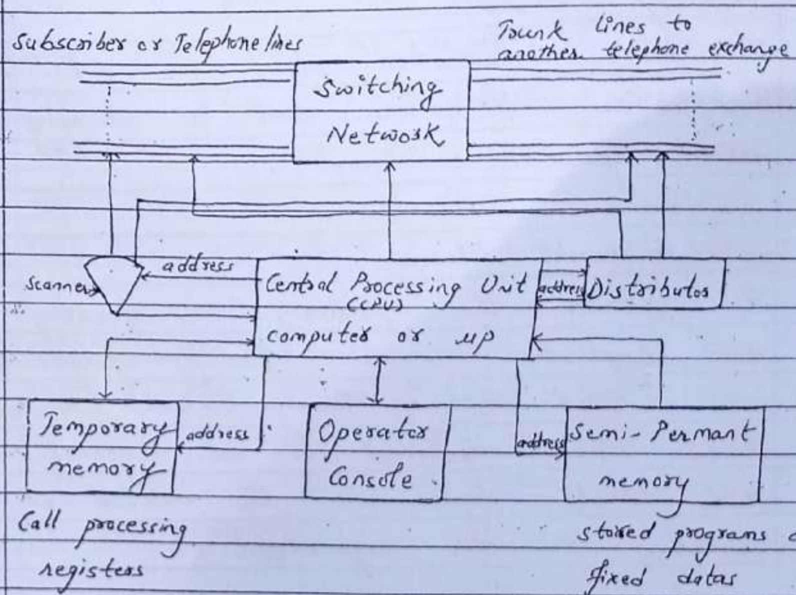


Fig:- Digital Telephone Exchange.

Switching Network.

The switching network provides the means of connecting or disconnecting of telephone lines and trunk lines under the direction of CPU with the existing program in the semi-permanent memory.

Scanned with CamScanner

Memory

1. Temporary Memory
2. Semi-permanent Memory

Temporary Memory

It stores information about the calls in process and other informations that is changing during normal operation.

Semi-Permanent Memory

It contains the program i.e. operating program and application programs (facility programs) which are Read Only Memory (ROM). It also contains fixed datas related to the telephone lines and trunks which can be deleted or written according to the requirement.

CPU

This is the interface between software and hardware. This operates all the hardwares according to the instructions given by the program present in semi-permanent memory according to the information read from the temporary memory. All the temporary informations related to telephone lines and trunks are stored in the temporary memory by CPU after receiving through scanner.

Scanner

The CPU knows the status of telephone lines and trunks by the means of continuous scanning (once in each 100 ms) operation carried out by scanner. After each scan, the CPU writes the information about the lines and trunks in temporary memory. Each line and trunk is equipped with sensing device.

Distributor

Signalling over the lines and trunks are provided by the distributor under the direction of CPU. The CPU tells the distributor when to start or when to stop signalling.

Operator Console.

It controls devices for different controls and observations and used by operating personnel. They are 1. alarm, 2. line and trunk task facility 3. display and 4. prime maintenance.

The operating personnel can also write or erase datas related to lines and trunk in semi-permanent memory.

Numbering Plan

14372257

- 1 — District code
- 4 — Exchange number
- 372257 — Real telephone number

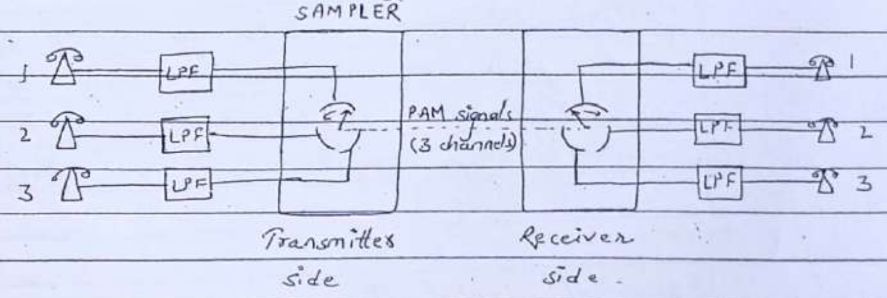
National code → 0

International code → 00

00 977 - 1 - 4372257

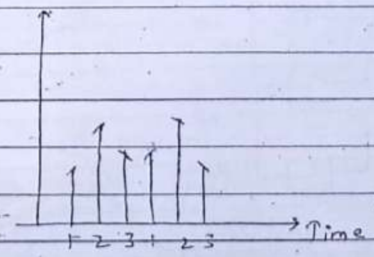
977 — country code, given by ITU

Time Division Multiplexing (TDM)



LPF → Low Pass Filter

PAM → Pulse Amplitude Modulation



VF Band

VF channel : 300 Hz – 3400 Hz

↑

4 KHz VF band.

Sampling frequency : 8 KHz

← F →

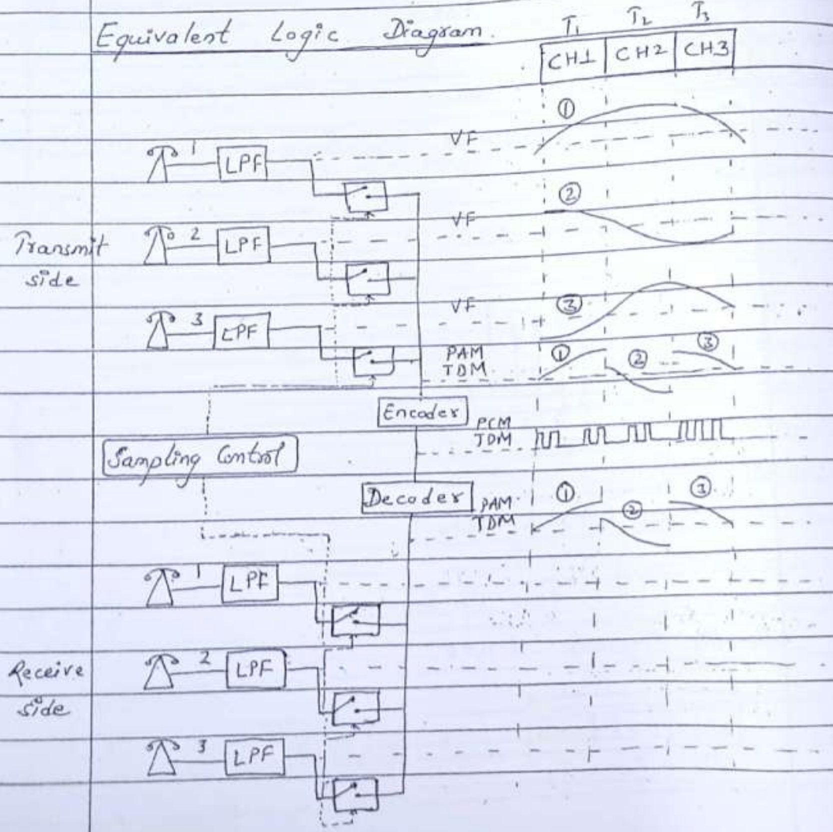
CH1	CH2	CH3
T_1	T_2	T_3

$T_1 = T_2 = T_3 = T$

F = Frame i.e, one complete cycle of switch SW_1 or SW_2.

T = Time slot i.e, time taken by switch SW_1 or SW_2 to pass or to receive signal of one chann[el] during one complete cycle of its movement i.e, during completion of one frame

Equivalent Logic Diagram



American TDM system used in Telecommunication.

T_1 Carrier System

→ North, South and Central America.

← 1 TDM Frame (125 µs) →

Sample 1	Sample 1	Sample 1	Sample 1	
Channel 1	Channel 2	Channel 3-23	Channel 24	Framing
S-bit 7 PCM bits	S-bit 7 PCM bits	168 bit	S bit 7 PCM bits	bit
(8 bit)	(8 bit)		(8 bit)	(1 bit)

Frame 1	Fram 2	Frame 3 to Frame 8000
193 bits	193 bits	Sample 3 to Sample 8000
Sample 1	Sample 2	

Note

VF Band → 4 KHz (300 Hz – 3400 Hz)

Sampling frequency or rate → 8000 cycles per sec.

So there is 8000 frames per sec.

Time for each frame → 125 µs.

24 Telephone channel or VF channel TDM system

8 bits per time slot

i.e, 1 2 3 4 5 6 7 8

1 → S-bit (signalling bit) 1st bit

2 3 4 5 6 7 8 → PCM bits (Information)

Last one bit → Framing bit or Frame Alignment signal

Each Telephone channel for example CH1

$$= \frac{8 \text{ bits}}{\text{sample}} \times \frac{8000 \text{ samples}}{\text{second}}$$

$$= 64 \text{ kb/sec} \quad (\text{channel speed})$$

Line speed = channel speed × 24 channels + framing bit × 8000

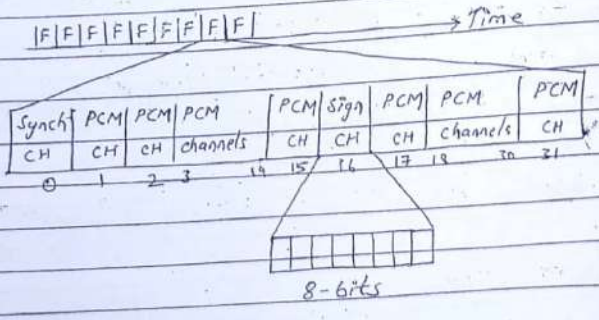
$$= 64 \text{ kb/s} \times 24 + 1 \text{ bit} \times 8000$$

$$= 1.544 \text{ Mb/sec}$$

$$\approx 1.5 \text{ M bit system.}$$

European TDM system used in Telecommunication.

E1 carrier system.



Synch CH (CH 0) → Synchronization or frame alignment signal.

Sign CH (CH 16) → Signalling Channel

CH 1-15 } 30 VF Channels OR,
CH 17-30 } Telephone Channels.

Sampling rate → 8000 cycles/sec i.e, 8000 frames

VF Band or Telephone Channel → 4KHz (30Hz - 3400Hz)

From CH 1-15 and CH 17-30

Each time slot in each frame contains 8-bit PCM.

$$\underbrace{1 \quad 2 \quad 3 \quad 4 \quad 5 \quad 6 \quad 7 \quad 8}_{\text{Information}}$$

Time slot 0 or CH0 in each frame contains 8 bits coded digital signal for frame alignment or synchronization at the receiver side.

Time slot 16 in each frame contains 8-bits digital signals for 30 CH VF channel.

$$\text{Channel speed} = \frac{8 \text{ bits}}{\text{sample}} \times \frac{8000 \text{ samples}}{\text{sec}}$$

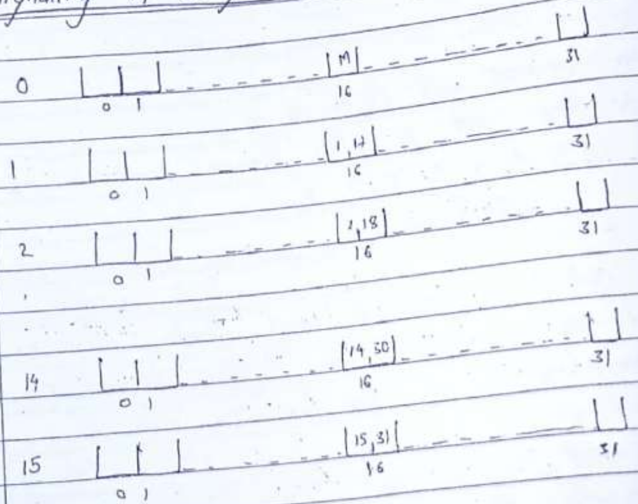
$$= 64 \text{ kb/sec}$$

line speed = synch CH speed + Signalling CH speed + 30 VF channel speed.

= 64 Kb/sec + 64 Kb/sec + 64 Kb/sec × 30 VF channel

= 2.048 Mb/sec.

Signalling in European TDM system.



Group of 16 frames is known as multi-frame.

Time slot no. 16 in '0' frame of a multi-frame represents the Multiframe Alignment signal and consist of coded 8-bit digital signals.

Time slot no. 16 in frame 1 of a multiframe.

1 2 3 4 5 6 7 8

Signalling bits for channel 1 VF CH

Signalling bits for channel 17 VF CH

upto frame 15 of a multiframe.

Approach to PCM switching

Digital switch architecture } made up of two elements called T(time) and S(space)

T for Time Division Switching
S for Space Division Switching

AT and T company (American Telephone and Telegraph Co.) - USA	No. 4 ESS (Electronic Switching System)	- TSSSST swi
	No. 3 EAX (Electronic Automatic Exchange)	- SSTSS s
Northern Telecom - Canada	DMS	- TSTS

Modern Digital Telephone Exchange → TST or STS switch
→ very large capacity upto 1,00,000 telephone lines

Time Switch (T) or Time Slot Interchanger (TSI)

$N = 30$

Incoming Frame | 1 | 2 | 3 | 4 | . . . . . | N | Memory | 1 | 2 | 3 | 4 | . . . . . | N | Outgoing Frame

Time Slot interchanger involves moving the data contained in each time slot from the incoming bit stream to an outgoing bit stream but with a different time slot arrangement in accordance with the destination of each time slot.

Thus time switch can transfer the information in one time slot to the another time slot but in same TDM line.

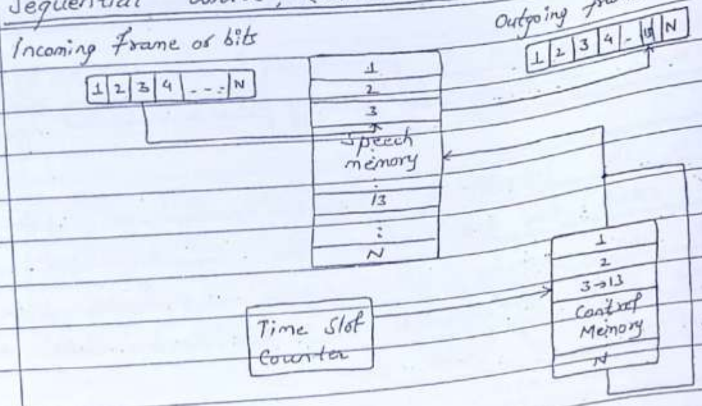
Basic Functional Blocks of a Time Switch (T)

1. Memory for speech.
2. Memory for control.
3. Time slot counter or processor

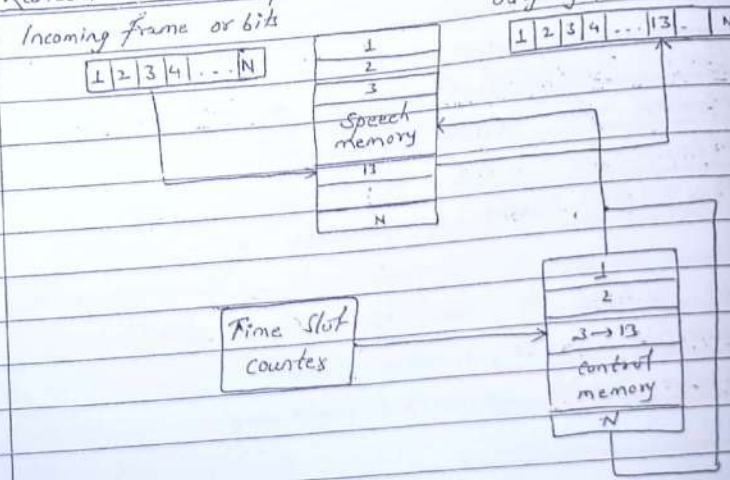
Two options in handling the T switch

1. Sequential write, random read.
2. Random write, sequential ~~writ~~ read.

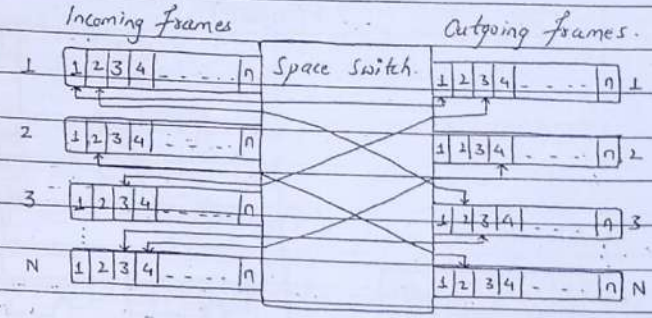
1. Sequential write, Random read.



2. Random write, Sequential read.



How can we increase a switch's capacity and space switch (s)?



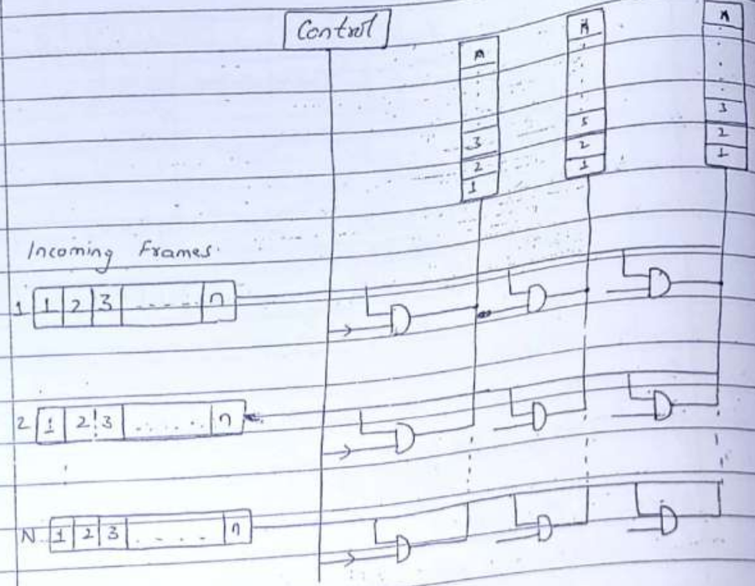
In space switch the switch capacity can be increased by increasing the TDM lines.

Working Principle.

Information in any time slot of an incoming TDM lines can be transferred to the same or any other outgoing TDM lines but in same time slot.

Thus TDM lines can or cannot be changed but the time slot will be the same incase of space switch(s).

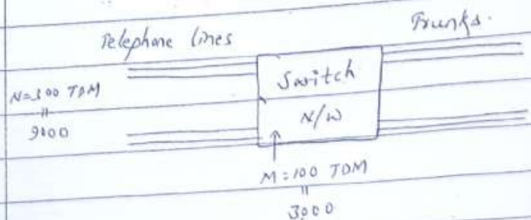
Logic Diagram



$N = M$, the switch is not blocking

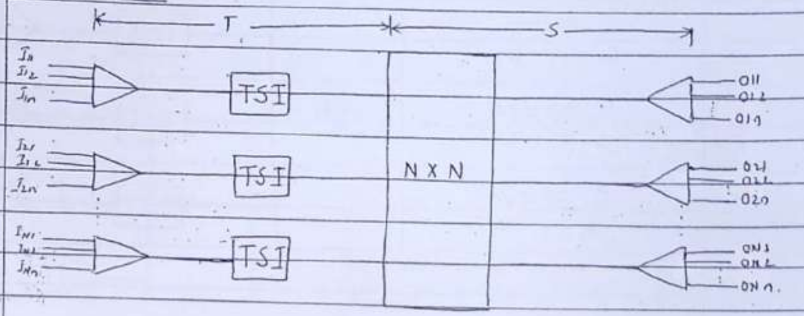
$N > M$, the switch is blocking

$N < M$, the switch expands.



Two stages Time-space (TS) or Space-Time (ST) switch.

1. TS switch.



Working Principle.

$I_{27} \longrightarrow O_{16}$

$I_{27} \xrightarrow{T} I_{26} \xrightarrow{S} O_{26}$

Drawback

$I_{27} \longrightarrow O_{19} \Rightarrow I_{27} \xrightarrow{T} I_{29} \xrightarrow{S} O_{19}$ ✓

$I_{23} \longrightarrow O_{49} \Rightarrow I_{23} \xrightarrow{T} I_{29}$ X X

Both the samples originated from the same TDM line with different time slots of inlet going to the same time slot of different TDM lines in the outlet only one call will pass and others blocked.

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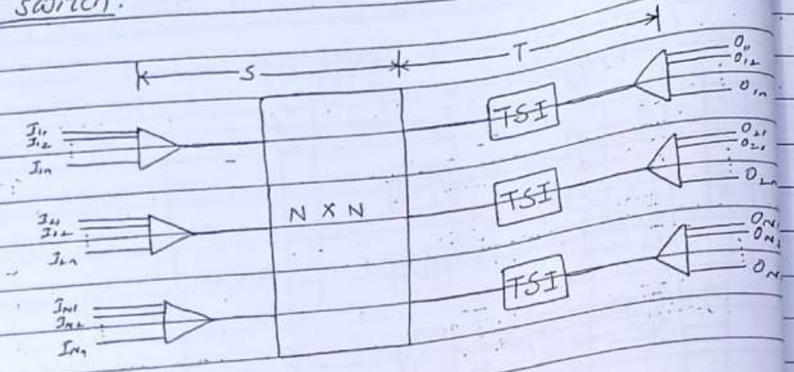
ST switch.



Working Principle

$I_{27} \longrightarrow O_{19}$

$I_{27} \xrightarrow{S} I_{17} \xrightarrow{T} O_{19}$

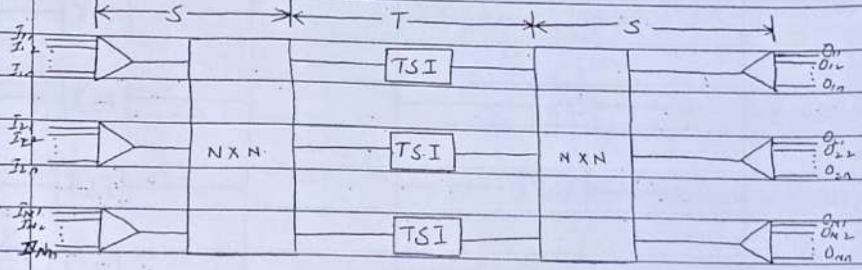
Drawbacks

$I_{75} \rightarrow O_{96}$, $I_{85} \rightarrow O_{92}$ $\Big\}$ $I_{75} \xrightarrow{S} I_{95} \xrightarrow{T} O_{96}$ ✓, $I_{85} \xrightarrow{S} I_{95}$ ✗

Both samples in the same time slot of different TDM lines in the inlet going to the different time slots of the same TDM lines in the outlet. Only one call will be passed, others will be blocked.

Three stage TST or STS switch.

STS switch.



This switch removes the drawbacks of ST and TS switches.

~~Drawbacks of TS switch~~

$I_{27} \longrightarrow O_{39}$, $I_{23} \longrightarrow O_{49}$ $\Big\}$ $I_{27} \xrightarrow{S} I_{37} \xrightarrow{T} I_{39} \xrightarrow{S} O_{39}$, $I_{23} \xrightarrow{S} I_{43} \xrightarrow{T} I_{49} \xrightarrow{S} O_{49}$

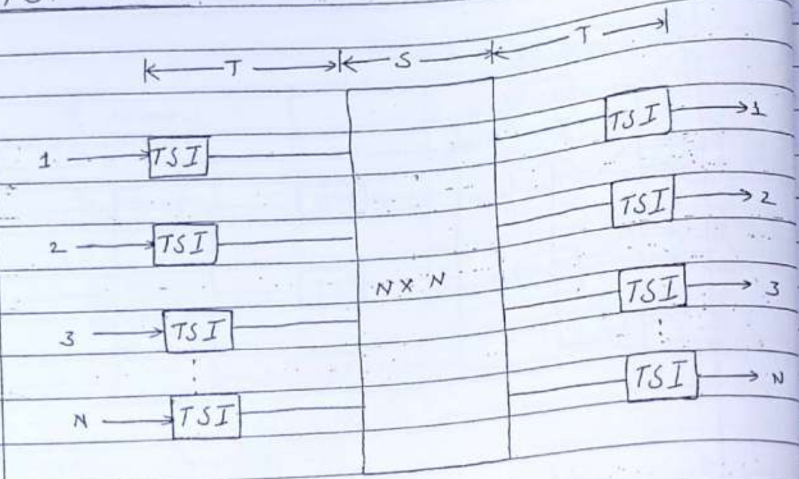
Drawbacks of TS switch.

$I_{27} \longrightarrow O_{19}$, $I_{23} \longrightarrow O_{49}$ $\Big\}$ $I_{27} \xrightarrow{S} I_{17} \xrightarrow{T} I_{19} \xrightarrow{S} O_{19}$, $I_{23} \xrightarrow{S} I_{53} \xrightarrow{T} I_{59} \xrightarrow{S} O_{49}$

Drawbacks of ST switch.

$I_{75} \longrightarrow O_{96}$, $I_{85} \longrightarrow O_{92}$ $\Big\}$ $I_{75} \xrightarrow{S} I_{95} \xrightarrow{T} I_{96} \xrightarrow{S} O_{96}$, $I_{85} \xrightarrow{S} I_{25} \xrightarrow{T} I_{22} \xrightarrow{S} O_{92}$

TST switch.



Removes the drawbacks of TS and ST switches,
same as STS switches.

Technical structure of a Telephone office or Exchange

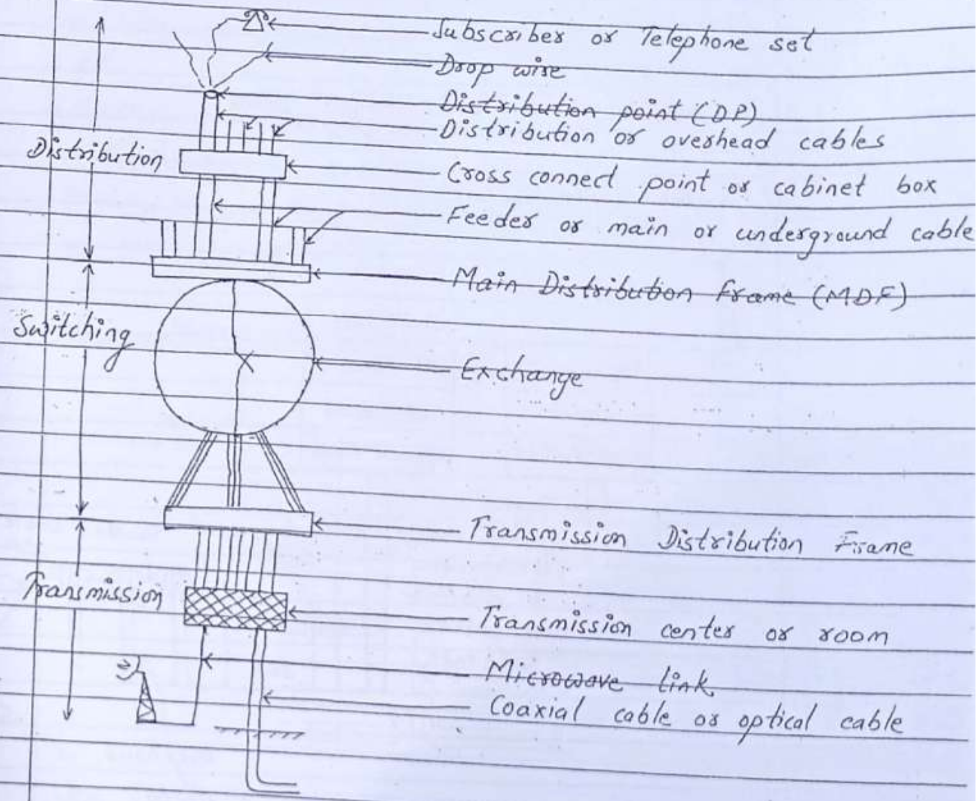


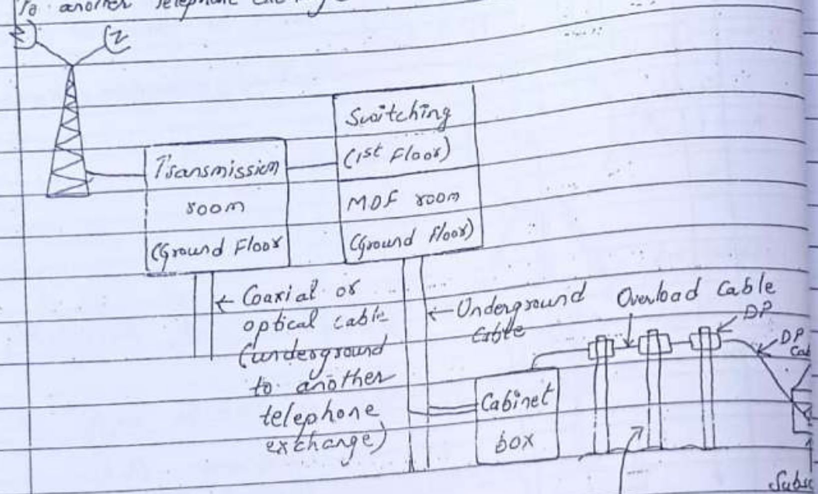
Figure : Technical structure of a Telephone office or Exchange.

The technical structure of a telephone office or exchange indicates the path that must be set-up to interconnect two telephones and the equipments used for this.

Scanned with CamScanner

Three sections.

1. Subscriber loop or subscriber line plant or distribution
2. Switching plant
3. Transmission or Interchange plant.



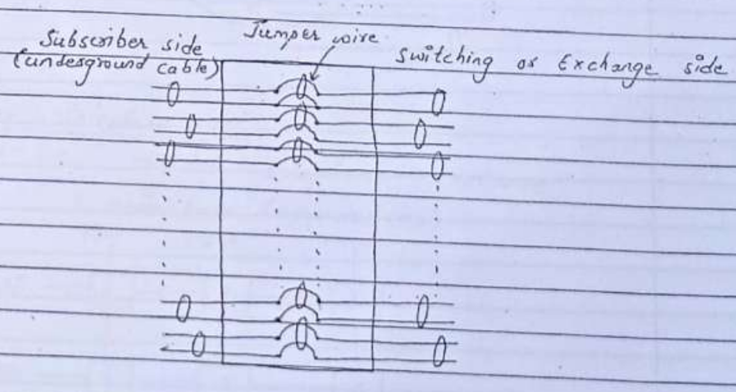
Distribution or Subscriber loop or line plant

The subscriber loop or line plant is the pa[rt] between a subscriber's telephone set and his local switching office or exchange.

This section consists of

- subscriber telephone set
- a 'drop wire' (pair of wires) per subscriber
- → DP
- → distribution or overhead cables
- → cabinet box
- → Main or underground or feeder cables
- → a main distribution frame (MDF)

Main Distribution Frame (MDF)



capacity → 25 pairs to 500 pairs.

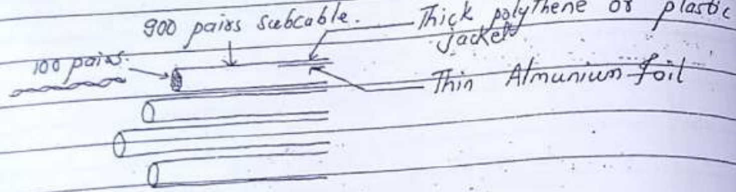
This is a physical interface between the underground cable (coming from subscriber side) and telephone exchange (cable coming from exchange side). They are connected by flexible wire known as jumper wire in MDF.

Underground or Main or Feeder Cable.

Each grouping the pairs from the several distribution cable 900 pairs, 1200 pairs, 1800 pairs, 2400 pairs, 3600 pairs cable.

Construction

3600 pairs cable

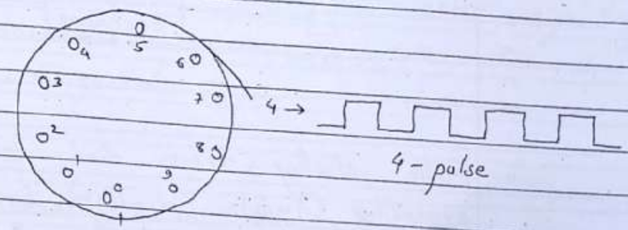


DTMF (Dual Tone Multi-frequency) Telephone set.

OR, Touch Tone Telephone Set

OR, Rotary dial or Pulse dialing Telephone set

→ used in electromechanical Telephone exchange.



DTMF Telephone set

used in digital telephone exchange

upper band frequencies (Hz)

Lower Band frequencies (Hz)	1209	1336	1477
697	1	2	3
770	4	5	6
852	7	8	9
947	#	0	*

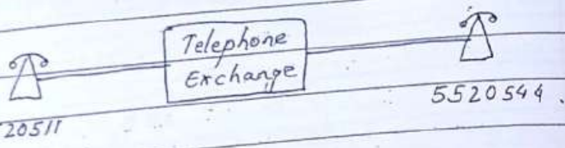
How to dial '4' ?

→ combination of

Lower Band frequency → 770 Hz and

Upper Band frequency → 1209 Hz

in sine wave form.



Pulse dialing (Rotary dial telephone) is limited signalling (number dialing) between exchange a the subscriber and no signalling is possible end-to-end i.e, between two subscribers.

End-to-end signalling is possible in case o DTMF telephone. This is possible only beca of signalling done in VF band frequency.

In DTMF telephone set two frequencies are mixed from a set of seven frequencies (co from upper band and other from lower frequency band) as shown in figure

The need for DTMF signalling frequencies t be in the VF band brings with it the problem of vulnerability to "talk-off" which means that the speech signals may be mistaken for DTMF signals and unwanted control actions such as terminating a call may occur.

Another aspect of "talk-off" is that the speech signal may interface with the DTMF signalling if the subscriber happens to talk while signalling is being attempted.

Design considerations in order to protect from "talk-off"

i. Choice of code

ii. Band seperation

iii. Choice of frequencies.

iv. Choice of power levels-

v. Signalling duration.

vi. Human factors and mechanical aspects.

i & ii. Choice of code and Band seperation.

Simple single frequency structures are prone to easy installation as they occur frequently in speech or music. Hence, some form of multi-frequency code is required. In order to improve the performances, two measures are adopted.

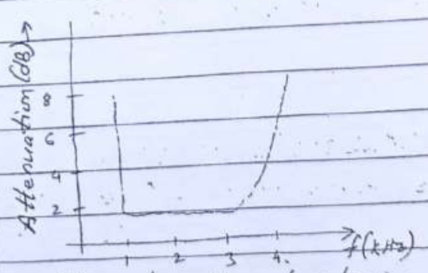
Firstly, combining two frequencies out of seven frequencies or 8 frequencies depending upon the number of code word desired.

Secondly, the choosen frequencies are placed 2 seperate bands and a restriction is applied such that one frequency from each band is choose to form a code.

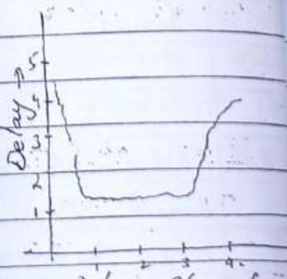
When multiple frequency are present in speech signal, they are closely spaced.

Band seperation of DTMF frequencies reduces the possibility of speech being able to produce DTMF combination.

iii. Choice of frequencies.



a. Attenuation Characteristics of VF band



b. Delay Characteristics of VF band.

Going through the attenuation and delay characteristics graph frequency range from 700 Hz to 1700 Hz have minimum attenuation and delay characteristics in case of travelling VF signalling from subscriber to the exchange or from subscriber to subscriber. The variation is also minimum. So, band frequencies are selected between this range.

iv. Choice of Power levels.

A nominal value of 1 dB above 1 mW is provided for the telephone set for the combined signal power of the 2 frequencies. The attenuation increases with the frequency. The upper band frequencies are transmitted at a level 3 dB than that of the lower band frequency.

v. Signalling Duration.

The probability of "talk-off" can be reduced by increasing the duration of the test applied to a signal by the receiver before accepting the signal as valid.

vi. Human Factors and Mechanical aspects.

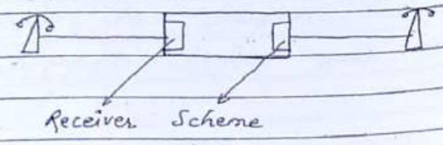
3/8" square buttons, seperated by 1/4".

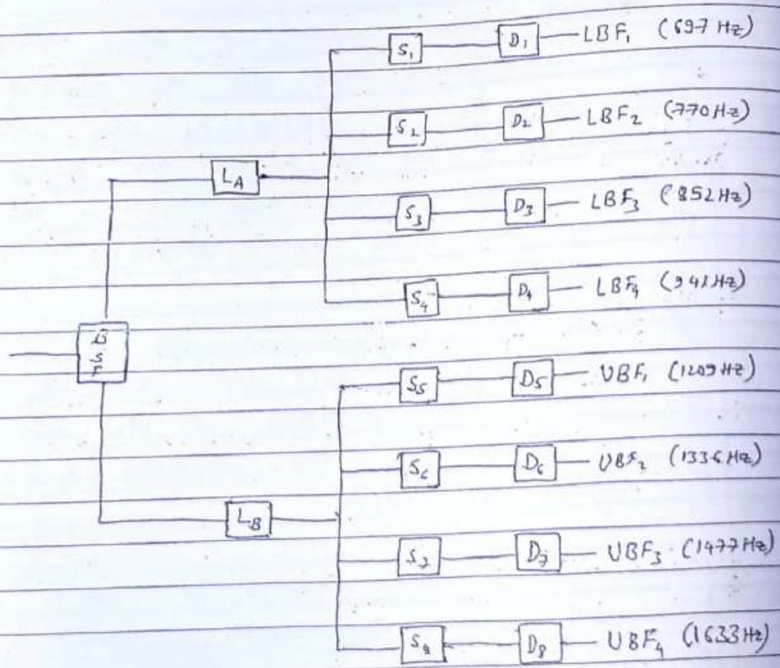
1/8" stroke length, 100 gm force at the bottom of stroke.

Scanned with CamScanner

4x3 array with the digits 1, 2, 3 in the top view and zero in the middle of the last row.

#, * → code for facilities.





BSF → Band Seperation Filter

LBF → Lower Band Frequency

UBF → Upper Band Frequency

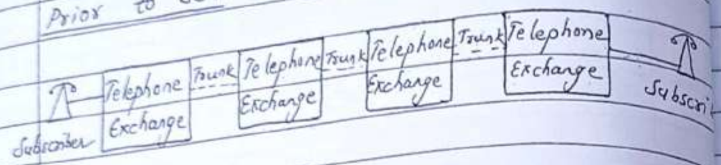
D → Detector

L → Limiter

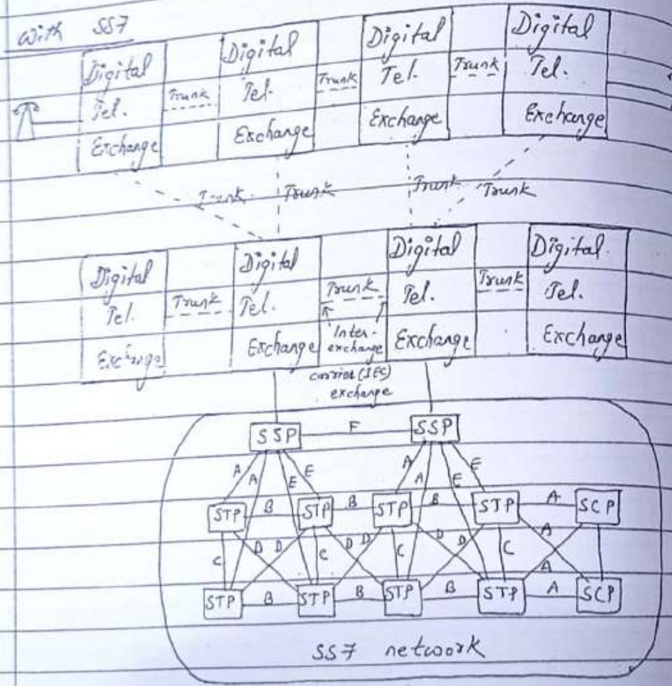
S → Selector Circuit

Common Channel Signalling System #7 (SS7) and network.

Prior to SS7.



With SS7



never receives the call from the originating IEC exchange.

If the called line is idle, the SS7 network issues signalling messages to each telephone exchange through IEC exchange that will be used to connect the call.

Each exchange will communicate over the SS7 network via IEC exchange and identify the trunk circuits to be used between each exchange. These trunk circuits are reserved for the call.

A signal is sent to the terminating exchange by the SS7 network via IEC exchange, instructing it to ring the called line.

The caller receives ring-back tone from the originating exchange under the direction of SS7 network via IEC exchange. When the called party answers the call, the reserved trunks are then connected and the voice circuit path is completed between the calling and called parties.

Prior to SS7
When a long distance call was made, the was connected exchange by exchange through long distance telephone network.

On calls to a busy telephone, the call be established over the network and tie numerous trunk circuits in the telephone or PSTN (Public Service Telephone Networ just to return a busy signal from the end telephone exchange.

With SS7.
When a call is placed over the long dist network, the inter-Exchange Carrier (IEC) Ex will launch a query over the SS7 network to establish a call set up path and to the called number.

If the called number is busy, a busy ton or signal is returned to the calling part from the originating exchange under the dire of IEC exchange.

When a line is busy, the PSTN trunk

SS7 makes efficient use of trunk circuits in telephone network.

Common Channel Signalling technique, using a seperate network of computers to send and receive messages as signals, has been adopted by the International Community as CCIS #7 and is referred system 7 (SS7).

SS7 network consists of following components.
1. Signalling Transfer Point (STP)
2. Signal Control Point (SCP)
3. Service Signalling Point (SSP)

1. Signal Transfer Point (STP)
It is a packet based switch. It receives and sends message signal in packet form to proper distribution. It also performs specialized routing function.

2. Signal Control Point (SCP)
This node acts as a database which provides the information for the advanced call in the network.

Service Signalling Point (SSP)

This node is the end exchange. equipped with SS7 enabled software and having signal links. Each IEC exchange will have SSP to connect with SS7 network.

SSP behaves like an exchange for performing different functions like starting the transmission as well as terminating of the link in the network. SSPs are connected with other STPs and SCPs in SS7 network.

A Link → Access Link → used to connect SSP or SCP to STP

B Link → Bridge Link → used to connect two STPs.

C Link → Cross Link → used to connect paired STPs for reliability.

D Link → Diagonal Link → used to connect two STPs diagonally & similar to B link

E Link → Extended Link → used to connect SSP with the remote STP which cannot be connected by A link because of too far.

F Link → Fully Associated Link → used two connect two SSPs

Signalling Tones

A number of signalling functions are involved in establishing, maintaining and releasing a telephone connection.

Five subscriber related signalling functions are performed in Automatic Telephone Exchange.

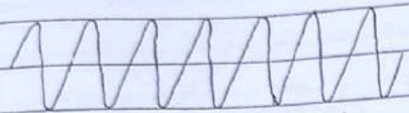
1. Respond to the calling subscriber to obtain the identification of the called party → Dial Tone.

2. Inform the calling subscriber that the call is being established → Ringback Tone

3. Ring the bell of called party → Ringing Tone

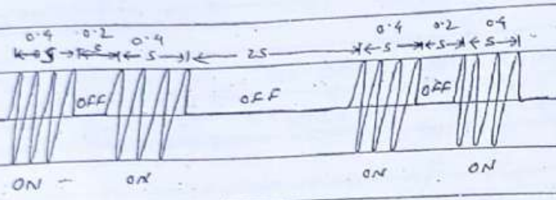
4. Inform the calling subscriber, if the called party is busy → Busy Tone

5. Inform the calling subscriber, if the called party line is unobtainable for some reason → silent or dial Tone.



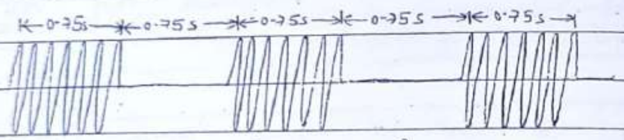
33 or 50 or 400 Hz continuous (sine wave)

a. Dial Tone



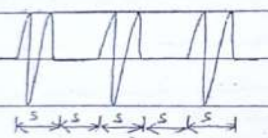
400 Hz (sine wave)

b. Ringing Tone



400 Hz (sine wave)

c. Busy Tone



400 Hz

d. Call-in progress Tone.

Numbering Plan for PSTN or Landline Telephone System

Rec. E. 163 (ITU) old version telephone Numbering Plan
→ for landline only.

Rec. E. 164 (ITU) new version telephone Numbering Plan
→ for landline and mobile.

Rec. E. 163 (ITU) old version telephone Numbering plan.

Prefix Country Code (cc) → Area Code — Exchange — Customer (Customer Local Number)

Area Code + Exchange → National Destination Code

Area Code + Exchange + Customer → National number

Example.

Nepal	00	977	14 (Kath)	37	2257
UK	00	44	171 (London)	234	5678

National Call → 0
International Call → 00
} not counted in telephone numbering plan.

National Destination Code (NDC) → Area Code + Exchange

National Significant Number (NSN) → (12 - Country Code digit)
→ (12 - n digits i.e. country code digit)

Customer dialing digits → Max. 12 digits.

National Number Digits → Area Code + Exchange + Customer

World Telephone Numbering Zone.

Zone 1.
Canada, USA, Jamaica, Barbadas, Bermuda,
Grenada, Monserat, Saint Lucia, etc.

Zone 2
Egypt → 20, Morocco, Algeria, Tunisia, Senegal = 220
Angola, Greenland

Zone 3 and 4.
Greece = 30, Belgium = 31, France 33
Portugal, Bulgaria, UK, Germany

Zone 5.
Fukland = 500, Haiti = 509, Peru = 51
Cuba, Brazil, Oruguay.

Zone 6.
Malaysia = 60, Australia = 61, Thailand, Brunei, Fiji,
Marshall Ireland.

Zone 7
USSR--

Zone 8
Japan, Korea, Hongkong = 850, China, Bangladesh

Zone 9
India, Nepal, Burma, Turkey, Lebanon

ITU E.164 New Version Telephone Numbering Plan.

Prefix. Country Code (CC) | Area Code | Customer Local Number
(1 to 3 digits) | (1 to 4 digits) | Exchange | Customer
← National Significant Number →
← Max. 15 digits →

Country Code (CC) = n (1 to 3 digits)
National Destination Code (NDC) → Area Code + Exchange
Subscriber Number or Customer Local Number → Exchange + Customer
Area Code → 1 to 4 digits i.e, x digits
National Significant Number (NSN) = (15-n) digits where 'n' is CC
1 to 3 digits

Prefix
National Call → 0
International Call → 00 } not counted in telephone numbering plan.

Traffic Engineering

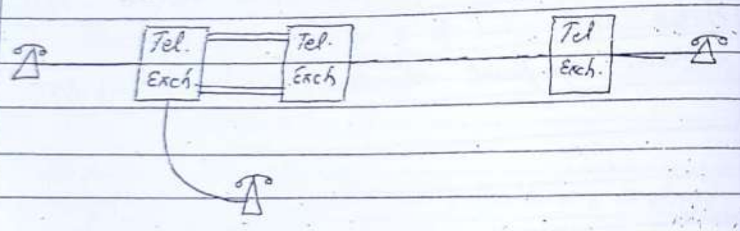
Introduction

Traffic engineering provides the basis for the analysis and design of telecommunication network

Server

→ all common equipment involved in connecting all telephone subscriber to another subscriber in telephone network.

- digital receivers
- interstage switching links.
- call processors
- trunk lines between the exchanges.



Telephone Network.

The load or the traffic pattern on the network varies during the day with heavy traffic at certain times and low traffic at other times. This can be shown by following figure.

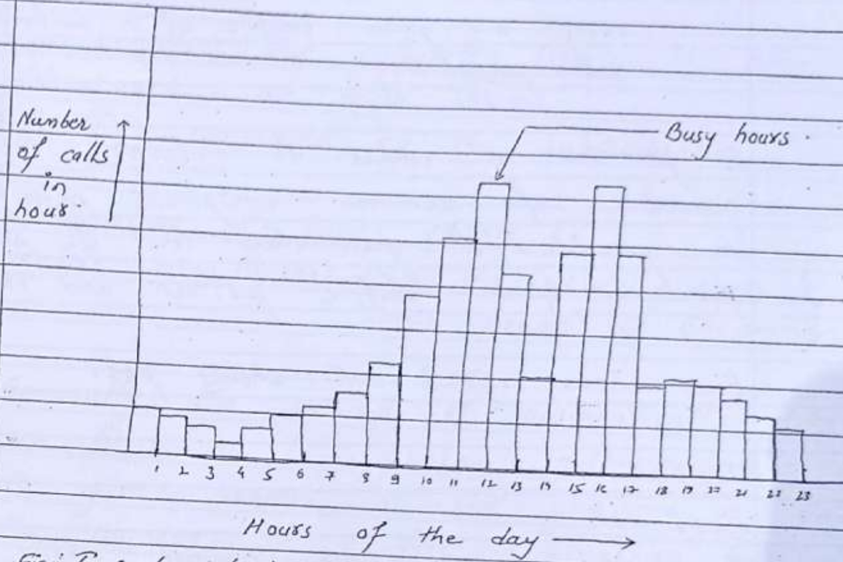


Fig: Typical telephone traffic pattern on working day.

Task
↓
Designing cost effective networks that provides
↓
Quality of service under varied traffic conditions
↑
This demands a formal Scientic basis.
↓
Provided by traffic engineering
↓
Traffic engineering analysis enables one to determine the ability of a telecom network to

carry a given traffic at a particular loss probability.

It provides a means to determine the quantum of common equipment required to provide a particular level of service for a given traffic pattern and volume.

Busy hour → 70% call should pass.

Traffic Parameters.

Busy Hour

Continuous one hour period wholly in the time interval concerned, for which the traffic volume or the number of call attempts is greatest.

Call Completion Rate (CCR)

$$CCR = \frac{\text{Number of successful calls}}{\text{Number of call attempts}}$$

Busy Hour Call Attempt (BHCA)

Number of call attempts in busy hour.

Two methods for calculating Traffic in Telecom N/w.

1. Observation of busy servers.
2. Traffic generated by the subscribers.

1. Observation of busy servers.

$$\text{Traffic intensity } (A_0) = \frac{\text{Period for which a server is occupied}}{\text{Total period of observation}}$$

Generally, the period of observation is taken as one hour. A_0 is generally dimensionless. In order to give respect to the engineer A.S. Erlang, the pioneer of traffic engineering calculation, the unit of A_0 is known as Erlang.

A server is said to have one erlang of traffic if it is occupied for the entire period of observation.

Examples

1. In a group of 10 servers, each is occupied for 30 minutes in an observation interval of two hours. Calculate the traffic carried by the group.

⇒ $$\text{Traffic carried per server} = \frac{\text{occupied duration}}{\text{observation duration}}$$

$$= \frac{30 \text{ minutes}}{2 \times 60 \text{ minutes}}$$

$= 0.25$ Erlang.

∴ Total traffic carried by the group of 10 servers

$= 10 \times 0.25$

$= 2.5$ Erlang.

2. A group of 20 servers carry a traffic of 10 erlangs. If the average duration of a call is 3 minutes, calculate the number of calls put through by a single server and the group as a whole in an one-hour period.

⇒ Traffic per server $= \frac{10 \text{ erlangs}}{20 \text{ servers}}$

$= 0.5$ erlang.

i.e, a server is busy for 30 minutes in one hour observation.

∴ number of calls put through by one server

$= \frac{30 \text{ min}}{3 \text{ min/call}}$

$= 10$ calls

∴ Total no. of calls put through by 20 servers

$= 20 \times 10$

$= 200$ calls.

2. Traffic generated by the subscriber.

In this case, the traffic intensity is a call-time product.

i.e,

$A_0 = C \cdot t_h$

where, C = Average call arriving rate

t_h = Average holding time per call.

Note.

C and t_h both should be expressed in like time units. If C is calls/min then t_h is min/call.

1. Example

Over a 20 minutes observation interval, 100 subscribers initiates call. Total duration of the calls is 9600 seconds. Calculate the load offered to the network by the subscribers and the average subscriber traffic.

⇒ Mean arrival rate, $C = \frac{100}{20} = 5$ calls/min.

Mean holding time, $t_h = \frac{9600/60}{100}$
$= 1.6$ min/call

$\therefore$ offered load $A_0 = C \times t_h$
$= 5 \times 1.6$
$= 8E$

Average subscriber traffic $= \frac{8}{100}$
$= 0.08E$

Overload Traffic

Two ways to handle overload traffic.

In case of telephone network, overload traffic is rejected without being serviced.

Incase of computer n/w, the overload traffic is held in a queue until the network facilities become available.

Incase of telephone network the overload calls are lost → Loss system
a. Grade of service (GOS)
b. Blocking Probability

Incase of computer network the calls are delayed → Delay System
Queing model or theory

Loss System.

a. **Grade of Service**

It is defined as the ratio of lost traffic to offered traffic

i.e., $GOS = \frac{A - A_0}{A}$

where, A = offered traffic
A_0 = carried traffic
$A - A_0$ = Lost traffic

Smaller the value of GOS, better is the service

Generally GOS is called as Call Congestion or Loss Probability.

b. **Blocking Probability**

It is defined as the probability that all the servers in a system are busy.

The blocking probability is also called time congestion.

IP telephony

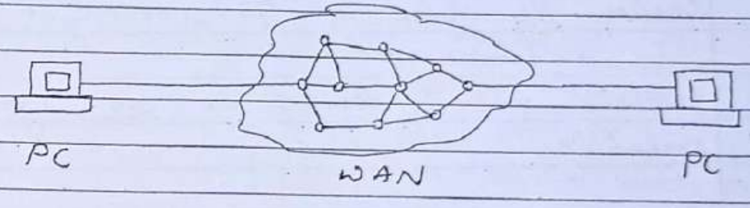
IP telephony is also called as VoIP Telephony, soft phone or SIP phone.

VoIP → Voice over Internet Protocol

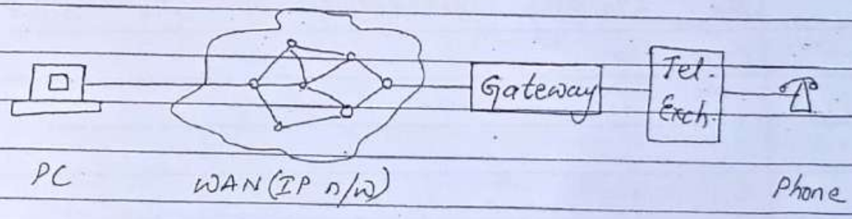
SIP → Session Initiation Protocol.

IP telephony (Basic VoIP Phone)

1. PC to PC.
   - needs a PC with a sound card
   - IP telephony software.



2. PC to Phone
   - needs a gateway that connects IP network to phone network.



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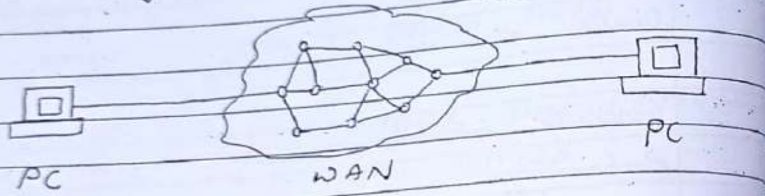

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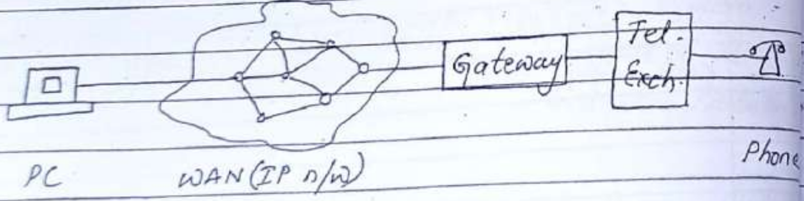

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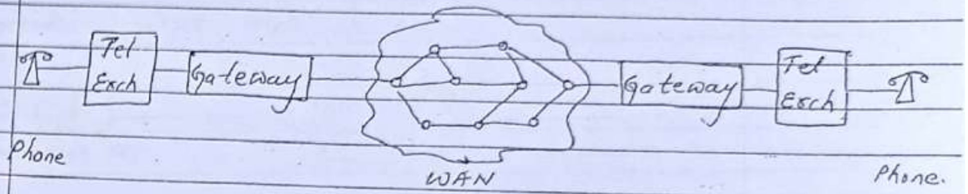
3. Phone to Phone



- needs more gateways that connects IP network
- IP network → Internet

VoIP (Voice over Internet Protocol)

- Transmission of voice over internet i.e, WAN

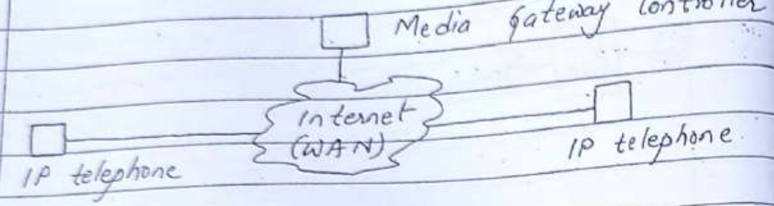
VoIP converts voice into data and sends the voice packets over the network (WAN). These packets get mixed with other voice and data packets and are reassembled into voice by an end point device (telephone). The result is a telephone call.

Working

- continuous sampling audio
- convert each sample to digital form
- send digitized stream across internet (WAN) in packets.
- convert the stream back to analog for playback.

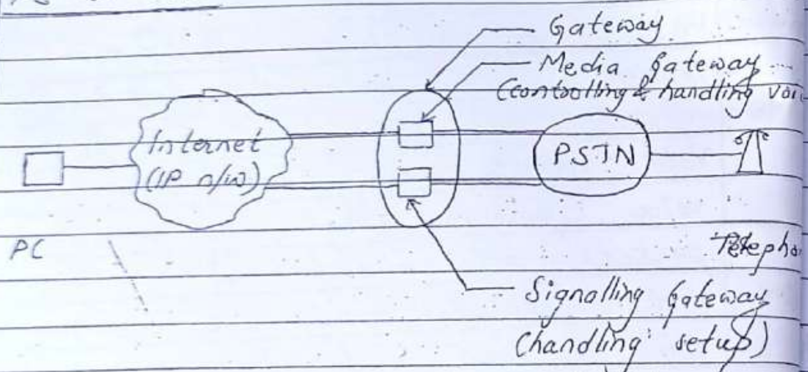
Scanned with CamScanner

A basic IP telephone system.



Interconnection with PSTN (Public Switched Telephone Network)

PC to PSTN



IP telephone system needs to interoperate with PSTN

Two additional components needed for such interconnection

i. Media Gateway
- translates and controls audio between IP network and PSTN

ii. Signalling Gateway
- translates signalling operation
  - Supervisory signal
    - Hook - off
    - Hook - ON

    } Handset

- set up signalling
  - ringing tone
  - ringback tone
  - busy tone
  - dial tone

Signalling Protocol

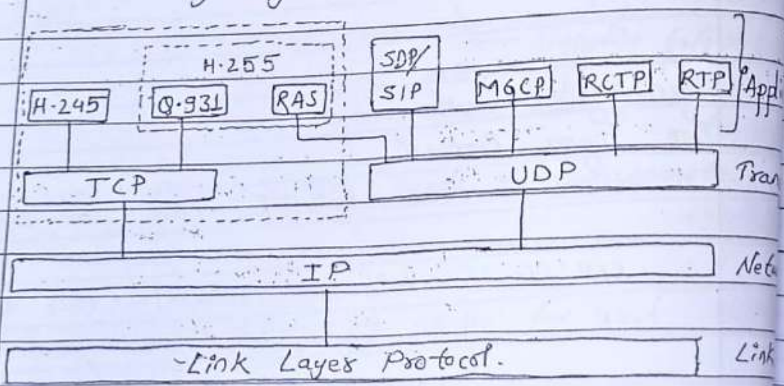
Two major Protocols
- H.323 invented by ITU (old and difficult to program)
- SIP (Session Initiation Protocol) → (easy to program and latest and presently in use)

H.323 characteristics

H.323 consists of a set of protocols th[at] work together to handle all aspects of communication, including

- transmission of a digital audio phone call
- signalling to set up and manage phone call
- allows transmission of video and data while a phone call in progress
- sends binary message
- incorporates protocols for security
- uses a special hardware multipoint control unit for conference calls.
- defines servers for address resolution, authentication, accounting features, etc

H.323 layering (TCP/IP)



TCP → Transport Control Protocol
Reliable, Virtual circuit
(connection, data, email)

UDP → Unreliable Datagram Protocol
Unreliable, datagram
(video and audio)

H.225 → call control signalling
H.245 → control channel signalling and media control.
Q.931 → ISDN signalling
RAS → Registration, Admission, Status.

RTP (Realtime Transport Protocol)

- used to transport real-time data, such as voice or video.

This is an unreliable protocol built on top of the UDP protocol that doesnot generate delivery of packets, but which has little overhead.

RTCP

- Realtime Transport Control Protocol
- It is used to report or control the performance of a particular RTP transport session.

Scanned with CamScanner

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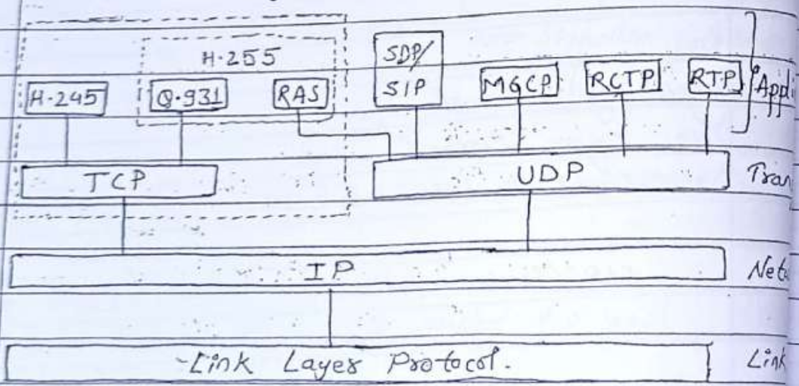

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H.323

An ITU-T standard for handling video, da[ta]
and voice call information.

This protocol is actually an umbrella for
multiple protocols each responsible for
different items like H.225, H.245, etc.

Telephone Number Mapping and Routing.

* How should users be named?
- PSTN follows ITU standard E.164 for phon[e] numbers
  i.e, 1-613-123-4567
- SIP uses IP addresses i.e, sip:smith e.uottowa.

* In an integrated network (PSTN + IP), two problems defined
- Locate an user
- find an efficient route to the user.

Two protocols for the above.
- ENUM : E.164 Number
- TRIP : Telephone Routing over IP

ENUM

- converting E.164 phone number into an Uniform Resource Identifier (URI)
- Using Domain Name System (DNS) to store mapping
- a phone number is converted in a special domain name : e164.arpa
  i.e, 1-1800-555-1234 → 4.3.2.1.5.5.5.0.0.8.1.e164.arpa

TRIP

- finding an user in an integrated network.
- used by location server
- independent of signalling protocols.
- dividing the world into a set of IP telephone administration domain (ITAD's)

Soft switch.

Soft switch is simply centralized devices working with the help of a processing machine such as a computer.

It supports VoIP system, such as providing simultaneous and transparent support of the SIP and H.323 protocols.

Scanned with CamScanner

DSL (Digital Subscriber Line)

DSL is a technology for bringing high bandwidth information to homes and small business over ordinary copper telephone line i.e, subscriber line or loop.

XDSL refers to different variation of DSL such as ADSL (Assymmetric DSL)
HDSL (High bit rate DSL) →SHDSL (Single Pair H...
RADSL (Rate Adaptive DSL)
VDSL (Pre-standard Very high-bit-rate DSL)

Block Diagram

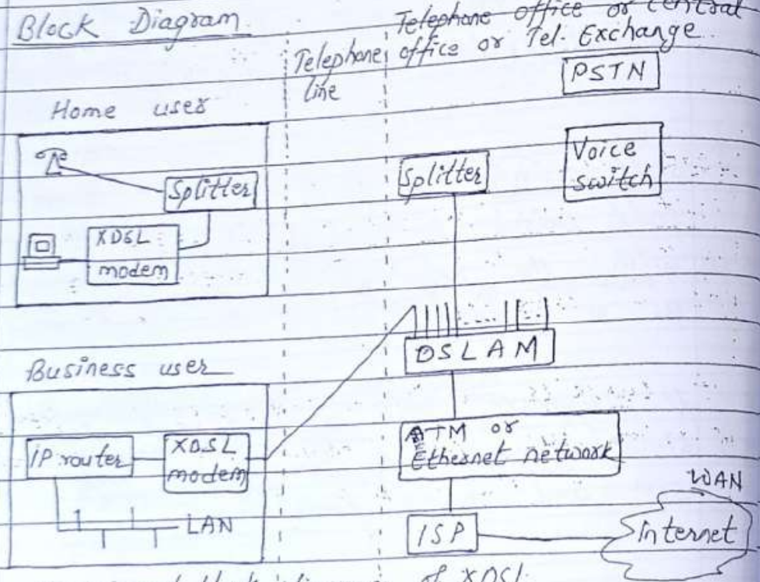
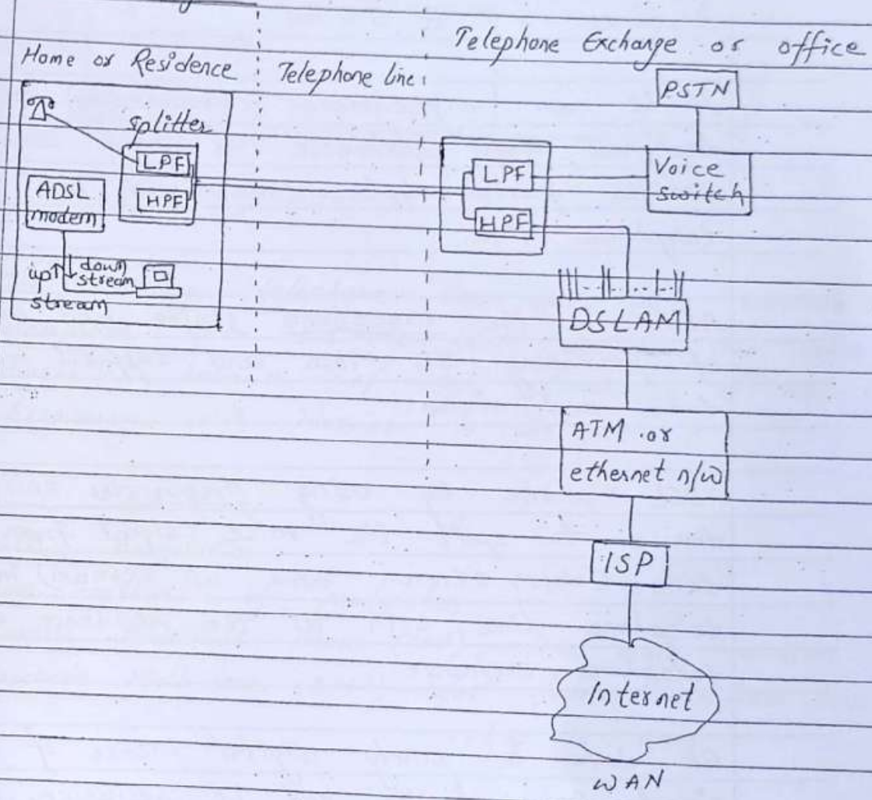


Fig - General block diagram of XDSL

XDSL can be explained with an example of ADSL.

ADSL
Block Diagram



LPF = Low Pass Filter
HPF = High Pass Filter
DSLAM = DSL Access Multiplier

ATM = Asynchronous Transfer Mode
ISP = Internet Service Provider
Down Stream bits → upto 8.1 Mbps
Up Stream bits → upto 800 Kbps
Distance → 0 to 5.4 km

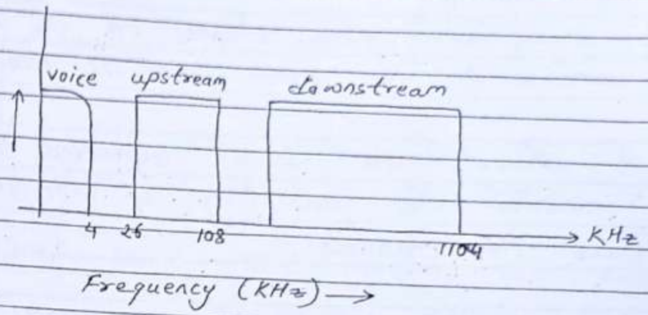
It is an asynchronous communication technology designed for residence or home user. It enables faster data transmission over copper telephone line.

ADSL is the broadband choice and capable of providing 50 Mbps and supports voice, data and video.

ADSL works by using frequencies splitter device to split the voice signal from ADSL data (down stream and up stream) in the telephone line both at the residence and telephone exchange.

All data is purely digital incase of ADSL signal. Only at the end i.e, residence and telephone exchange modulation to be carried over the line.

Bandwidth division in ADSL



The existing telephone line can handle bandwidths maximum upto 1.1 MHz. With distance this value decreases because of high attenuation of high frequency with increase in distance.

ADSL modulation
Two methods of modulation
1. Carrier Amplitude Phase (CAP) → line coding in QAM
2. Discrete Multi-tone (DMT) → fast Fourier Transform line coding in QAM.

CAP
It is an encoding method that drives the signals into two distinct bands
- the upstream data channel (to the service

provider) which is carried in the band between 25 to 160 KHz. the downstream data channel (to the user) which is carried in the band from 200 KHz to 1.1 MHz.

These channels are widely seperated in order to minimize the possibility of interference between the channels.

At start up, CAP tests the quality of the twisted pair and implements the most effective version of QAM (i.e, 16-QAM) for the channel.

It is a relatively single carrier system and has been a popular choice for some earlier implementations of ADSL modem. It is however replaced by the DMT method, which is superior in performance but more complex.

DMT

This method is multi-carrier (Carrier spacing of 4 KHz) modulation technique that seperates the available bandwidth into 256 sub channels or tones using discrete fourier transform method.

Because high frequency signals on twisted pairs suffer more in presence of noise, the transmission of data can be allocated so that lower frequency channels are utilized more.

Like CAP the system checks the line quality at start-up to determine the capacity of 256 channels.

DMT based ADSL modem can be thought of a 256 "mini modems" running simultaneously. Each channel is modulated using QAM and can carry between 0-15 bits/symbol/Hz.

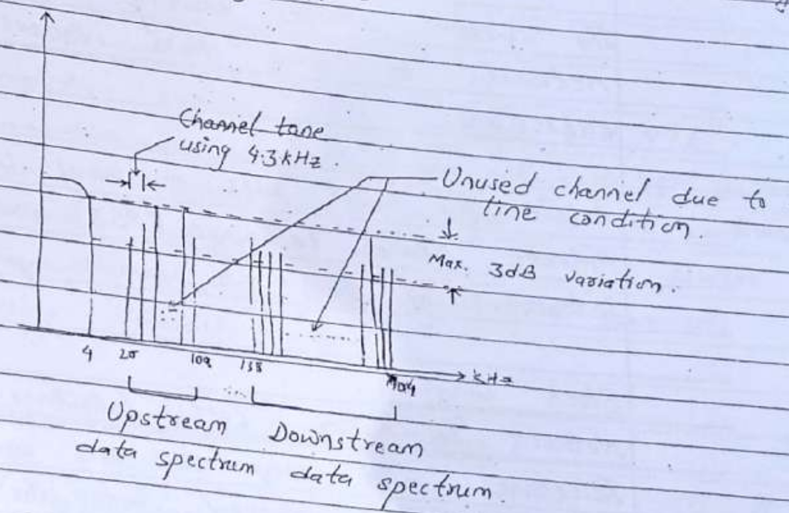


Fig:- DMT ADSL Spectrum

Upstream frequency i.e, carries → 32 ← FDM
Downstream frequency i.e, carries → 224 ← FDM

ISDN

The telecom service known as ISDN, incorporates the voice, data, video and fax information on the same carrier i.e, telephone line (twisted pair).

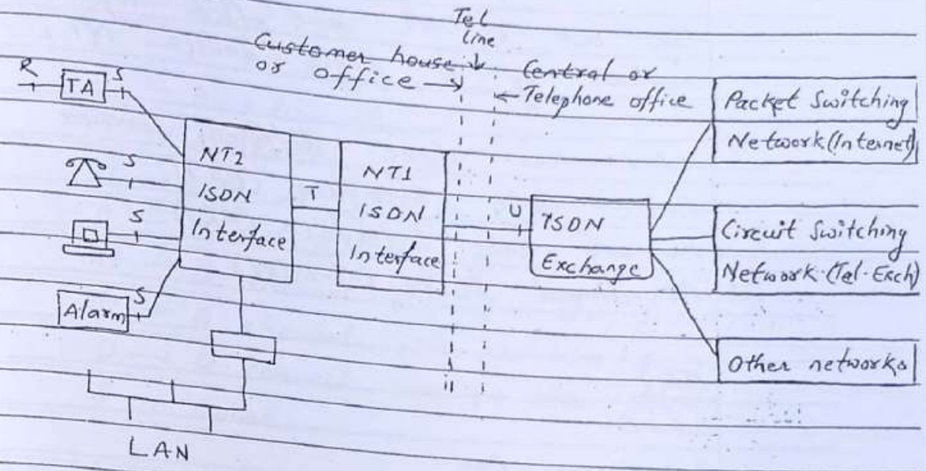
The system utilizes existing public telephone networks to bring these various services to subscribers.

Additionally, ISDN uses numerous LANs, WANs, MAN, Private Branch Exchange (PBX) and so on, interconnecting them via common public carriers.

Block Diagram:

Network Termination 1 (NT1) Interface

Telephones and computers are connected through NT1 which is plugged into the U-connector and provides the signal multiplexing onto the ISDN line. NT1 interface can connect upto 8 ISDN devices i.e, telephone or computer or both.



CCITT Standard

U → Connection between ISDN exchange and NT1

~~T → Connection between NT1 and/or NT2 and ISDN devices.~~

T → Connection between NT_1 provider to the customer

S → interface between NT_1 or NT_2 and ISDN devices.

R → Terminal Adapter and Non-ISDN terminals using different protocols.

Network Termination 2 (NT_2) Interface

For large business NT_2 is connected to NT_1. It connects to larger number of telephones or computers or both and other devices like PBX, LANs etc.

The use of NT_1 and NT_2 together can be replaced by a single device NT_{12}.

ISDN has three different services

1. Basic Rate Interface (BRI)
2. Primary Rate Interface (PRI)
3. Broadband ISDN (B-ISDN)

BRI

2B+D

B → Bearer line

B = 64 kb/sec

D = 16 kb/sec

144 kb/sec Information

160 kb/sec line speed

Frame structure

← 48 bits →

F	L	B1	L	D	L	F/A	L	B2	L	D	L	B1	L	D	L	B2	L	D	L

B1, B2 → 8 bits

Fig: Terminal to ISDN exchange

← 48 bits →

F	L	B1	E	D	A	F/A	N	B2	E	D	S	B1	L	D	L	B2	L	D	L

Fig: ISDN exchange to terminal.

F/A → Frame bit

L → DC balancing bit

E → D-echo-channel bit

A → Activation bit

N → set to opposite of F/A

D → Control signalling for ISDN devices.

B1 → B channel bits (16 bits/frame)

B2 → B channel bits (16 bits/frame)

D → D-channel bits (4 bits/frame)

S → Spare

PRI

23B + D

B = 64 kb/sec

D = 64 kb/sec

1.536 Mb/sec Information

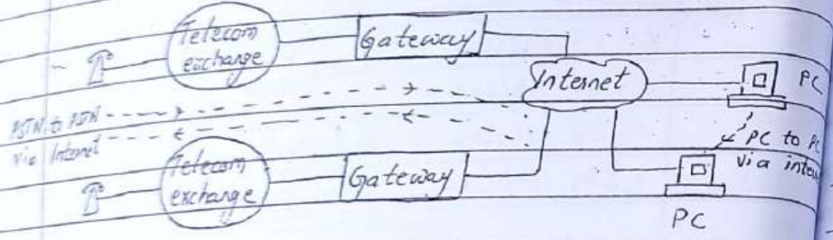
1.544 Mb/sec Line speed

Soft switch

A softswitch is a control device in a telecommunication network which connects telephone calls from one phone line to another, accross a telephone network or the public internet, entirely by means of a software running in a general purpose computer system.

The softswitch generally resides in a building owned by the telephone company called a telephone exchange. Most landline calls are routed by purpose built electronic hardware. However, softswitches uses servers and VoIP technology.

Many telecom network now make use of combination of softswitches and more traditional purpose-built hardware. Technically, soft switch refers to any IP technology, i.e IP to IP phone calls, IP to PSTN, PSTN to PSTN via IP Internet.



Transmission Media.

Defination

The transmission media or medium is the physical path between the transmitter and receiver in a data transmission system.

The characteristics and quality of data transmission are determined both by the nature of the signal and the nature of the medium.

Types

Two Types

1. Guided Media

In this case signal is guided along a physical path

- Twisted Pair Cable
- Coaxial Cable
- Wave guide
- Optical fibre

2. Unguided Media.

In this case the signal cannot be guided along physical path.

- air
- sea water
- Vaccum (space)

Comparision of Guided and unguided media.

In the case of guided media, the medium itself is more important in determining the limitation as shown below.

Transmission media	Total data rate	Bandwidth	Repeater spacing	Frequency Range
Twisted pair cable	4 Mb/sec	250 KHz	2-10 Km	upto 30MHz
Co-axial cable	500 Mb/sec	350 KHz	1-10 Km	30MHz-3G
Optical fibre	2Gb/sec	2GHz	10-100 Km	Light freq

waveguide → 1GHz-60G

For unguided media, the spectrum or frequency band of the signal produced by the transmitting (T) antenna is more important than the medium in determining transmission characteristics.

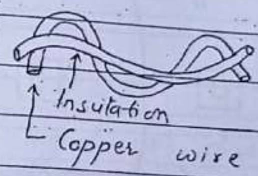
The higher the centre frequency of a signal, the greater the potential bandwidth and hence data rate. Another proper signals transmitted by antenna is directionality.

In general, at lower frequencies, signals are omni-directional. At higher frequency, it is possible to focus the signal into a directional beam.

Twisted Pair Cable

It consists of two insulated conductors which are twisted together to prevent physical seperatio but, more important, to give a well defined characteristics impedance.

Twisting the wire together increases the immunity to electromagnetic interference by increasing the coupling between the two wires, so that, interference will affect both more equally.



Twisting the wires together also reduces radiation and cross-talk between pairs in a cable.

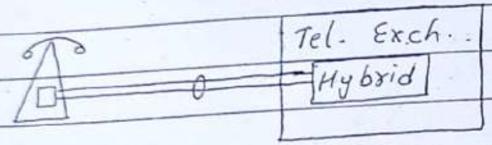
It is used to transmit baseband data. Commonly used to connect telephone subscribers to the local exchange because of low cost and well-defined characteristics.

Hybrid circuit or Transformer

The circuit that performs the 2-wire to 4-wire conversion or vice-versa is called Hybrid circuit.

Application

connection of subscriber and telephone exchange



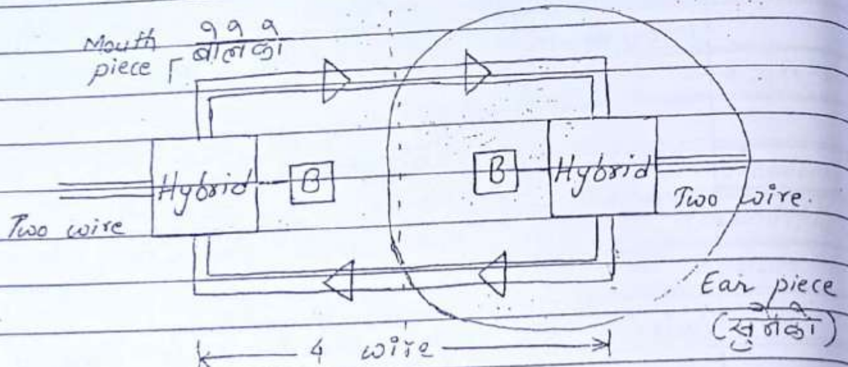
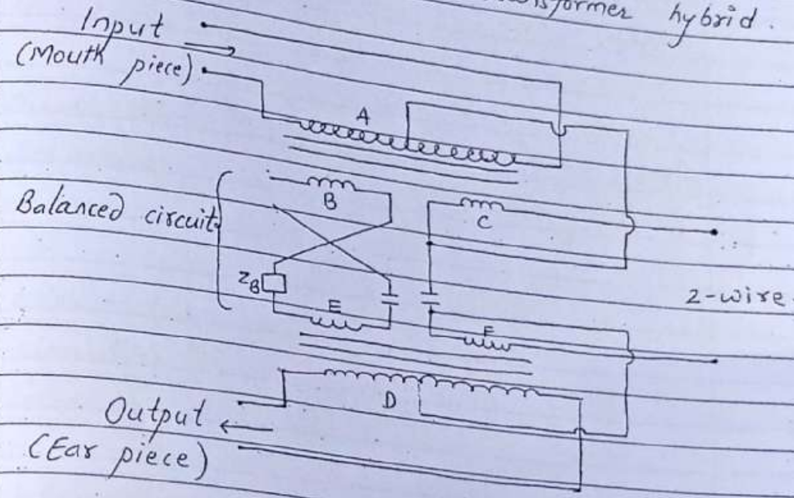


Fig- Simplified schematic of two-wire/four-wire operation

Actual

Two wire to four wire Transformer hybrid.



Operation

The input signal is coupled to B and F windings equally. Through C winding, the input is coupled to the 2-wire circuit. The same signal when it flows through the balanced 2-wire couples the signal to the winding D through winding C. The signal induced in B flows through E and induces current in D that opposes the current induced by F.

If the impedance Z_B exactly matches that of the 2-wire circuit, the effect of input signal on the output winding D is completely nullified.

In a similar way, the input signal from the subscriber end is completely nullified from coupling into the winding A.

Signal and noise measurement

Read yourself.

Signalling

It is a technical procedure to set up a path between two or more devices in a network in order to communicate information.

Three forms of signalling in a Telecom network

1. Subscriber loop signalling
- between subscriber and his local telephone exchange

2. Intra exchange signalling
- signalling among the various units within an telephone exchange

3. Interexchange signalling
- signalling between two or more exchanges.

Types of signalling

1. In-band signalling
- signalling message carried in some channel as user information in VF band

example → DTMF telephone set
→ TCP/IP

2. Out-band signalling
- seperate channels for signalling
example → SS7 signalling
→ ISDN access links

Types of out-band signalling
1. Channel Associated Signalling (CAS)
2. Common Channel Signalling (CCS)

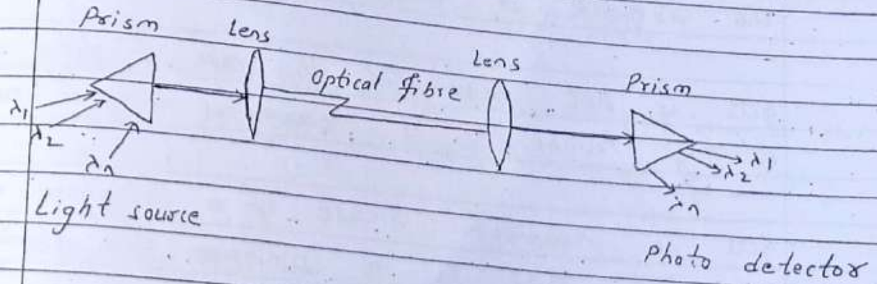
CAS
In this case, each and every speech channel associated with a signalling channel. This me[ans] for each speech channel a seperate signalling channel is required.
example → American TDM system
→ DTMF Telephone

CCS
In this case, a common signalling channel which takes care of all signalling information to be exchanged during communication. All of channels can be used for speech or data as required.
example → European TDM system
SS7 signalling system

Space Division Multiplexing
It is the combination of physically seperate channels being transmitted on seperate cables into a bundled cable.
example → number of subscriber lines in a cable
(overhead or underground cable).

Wavelength Division Multiplexing
WDM multiplexes multiple data streams with different wavelengths of light onto a single optical fibre.

Mechanism



Example of GOS and Blocking Probability (Chapter 6)

In a system with equal number of subscribers and servers, the GOS is zero, as there is always a server available to a subscriber.

On the otherhand, there is definite probability that all the servers are busy at a given instant and hence the blocking probability is non-zero.

The fundamental difference is that the GOS is a measure from the subscriber point of view whereas blocking probability is a measure from the network or switching system point of view.

GOS is not meaningful in case of delay system. Blocking Probability is meaningful in this case.

GOS is meaningful incase of a Telephone network in which there is no provision for storing information.

STS	TST
1. i/p space block is interfaced to o/p space block using a time block in between.	1. i/p time block is interfaced o/p time block using a space block in between.
2. Not preferred now since size of 'S' block being a matrix increases as square of i/p	2. Preferred now as size of time switch increases linearity with increase in i/p & o/p buses.
3. Now costly due to cost of switching hardware because two space switch blocks are required.	3. Now economical due to availability of low cost high speed memories required for T blocks.
4. For large size, size can be limited by splitting S blocks S-S-T-S-S.	4. Small size can be further reduced by T-S-S-S-T.
5. Peripheral functions cannot be incorporated into S-block	5. Peripheral functions such as MUX alignment of PCM with exchange frame can be incorporated into T block.
6. Simple design with large memory size	6. Complicated design with small memory size.
7. Only few applications.	7. Applications are more.

Scanned with CamScanner

Telephone Network :

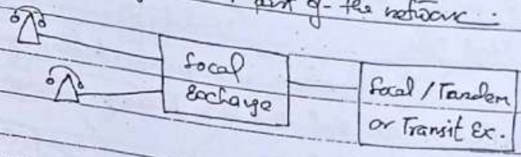
- In telephone network, a number of telephone exchanges are connected in such a way that a subscriber in any exchange can communicate with subscriber of any other exchange.

Types of Telephone Exchange :

- Exchange is the central part of the telephone which is used to connection several subscribers in a network.
- 3 major types of telephone exchange:

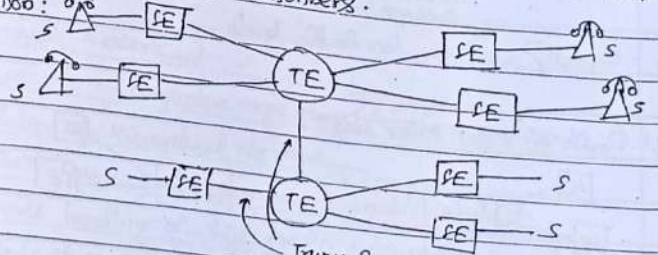
(1.) Local Exchange :

- It provides connection to its own subscribers.
- It also acts as a Gateway to other part of the network.
- Illustration :



(2.) Tandem Exchange :

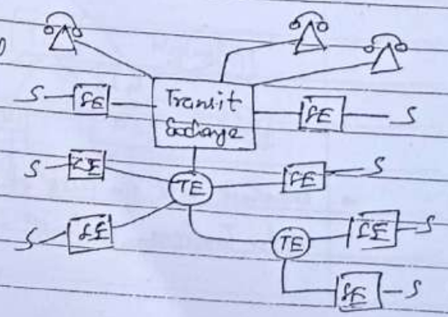
- An exchange used to connect several local exchanges within a multiple exchange area.
- It does not have its own subscribers.
- Illustration :



Trunk line or TG (Trunk Groups)
→ O/F, μWave, copper or satellite etc.

(3.) Transit Exchange :

- An exchange used to connect other local exchanges & tandem exchanges.
- It also has its own local subscribers.
- It may be connected with one or more other transit exchanges.



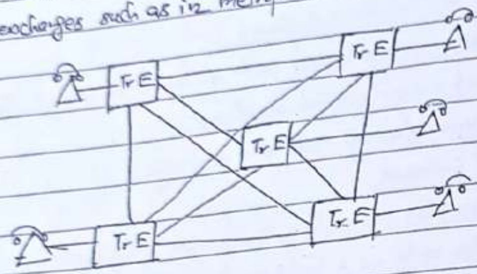
Bases of Network Configuration:

→ 3 major methods of connection of several exchanges in conventional exchange:

(1) Mesh Type:

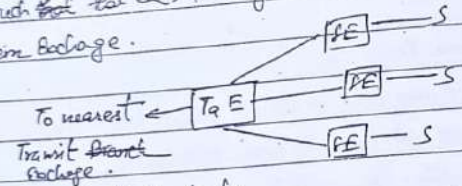
→ In this type, each & every exchange is connected by trunks.

→ Such connections are used when there are comparitively high traffic levels between exchanges such as in metropolitan networks.



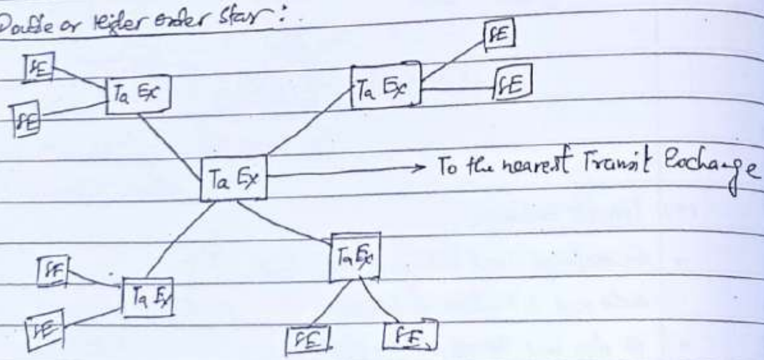
(2) Star Type:

→ A star type connection utilizes an interleaving exchange, called a Tandem Exchange, such that each & every exchange is interconnected via a single Tandem Exchange.



→ Generally used for low traffic levels.

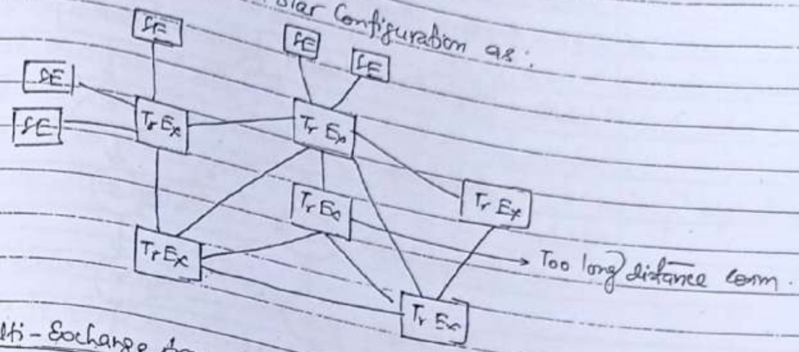
(3) Double or higher order star:



→ One where sets of pure star subnetworks are connected via higher order Tandem Exchange.

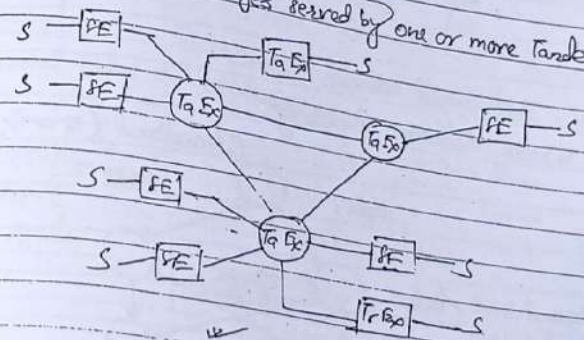
Actual Network

→ Combination of both Mesh & star Configuration as:



Multi-Exchange Area

→ A group of local exchanges served by one or more Tandem Exchanges.



Hierarchical Network:

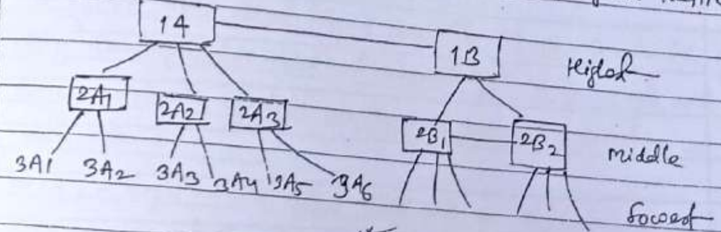
→ There are lots of confusion regarding Telephone network.

→ Evolution of Hierarchical Network

→ Reduces Trunk groups to some reasonable amount.

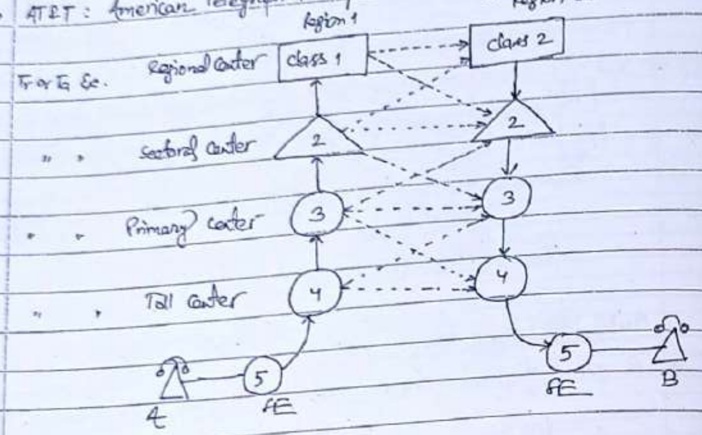
→ Permits handling of high traffic intensities in certain routes where necessary.

→ Three levels or ranks :→ Lowest, Middle & Highest Traffic.



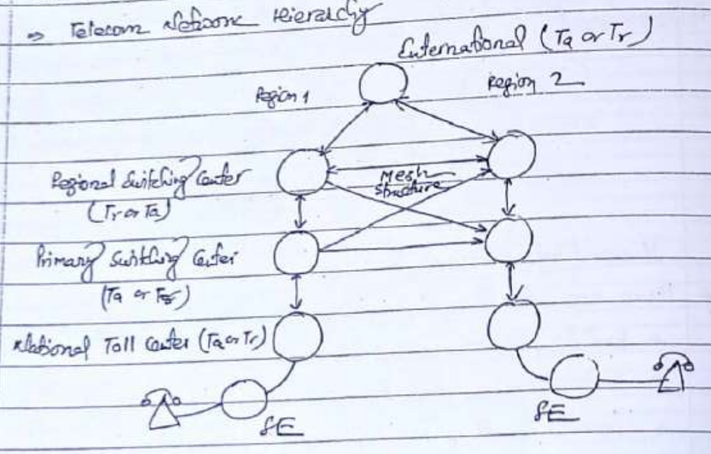
North American Hierarchical Network :

→ AT&T : American Telegraph & Telephone company



→ ITU : International Telecom Union

→ Telecom Network Hierarchy



#3: Signal Multiplexing (4 Hours)

Introduction:

→ Multiplexing is defined as any process of sending a no. of separate signals together, over same cable or bearer, simultaneously without interference.

→ More economical & efficient due to single channel occupation.

→ Signals (voice, video or data) are multiplexed together & resulting signal is transmitted over a system with a suitably high bandwidth.

→ At Rx, it is split up into the separate signals – Demultiplexing.

→ In telecommunication, multiplexing means the use of one telecommunication line to handle several channels of voice or data.

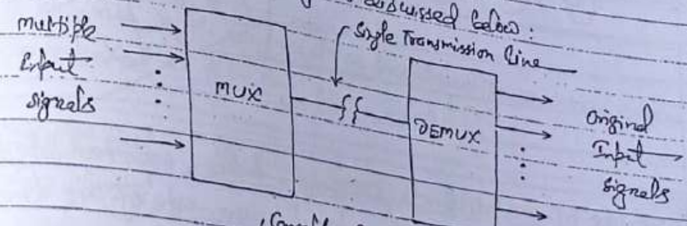
→ The best example is our TV cable.

→ The combination of multiplexer & demultiplexer at terminal station – MUX.

→ Primary use – To save communication line costs.

→ Several types of multiplexing as discussed below:

(★)



(Concept of Multiplexing)

Wavelength Division Multiplexing (WDM):

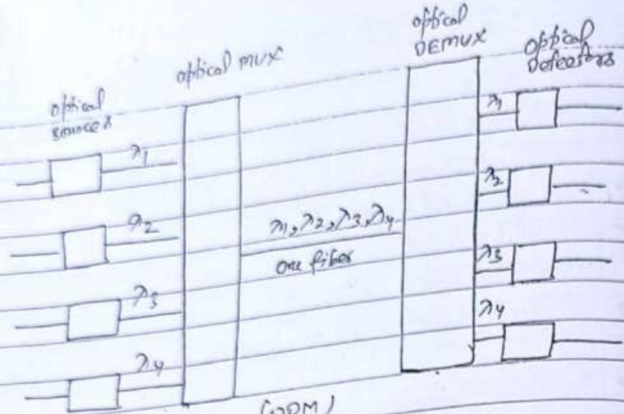
→ A cost-effective way to increase the capacity of fiber optic communication.

→ The key elements of a WDM optical systems – Tunable semiconductor lasers, electro-optical modulators, multiplexing components, single-mode optical fiber & optical amplifiers.

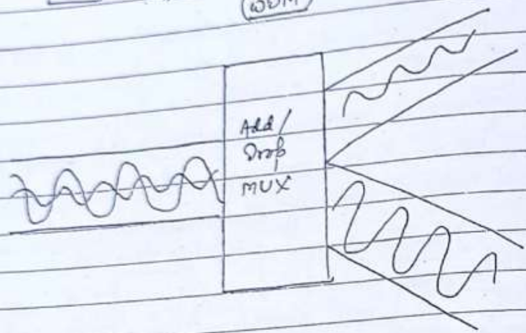
→ This system uses optical fiber's available intrinsic bandwidth by multiplexing many wavelengths (or colors) of coherent light along a single mode optical fiber.

→ Each wavelength of light can transmit encoded information at the optimum data rate.

→ For eg, a single-mode optical fiber with an attenuation of 0.2dB/km at 1550 nm is capable of accomodating a set of wavelengths each spaced apart by a few tenths of nm (50 GHz – 100 GHz).



(WDM)



→ The use of 48 distinct wavelength lasers, each modulated at 2.5 Gbps, represents an effective transmission rate of 48 times 2.5 Gbps which is equal to 120 Gbps.

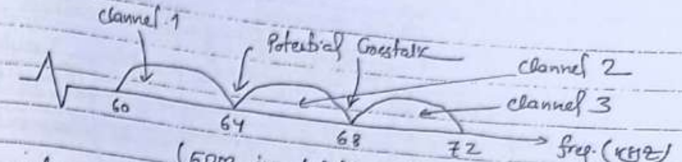
→ also called DWDM (High-Density or Dense WDM).

Frequency Division Multiplexing (FDM):

→ A broadband analog transmission technique in which multiple signals are transmitted over a single cable simultaneously as shown below:

30 KHz	30.3 - 33.1 KHz	30 KHz Carrier	Bandwidth transmitted
	34.3 - 37.1 KHz	34 KHz Carrier	
40 KHz	38.3 - 41.1 KHz	40 KHz Carrier	

(Telephone Multiplexing)



(FDM in telephone transmission)

→ FDM system divides the available BW of the transmission medium into a no. of narrow band or sub channels.

→ The channels are sent over a common path by modulation each channel to a different carrier frequency.

→ The signal thus occupies a relatively narrow BW which is a part of a much wider BW transmitted.

⇒ Each speech channel occupies 4KHz of available B/W as shown above.

→ These modulated carriers are all amplified & transmitted together over the channel.

→ Extensively used in: Point-to-point microwave radio, coaxial cable etc.

→ FDM Hierarchy:

→ To standardize the equipment in broadband transmission systems, CCITT recommended the following FDM hierarchy:

Multiplex level	No. of Voice Channels	Formation	Frequency Band (KHz)
Voice channel	1	—	0-4
Group	12	12 voice channels	60-108
Super Group	60	5 groups = 60 voice channels	312-552
Master Group	600	10 super group	564-3084
Jumbo Group	3600	6 Master "	564-17548
Jumbo Group Mix	10800	3 Jumbo "	3000-60000

→ All multiplex equipment in the FDM hierarchy uses SSB modulation.

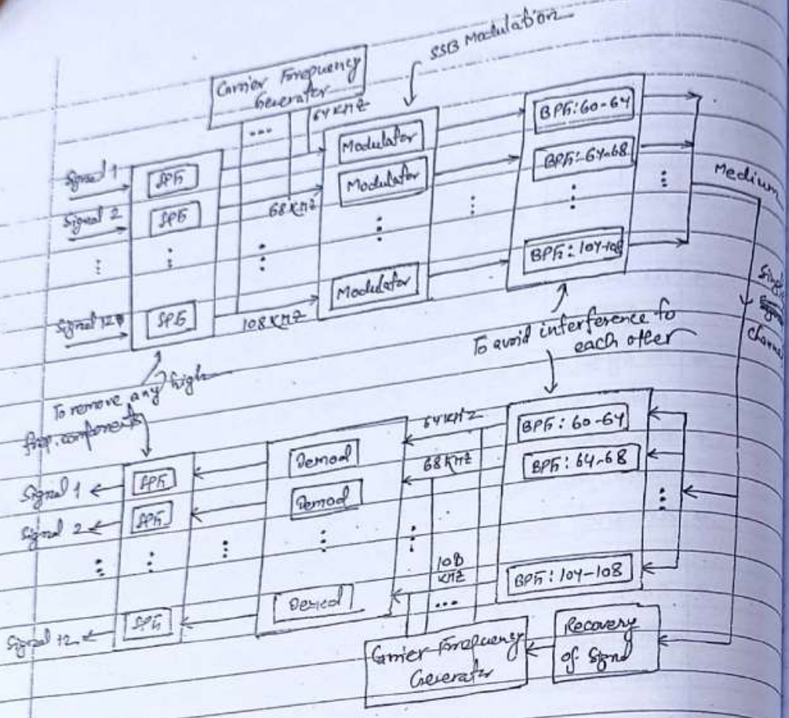
→ The multiplex equipment is independent of particular transmission media.

→ FDM Principle:

→ Consider FDM for 12 telephone channels (group multiplex level).

→ See figure on next page!

→ Explain in brief. !!!

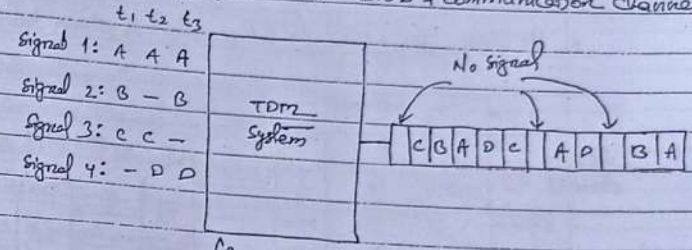


Space Division Multiplexing (SDM):

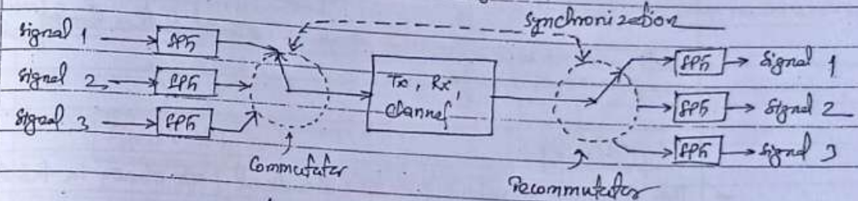
- More than one physical transmission path are grouped together.
- A telephone cable consisting of 100s (or 1000s) of twisted pair constitutes a space division multiplex system, wire pair cables are constructed containing many 100s of wire pairs.
- Several coaxial tubes bound together in one cable is also an example of SDM.
- A very large no. of separate telephone calls can be transmitted together down a coaxial system.
- Single wire pair commonly carries 12 or 24 voice channels, but one single coaxial tube commonly carries 3600 or higher capacity as 10800.
- For trunk cable lines, 70-300 lb copper wire are used.
- With coaxial cable 10800 two way telephone conversion are possible.
- SDM is not limited to voice freq. circuits alone.
- Systems using either FDM or TDM can also be space division multiplexed.

Time Division Multiplexing (TDM):

- Introduced in 1970s & now favored over FDM.
- It is the sharing of a common channel transmission medium in time.
- Time available is divided into small slots, and each of them occupied by a piece of one ~~time~~ of the signals to be sent.
- A base band technology in which individual channels of data or voice are interleaved into a stream of framed bits across a communication channel.



(Principle ~~is~~ of TDM)

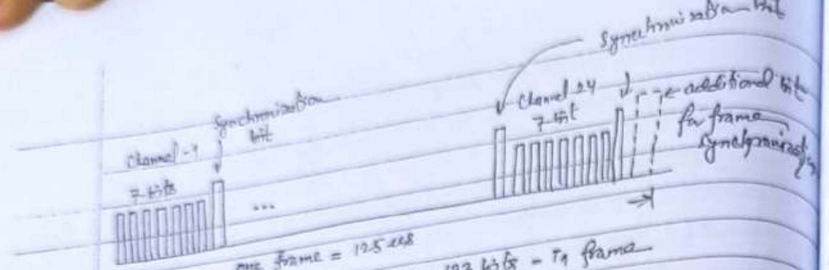


(TDM System)

- For analog signals, the signals should be sampled at first.
- Analog signals are digitized by using CODEC (Coder/Decoder) device ~~to capture~~
- Produces 7-8 bits, f_s = 8000 samples/sec (8 KHz), T_s = 125 us.
- Generally, PCM is used in TDM system with 4KHz telephone channel BW.

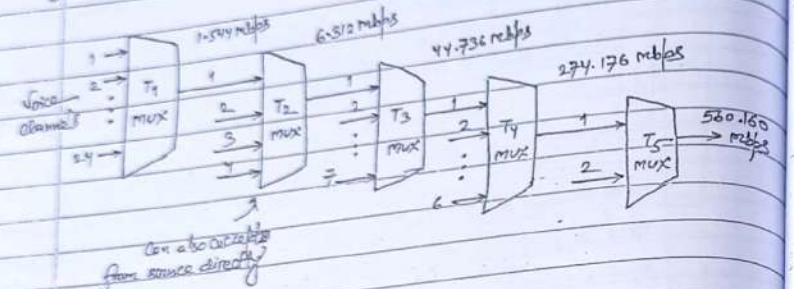
T1 Carrier System:

- also called North American Hierarchy or TDM-PCM hierarchy.
- Widely used in North America & Japan
- T1 consists of 24 voice channels which are sampled ~~separat~~ at f_s = 8 KHz (T_s = 125 us) and then quantized & converted into 7-bit PCM word with additional 8th bit for synchronization as shown on the next page:



one frame = 125 μs
= 24×8+1 = 193 bits – T_1 frame
= 1.544 mbps signalling rate.

→ The whole TDM-PCM (T_1) hierarchy is shown below:



E1 Hierarchy:

→ The European standard (CCITT Standard) as shown in tabular form:

S.N.	Level	No. of Inputs	Output Rate
		30	2.048 mbps
1	First level (E_1)	4	8.448 "
2	2nd " (E_2)	4	34.368 "
3	3rd " (E_3)	4	139.264 "
4	4th " (E_4)		

→ A PAM or PPM can be employed for TDM — now obsolete as the transmitted pulses dispersed due to attenuation & delay distortion.

→ They produce interchannel cross talk due to spreading.

#4: Digital Switching (8 Hrs)

Electronic Space Division Switching:

Stored Program Control (SPC):

→ Modern digital computers uses the stored program concept.

→ A program or a set of instructions to the computer is stored in its memory & the instructions are executed automatically one by one by the processor.

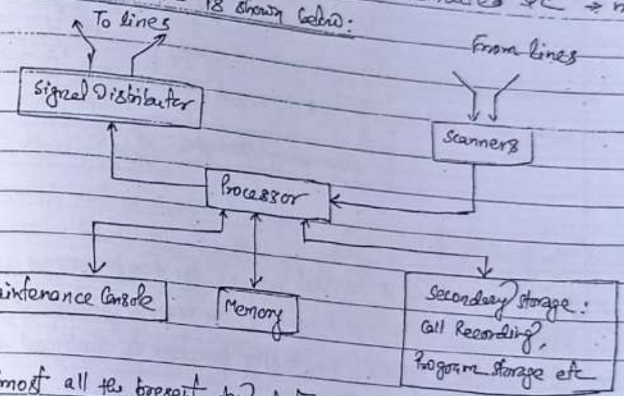
→ Carrying out the exchange control functions through programs stored in the memory of a computer — called SPC.

→ Application of SPC in telephone switching includes:

(1.) Automation of exchange functions
(2.) Common Channel Signalling (CCS)
(3.) Centralized maintenance & automatic fault diagnostics
(4.) Interactive human-machine interface, etc.

→ Two ways of organizing SPC → (1.) Centralized SPC → earlier
→ (2.) Distributed SPC → now-a-days

→ Typical centralized SPC is shown below:



→ In almost all the present day electronic switching systems using centralized SPC, only a two-processor configuration is used.

→ A dual-processor architecture may be configured to operate in one of three modes:

(1.) Standby Mode:

→ Simplest dual processor configuration operations.

→ Normally one processor is active & the other is on standby, both the hardware & software wise.

→ The standby processor is brought online only when the active processor fails.

⇒ Requirement – ability of the standby processor to reconstitute the state of the exchange system when it takes over the control.

⇒ Need to determine which of the subscribers & trunks are busy or free, which of the paths are connected through the switching network, etc.

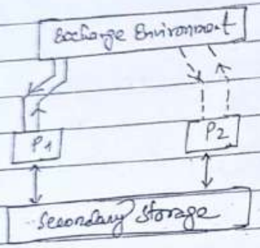
⇒ In small exchange, it is possible to by scanning all the status signals as soon as the standby processor is brought into operation.

⇒ In large exchange, it is not possible to scan all line within reasonable time.

⇒ Here, active processor copies the status of the system periodically (say 5 sec) into a secondary storage.

⇒ When a switch over occurs, the online processor loads the most recent update of the system status from the storage & continues the operation.

⇒ Only the calls which changed status between last update & the failure of the active processor are disturbed.



P1 - active Processor

P2 - standby "

(2.) Synchronous Duplex Mode:

⇒ Hardware coupling is provided between the two processors which execute the same set of instructions & compare the results continuously.

⇒ If a mismatch occurs, the faulty processor is identified & taken out of service within few milliseconds.

⇒ When operating normally, both processors have same data in their memories.

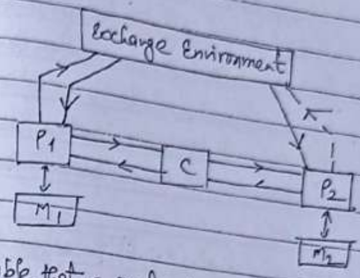
⇒ One of the processor is actually controls the exchange, whereas the other is synchronized with the former but does not participate in exchange control.

⇒ If a fault is detected by the comparator, P1 & P2 are decoupled & a check-out program is run independently on each of processor to determine which one is faulty without disturbing call processing.

⇒ When faulty processor is taken out of service, the other processor operates

independently.

⇒ When faulty processor is repaired & brought into service, the memory contents of the active processor is copied into its memory, brought into synchronous operation with the active processor.



C: Comparator

Mi = Memory

Pi = Processor

⇒ It is possible that a comparator fault occurs on account of a transient failure which does not show up when the check-out program is run.

⇒ In such cases, the decision as to how to continue operation is arbitrary & three possibilities exist:

(i) Continue with both the processors.

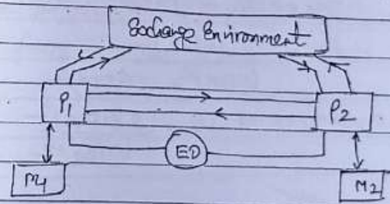
(ii) Take out the active processor & continue with the other processor

(iii) Continue with the active processor but remove the other processor from service.

(3.) Load Sharing Mode:

⇒ An incoming call is assigned randomly or in a predetermined order to one of the processors which then handles the call right through completion.

⇒ Thus, both the processors are active simultaneously & share the load and the resources dynamically.



ED: Exclusion Device

⇒ There is an interprocessor link through which the processor exchange information needed for mutual coordination & verifying 'State of Health' of the other.

⇒ If the information exchange fails, one of the processors which detects the same takes over the entire load including the calls that are already set up by the failing processor.

⇒ But the calls that were being set up / established by the failure processor are usually lost.

⇒ ED mechanism is used so that both the processors do not seek the same resource at the same time — can be implemented in hardware or software.

⇒ Under normal operation, each processor handles 1/2 calls on a statistical basis.

⇒ Has much better performance in the presence of traffic overloads.

Availability Figures of a Single / Dual Processor System :

⇒ Single Processor :

Availability, $A = \frac{MTBF}{MTBF + MTTR}$

where MTBF = Mean Time Between Failure

MTTR = " " To Repair

Unavailability, $U = 1 - A = \frac{MTTR}{MTBF + MTTR}$

If $MTBF >> MTTR \Rightarrow U = MTTR / MTBF$

⇒ Dual Processor :

$$MTBF_D = \frac{(MTBF)^2}{2MTTR}$$

$$\therefore A_D = \frac{MTBF_D}{MTBF_D + MTTR} = \frac{(MTBF)^2}{(MTBF)^2 + 2(MTTR)^2}$$

$$\therefore U_D = 1 - A_D = \frac{2(MTTR)^2}{(MTBF)^2 + 2(MTTR)^2}$$

⇒ If $MTBF >> MTTR \Rightarrow U_D = \frac{2(MTTR)^2}{(MTBF)^2}$

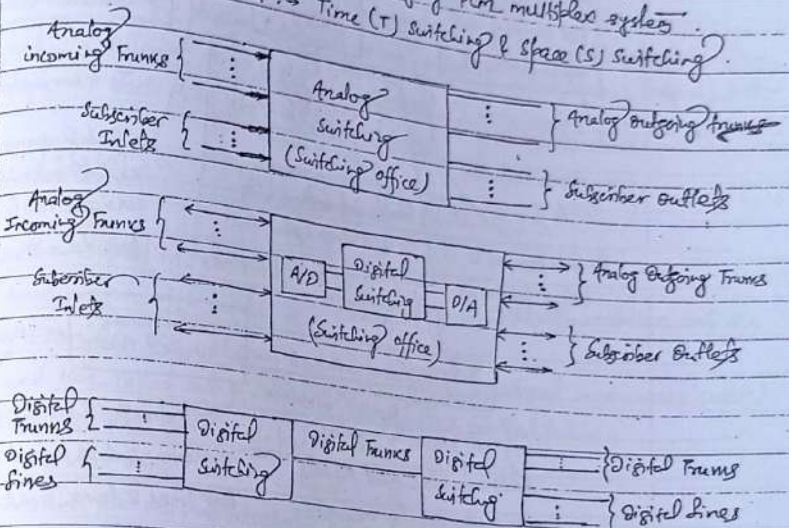
Digital Telephone Exchange :

⇒ Various facilities of digital switching & transmission is the reason why the analog switching is slowly getting replaced by digital switching.

⇒ A switching system is called digital when the input to and output from the switching system can be directly support digital signal.

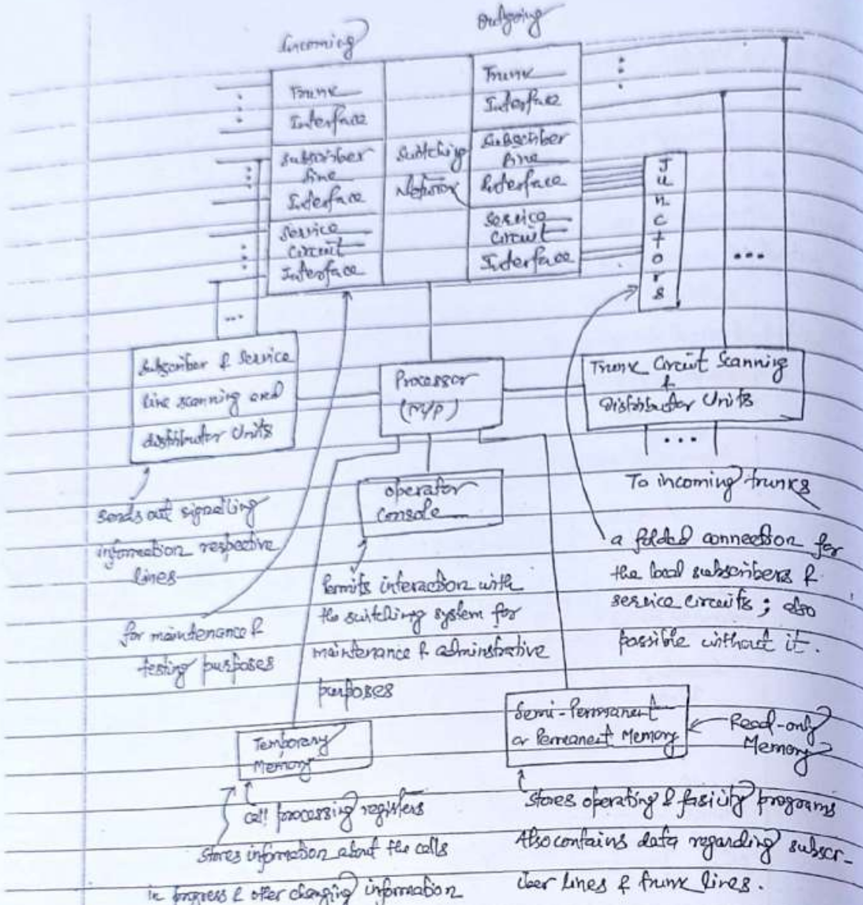
⇒ Function is to connect pairs of channels so that information arriving at the switching center in a particular channel on one PCM multiplex system can be passed to some other channel on an outgoing PCM multiplex system.

⇒ Two processes are involved :→ Time (T) switching & Space (S) switching.



(Evolution of Digital Switching)

⇒ The basic configuration of a Digital Telephone Exchange its its basic elements is shown on the next page:

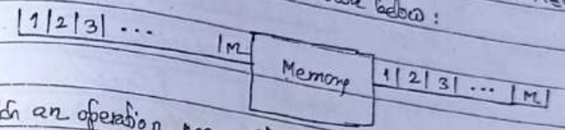


Approach to PCM switching:

- A digital switch architecture is made of two elements, called T (Time) and S (Space) or time division switch & space division switch, respectively.
- It may has one or more T & S switches, forming 2nd order or higher order switching architecture.

Time (T) switch:

- This is also called TSI (Time slot Interchange) switch.
- This switch involves moving the data contained in each TS (Time slot) from the incoming bit stream (frame) to an outgoing bit stream (frame) but with a different time-slot arrangement in accordance with the destination of each time slot as shown below:

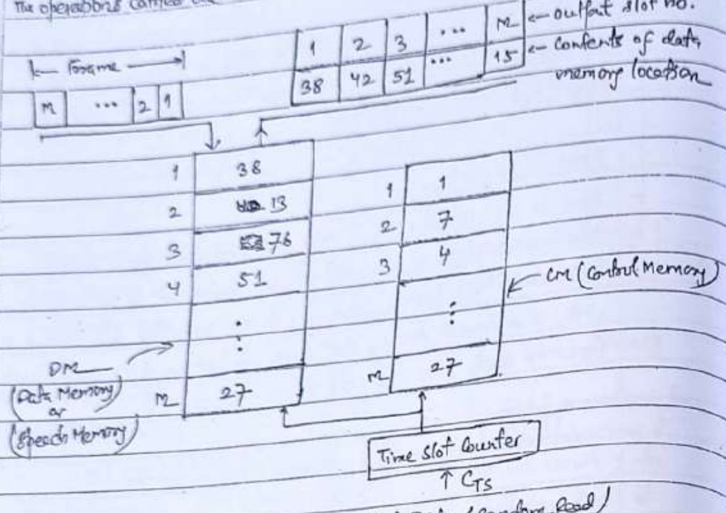


- Such an operation necessarily implies a delay between the reception and the transmission of a sample.
- To accomplish this, at least one time-slot must be stored in memory (write) and then called out of memory (Read) in a changed position.
- There are two choices of handling this approach:

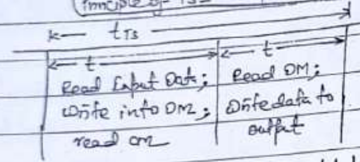
(1.) Sequential Write / Random Read:

- Consider the illustration shown on next page with one incoming & one outgoing trunk containing M channels multiplexed on each trunk.
- Thus, time-slot duration: $t_{TS} = 125/M$ us.
- The time slot clock runs at the time slot rate, i.e., at the rate of 1 pulse every 125/M us.
- The time slot counter is incremented by one at the end of each time slot.
- The contents of the counter provides location addresses for the data memory and the control memory.
- Data memory & control memory accesses take place simultaneously in the beginning of time slot.
- There after, the contents of control memory are used as the address of the data memory & data read out to the output trunk.

- The operations carried out in one time slot are shown below in next figure :

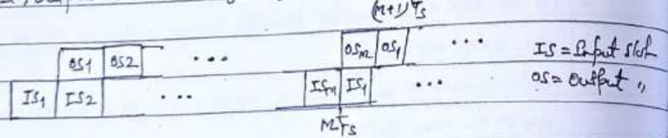


(Principle of TSI - Sequential Write / Random Read)



(operation in a time slot)

- The input sample is available for reading in at the beginning of the time slot & sample is ready to be clocked on output stream at end of time slot.
- Even if there is no time slot interchange, a sample is delayed by a min. of one time slot in passing from input to output stream due to storage action.
- In effect, output stream is delayed by t_{TS} us as shown below :



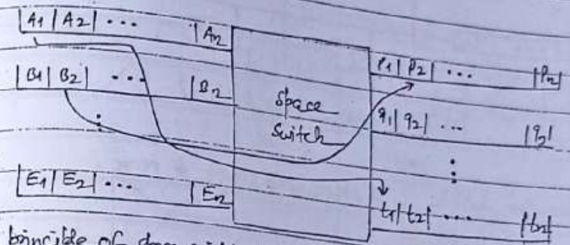
- Depending on the output time slot to which an input slot contents are switched, the sample experiences a delay in the range of t_{TS} to $M t_{TS}$.

- In the example entries, the 1st location in CM contains value 1. The sample in this case, experiences a delay of t_{TS} us.
- The 2nd location in CM contains the value 7 & therefore, input time slot 7 is switched to output time slot 2 with $(M-4) t_{TS}$ delay.
- Output time slot 3 carries the contents of input time slot 4 and delay $= M t_{TS}$ us.

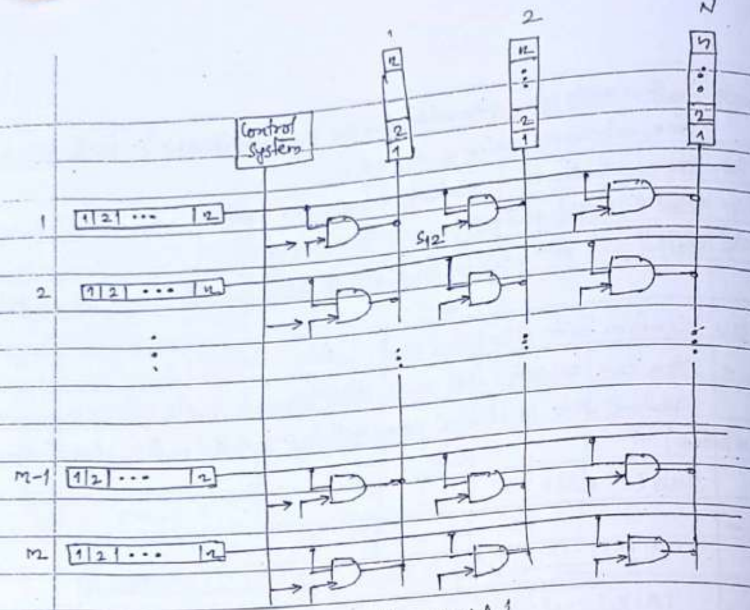
(2) Random Write / Sequential Read :

- You can explain with same block diagram ; with one difference that random write in place of sequential and sequential read in place of random.

Space (S) switch :



- The principle of space switch is to transmit information of a time slot in one inlet to same time slot of other outlet as shown below.
- The actual configuration of S switch is shown on the next page :
- There are M horizontal & N vertical bars ; depending on their relative values, we get 3 different senarios :
  (i) $M > N$ ⇒ The switch is of blocking nature.
  (ii) $M < N$ ⇒ The switch is of expandable nature.
  (iii) $M = N$ ⇒ The switch is of non-blocking nature.
- If is desired to transmit a single from input 1 (horizontal) to output 2 (vertical), the gate at the intersection would be activated by placing an enable signal on S_{12} during the desired time slot period.
- Then bits of that time slot on inlet would pass through the logic gate onto the same time slot on outlet.



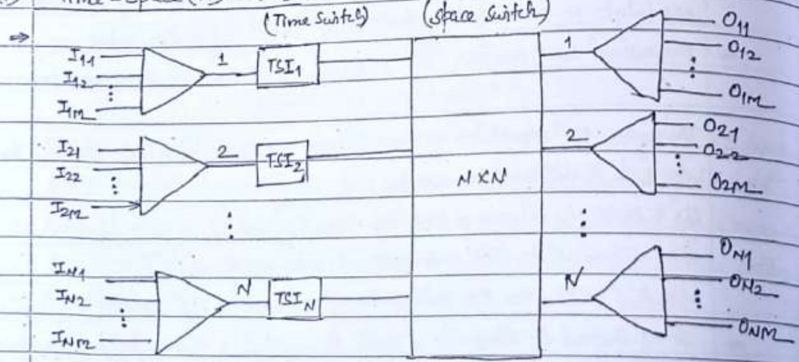
(Time - Division Space Switch)

Combination Switch : 2 stage, 3 stage & more.

Two - stage Switch:

(1) Time - Space (TS) Switch :

⇒



N = No. of TDM lines & M = No. of time slots.

⇒ A subscriber on the input side is assigned to one of inlets and a time slot in that inlet as I_{47} = 7th time slot of inlet 4.

⇒ A subscriber connected to the time slot 6 of outlet 5 = O_{56}

⇒ Consider the connection between I_{47} and O_{56}:

⇒ $I_{47} \xrightarrow[T]{TSI} I_{46} \xrightarrow{S} O_{56}$.

(4.10)

⇒ While this configuration ensures full availability, it is not nonblocking.

⇒ Consider two connections: I_{47} & O_{29} and I_{43} & O_{69}:

⇒ $I_{47} \xrightarrow{T} I_{49} \xrightarrow{S} O_{29}$

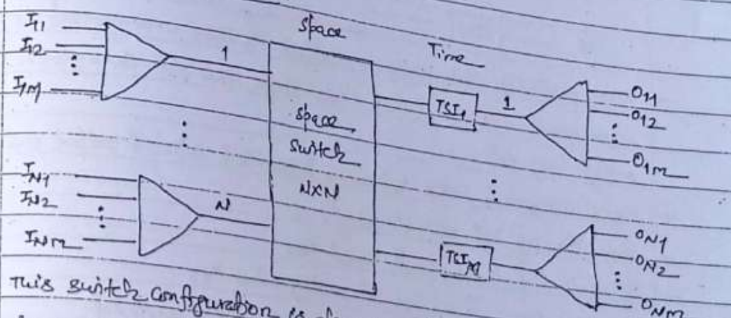
$I_{43} \xrightarrow{T} I_{49}$

↳ not feasible as 9th time slot of 4th inlet has been used.

⇒ Drawback - Blocking occurs if two input I_{ij} & I_{ik} are destined to outputs O_{pq} & O_{rq}, respectively.

⇒ Theoretically, the TS switch can be made nonblocking by using an expanding T switch & a concentrating S switch.

(2.) Space - Time (ST) Switch :



⇒ This switch configuration is also not nonblocking.

⇒ Eg: → (1.) I_{27} and O_{19}

⇒ $I_{27} \xrightarrow{S} I_{17} \xrightarrow{T} O_{19}$

⇒ (2.) I_{75} & O_{96} and I_{85} & O_{92}.

⇒ $I_{75} \xrightarrow{S} I_{95} \xrightarrow{T} I_{96}$

$I_{85} \xrightarrow{S} I_{95}$

↳ not feasible as already been used!

⇒ Drawback :- Blocking happens when the input samples originate from two different inlets during the same time slot and are destined to the same outlet through to different time slots.

Sol

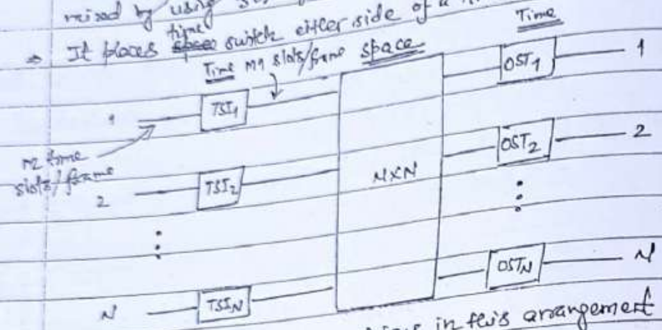
(4.11)

Three - stage switch :

(1) Time - Space - Time (TST) switch :

⇒ Problem associated with 2 stage networks (for blocking) can be minimized by using 3 or higher stage networks.

⇒ It places ~~space~~ time switch either side of a time switch as shown below:



⇒ First flexibility that becomes obvious in this arrangement is that there is no need to have a fixed space stage time slot for a given input or output time slot.

⇒ Many alternative paths between a prescribed input & output unlike earlier.

⇒ Less blocking but still blocking.

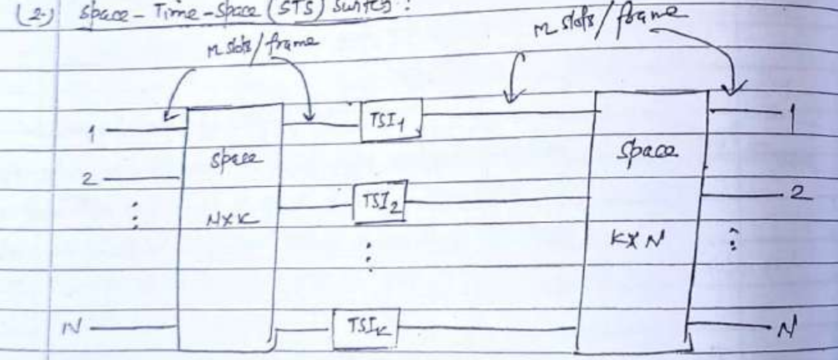
⇒ Eg:→ (i) I_{47} & O_{29} and I_{43} & O_{69}

⇒ $I_{47} \xrightarrow{T} I_{49} \xrightarrow{S} I_{19} \xrightarrow{T} O_{19}$

$I_{43} \xrightarrow{T} I_{49}$ – not as already been used.

$\xrightarrow{T} I_{48} \xrightarrow{S} I_{68} \xrightarrow{T} I_{69}$

(2) Space - Time - Space (STS) switch :



⇒ operation: ① I_{27} & O_{19} and I_{23} & O_{49}

⇒ $I_{27} \xrightarrow{S} I_{17} \xrightarrow{T} I_{19} \xrightarrow{S} O_{29}$

$I_{23} \xrightarrow{S} I_{43} \xrightarrow{T} I_{49} \xrightarrow{S} O_{49}$

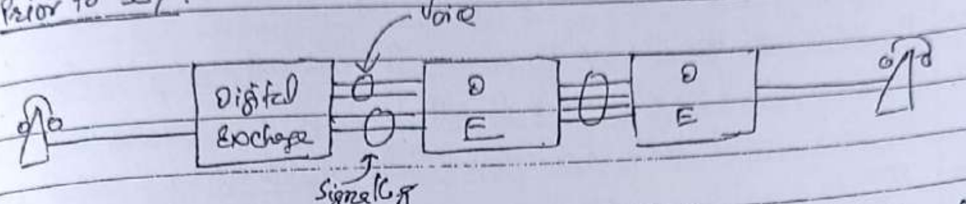
⇒ As earlier, better nonblocking but still blocking.

Chapter-5 [See class notes also for this chapter.]

Signalling System 7 and Its Network:

⇒ Common channel signalling technique, using a separate network of computer to send & receive message as signals, has been adopted by ITU as CCIS#7 and is referred to as SS7.

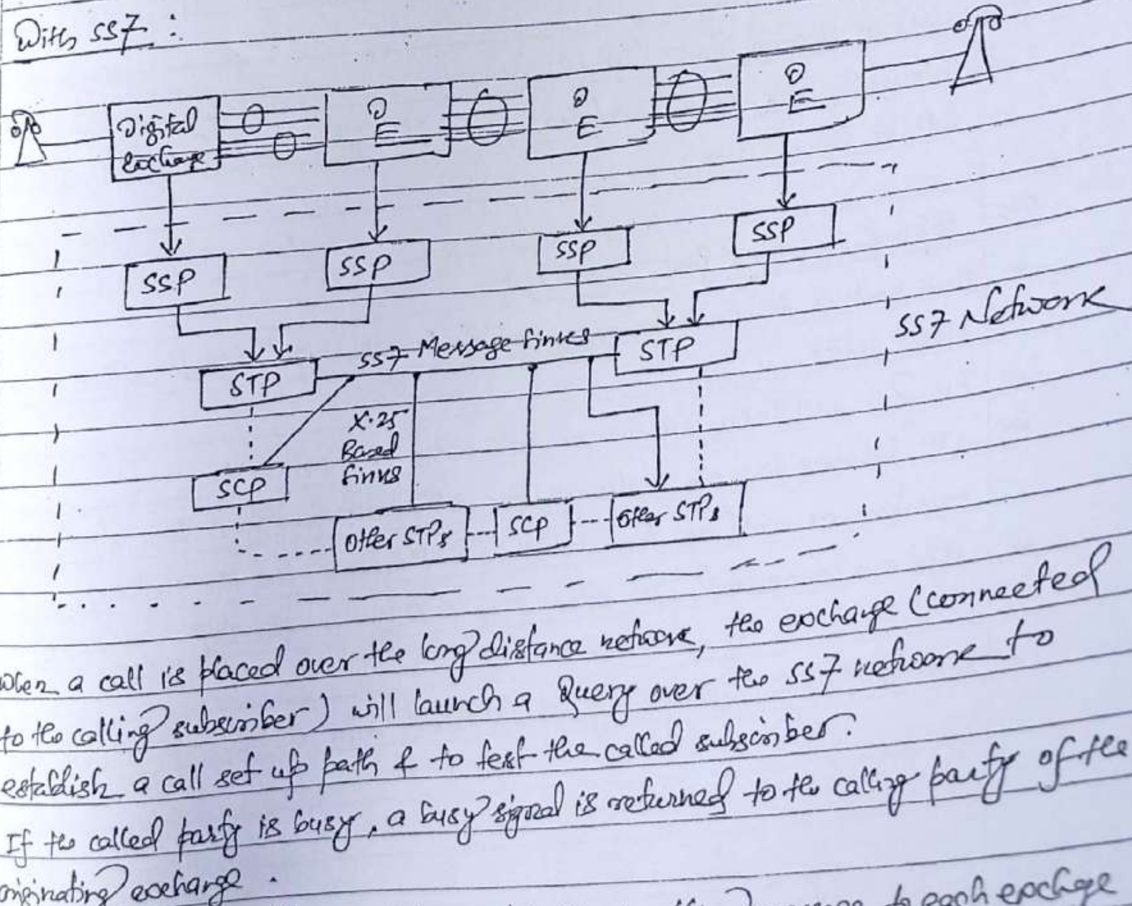
⇒ Prior to SS7:



⇒ When a long distance call was made, the call was connected Exchange by Exchange through the long distance network.

⇒ On calls to a busy telephone, the call would be established over the network and tie up numerous circuits in the PSTN just to return a busy signal from the far end exchange.

⇒ With SS7:



⇒ When a call is placed over the long distance network, the exchange (connected to the calling subscriber) will launch a Query over the SS7 network to establish a call set up path & to test the called subscriber.

⇒ If the called party is busy, a busy signal is returned to the calling party of the originating exchange.

⇒ If the called line is idle, the SS7 now issues signalling message to each exchange that will be used to connect the call.

⇒ Each exchange will communicate over SS7 network and identify the trunk circuits to be used between each exchange for the voice circuit and these trunk circuits are reserved for the call.

⇒ A signal is sent to the terminating EO, instructing it to ring the called line.

⇒ The caller receives the ring back tone from its originating exchange.

⇒ When the called party answers the call, the reserved trunks are then connected & voice circuit path is completed between the calling & called parties.

⇒ SS7 is a network of its own.

⇒ This network consists of 3 major parts : STP, SCP & SSP.

(1.) Signal Transfer Point (STP):

⇒ It is a packet based switch.

⇒ It receives & sends message signal to proper destination.

⇒ It also performs specialized routine functions.

(2.) Service Control Point (SCP):

⇒ This node acts as a Database which provides the information for the advanced call in the network.

(3.) Service Signalling Point (SSP):

⇒ This node is the end exchange equipped with SS7 enabled software and having signal links.

⇒ Every EO will have SSP to connect with SS7 network.

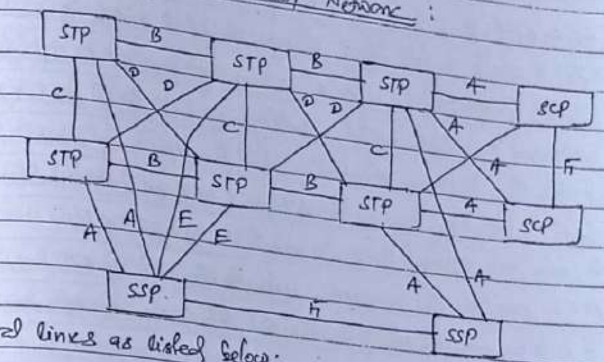
⇒ SSP behaves like an exchange for different functions like starting the transmission as well as termination of the link in the network.

⇒ SSPs are connected with other STPs & SCPs in the SS7 network.

⇒ SS7 is characterised by high speed packet Data & out-of-Band signalling.

Different Signalling links in SS7 Network :

⇒



⇒ Several links as listed below:

(1.) A link ⇒ Access link - used to connect SSP or SCP to STP.

(2.) B link ⇒ Bridge " - " " " two STPs.

(3.) C link ⇒ Cross " - " " " paired STPs for reliability.

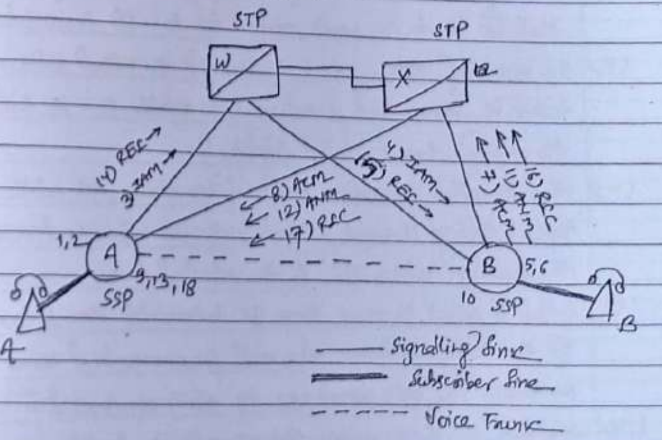
(4.) D link ⇒ Diagonal " - " " " STPs diagonally & similar to B link.

(5.) E link ⇒ Extended " - " " " SSP with the remote STP which can not be by A link because of too far distance.

(6) F link ⇒ Fully Associated " - used to connect two SSPs or two SCPs.

Basic Call set-up in SS7 Signalling System :

⇒



⇒ Subscriber on switch A places a call to subscriber B on switch B.

⇒ operation — 18 steps :

(1.) Switch A analyzes the dialed digits & determines that it needs to send ~~sort~~ the call to switch B.

(2.) Switch A selects an idle trunk between itself & switch B and formulates IAM (Initial Address Message), the basic message needed to initiate a call. The IAM is addressed to switch B. It identifies the initiating switch (A), the destination switch (B), the trunk selected, called & calling nos & other information.

(3.) Switch A picks up one of its A link (AW) and transmits the message over the selected link to route to switch (B).

(4.) STP W receives the message, inspects its routing label and determines that it is to be ~~sen~~ routed to switch B. It transmits the message to switch B over a link (BW).

(5.) Switch B receives the message. On analyzing the message, it determines that it serves the called number and the called number is idle.

(6.) Switch B formulates an ACM (Address Complete Message), which indicates ~~that~~ that the IAM has received its destination. The message identifies the recipient switch (A), sending switch (B) & the selected trunk.

(7) Switch B picks on link BX & transmits ACM over that link for routing to switch ~~B~~ A. At the same time, it completes the call path in the backward direction (towards switch A), sending a ringing tone over the trunk towards switch A & rings the line of called subscriber.

(8.) STP X receives the message, inspect its routing label and determines that it is to be ~~sent~~ routed to A. It transmits the message to A.

(9.) On receiving ACM, switch A connects the called subscriber to the selected trunk in backward direction (to switch B) so that the caller can hear the ringing tone sent by switch B.

(10.) When the called subscriber picks up the phone, the switch B formulates an ANM (Answer Message) identifying the intended recipient switch (A), the sending switch (B) & the selected trunk.

(11.) Switch B select the same link BX to transmit ANM & transmits it to STP X. By this time, the trunk also must be connected ~~there~~ to the called line ~~so that~~ in both directions to allow conversation.

~~(12.) Switch B STP X recognizes that the calling subscriber is connected to the outgoing trunk & conversation can take place.~~

(12.) STP X recognizes that the ANM is addressed to switch A and forwards it over the same link AX.

(13.) Switch A ensures that the calling subscriber is connected to the outgoing trunk & that the conversation can take place.

(14.) If the calling subscriber hangs up first, switch A will generate a message REL (Release Message) addressed to B, identifying the trunk associated with the call. It sends REL to STP W over the link AW.

(15.) STP W receives REL message, determines that it is addressed to switch B and forwards to switch B over BW link.

(16.) Switch B receives the REL, disconnects the trunk from the subscriber line, returns the trunk to idle status, generates a release complete message (RLC) ~~as~~ addressed back to switch A and transmit it to STP X over link BX.

(17.) STP X receives RLC, determines that is addressed to A and forwards to switch A.

(18) On receiving RLC, switch A idles the identified trunk.

→ Electrical Caracteristics – as recommended by CCITT a recommendation in X-21 which also includes the mechanical characteristics.
→ Eg:→ MODEM, RS-232 etc.

TCP/IP Protocol:
→ TCP – Transport Control Protocol – Used within a network
→ TCP/IP – TCP/Internet Protocol – Used both within a network and between other networks in Internet.
→ TCP/IP makes data communication between any two computers anywhere in the world.
→ OSI Versus TCP/IP Protocols:

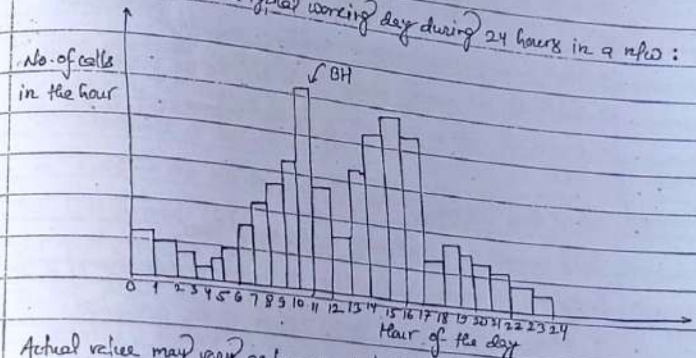
OSI		TCP/IP	
Application	→	HTTP	Application
Presentation		SMTP	
Session		DNS, RTP	
Transport	→	TCP, UDP	Transport
Network	→	IP (Routing)	Network
Data Line	→	ARP, RARP	Link
Physical		Cable, Microwave, o/f, satellite, wireless	

→ HTTP → Hyper Text Transfer protocol
  → used to send data on the web.
  → provides some advantages over FTP (File Transfer Protocol).
→ SMTP → Simple Mail Transfer Protocol
  → used for electronic mail.
→ DNS → Domain Name System
  → Translation of Host name as xam.tcp.edu.np as into an IP address as 128.135.40.0. via database lookup.
→ RTP → Real Time Protocol
  → used for real time data as on-line video, chat, etc.
→ UDP → Unreliable Datagram Protocol or User
→ TCP → Transport Control Protocol

#6: Telephone Traffic [9 Hours]

Introduction:
→ Provides the basis for analysis & design of telecommunication networks.
→ Blocking of a subscriber call in a telecommunication n/w are due to factors:
(i) Digit Receivers, (ii) Interstage switching links, (iii) Call Processors, (iv) Trunks between exchanges, (v) Subscriber line busy, etc.
→ Task of designing cost effective networks that provide the required quality of service under varied traffic conditions – Traffic Engineering / Teletraffic Theory.
→ Traffic engineering analysis enables / provides:
(i) To determine the ability of telecommunication network to carry a given traffic at a particular loss probability.
(ii) To determine the quantum of common equipments required to provide a particular level of service for a given traffic pattern & volume.

Network Traffic Load & Parameters:
→ The traffic load on a typical working day during 24 hours in a n/w:



→ Actual value may vary as per area but traffic pattern is the same.
→ 0-6 Hours – Little use of network when most population is asleep.
→ Large peak around mid-fore noon & mid-afternoon – busy traffic activities.
→ Low during lunch hour period: 12.00-14.00 Hours
→ Unpredictable peaks caused by stock market or money market activity, weather, natural disaster, international events, sporting events etc.
→ To account them, 3 types of Busy Hours are defined by CCITT in its recommendations E.600:

(1.) Busy Hour :⇒ Continuous 1-hour period lying wholly in the time interval concerned, for which the traffic volume or the no. of call attempts is greatest.

(2.) Peak Busy Hour :⇒ The busy hour each day; it is usually varies from day to day, or, over a no. of days.

(3.) Time Consistent Busy Hour :⇒ The 1-hour period starting at the same time each day for which the average traffic volume or the no. of call attempts is greatest over the days under considerations.

⇒ For ease of records, busy hour — 1 hour or ½ hour.

⇒ Not all call attempts materialise into actual conversations due to various reasons.

⇒ Some important Parameters :

(1.) Call Completion Rate (CCR) :

⇒ The ratio of no. of the successful calls to the no. of call attempts.

⇒ $CCR = \frac{\text{No. of successful calls}}{\text{No. of call attempts}}$ → successful if the called party answers.

⇒ Used in the dimensioning the network capacity.

⇒ Networks are usually designed to provide an overall CCR of over 0.70.

⇒ A CCR of 0.75 — Excellent ; further improvement is not cost effective.

(2.) Busy Hour Call Attempts (BHCA) :

⇒ Total no. of the call attempts in the busy hour.

⇒ Important parameter in deciding the processing capacity of a common control or stored program ~~concept~~ control system of an exchange.

(3.) Busy Hour Calling Rate :

→ The average no. of calls originated by a subscriber during the busy hour.

⇒ useful in sizing the exchange to handle the peak traffic.

⇒ Rural exchange — as low as 0.2, Business city — as high as 3.

(4.) Day-to-Busy Hour Traffic Ratio :

⇒ Ratio of busy hour calling rate to the average calling rate for the day.

⇒ May be over 20 for city business & 6-7 for a rural area.

(6.2)

(5.) Traffic Intensity (A_0) :

⇒ The traffic load on a given network may be on the local switching unit, interoffice trunk lines or other common subsystems.

⇒ For analytical treatments, all common subsystems are collectively — Servers.

⇒ The traffic on the network may be measured in terms of the occupancy of the servers in the network using term — Traffic Intensity as :

$$A_0 = \frac{\text{Period for which a server is occupied}}{\text{Total period of observation}}$$

⇒ Generally, period of observation is taken one hour.

⇒ A_0 - dimensionless — called E (earlang) to honor Danish Telephone Engineer A.K. Erlang who did pioneering work in this field — classical.

⇒ A server is said to have 1E of traffic if it is occupied for the entire period of observation.

⇒ Earlang measure indicates the average no. of servers occupied and is useful in desiring the average no. of calls put through during period of observation.

⇒ Also measured in another ways :

(i.) Centum Call Second (CCS) :

→ Represents a call-time product, valid only in telephone circuits.

⇒ 1 CCS — One call for 100 seconds duration or 100 calls for 1 second duration.

(ii) Call Seconds (CS) :

(iii) Call Minutes (CM) :

⇒ For present day networks which supports voice, data & many other services, earlang is ~~the~~ a better measure to use traffic intensity.

→ 1 E = 36 CCS = 3600 CS = 60 CM

⇒ Two important parameters required to measure traffic intensity are :

(i) Average call arrival rate — C
(ii) Average holding time per call — t_h
} → in like time units !!!

$\therefore A = C \cdot t_h$ → minutes/call

↳ no. of calls/minute

(6.3)

Some examples:

(1) An exchange serves 2000 subscribers. If the average BHCA is 10,000 & CCR is 60%, calculate the busy hour calling rate.

→ Average busy hour calls = BHCA × CCR = 6000 calls

Busy hour calling rate = $\frac{\text{average busy hour calls}}{\text{total no. of subscribers}}$ = 3.

(2) In a group of 10 servers, each is occupied for 30 minutes in an observation interval of 2 Hours. Calculate the traffic carried by the group.

→ Traffic carried per server = $\frac{\text{occupied duration}}{\text{total duration}} = \frac{30}{120} = 0.25\ E$

Total traffic carried by the group = $10 \times 0.25E$ = 2.5 E.

(3) A group of 20 servers carry a traffic of 10E. If the average duration of a call is 3 minutes, calculate the no. of the calls put through by a single server & the group as a whole in 1-hour period.

→ Traffic per server = $\frac{10}{20}$ = 0.5 E

→ A server is busy for 30 minutes in 1 hour.

No. of calls puts through by one server = 30/3 = 10 calls

Total " " " " " the group = 10×20 = 200 calls.

(4) A subscriber makes 3 phone calls of 3 minutes, 4 minutes & 2 minutes duration in a 1 hour period. Calculate the subscriber traffic in E, CCS, CM.

→ Subscriber traffic in erlang = $\frac{\text{busy period}}{\text{total period}} = \frac{3+4+2}{60}$ = 0.15 E

Traffic in CCS = $(3+4+2) \times 60/100 = 540/100$ = 5.4 CCS

" " CM = 3+4+2 = 9 CM

(5) Over 20 minutes observation interval, 40 subscribers initiate calls. Total duration of the calls is 4800 seconds. Calculate the load offered to the network by the subscribers & average subscriber traffic.

→ Mean arrival rate: C = 40/20 = 2 calls/minute

" holding time: t_h = 4800/40×60 = 2 minutes/call

= 2×2 = 4E

& average subscriber traffic = 4/40 = 0.1 E.

Grade of Service (GOS) & Blocking Probability:

Grade of Service (GOS):

→ In loss system, traffic carried by the network is generally lower than the actual traffic offered to the network by the subscribers.

→ The overload traffic is rejected & that amount is used as index of the quality of service offered by the network.

→ GOS = ratio of lost traffic to offered traffic.

→ Offered traffic – product of the average no. of calls generated by the users & the average holding time per call.

→ Carried Traffic – actual traffic carried by the network → average occupancy of servers → Given by A_o

∴ GOS = $\frac{A-A_o}{A}$, A = offered traffic, A_o = carried traffic

$A - A_o$ = lost traffic

→ The smaller the value of GOS, the better is the service. [0.002 in India].

→ The GOS of a full network is determined by the highest GOS value of the subsystem in it.

Blocking Probability (P_B):

→ Probability that all the servers in a system are busy.

→ When all the servers are busy, no further traffic can be carried by the system & arriving subscriber traffic is blocked.

→ GOS & P_B may seem the same – is not generally true.

→ Example: – In a system with equal no. of subscribers & servers, the GOS is zero.

→ There is a definite probability that all the servers are busy at a given instant & hence P_B is nonzero.

→ Difference: GOS – Measure from subscriber point of view ≡ Call Congestion.

P_B – a " " network / switching point of view ≡ Time Congestion.

→ Delay system:

→ Traffic carried by the network is the same as the load offered to the network.

→ Traffic is queued & all calls are put through the network until available.

→ Overload traffic is queued & all calls are put through the network until available.

→ GOS - not meaningful, so Delay Probability is a useful measure.

⇒ If the offered load or input rate or traffic far exceeds the network capacity, then the queue length becomes very large - undesirably long delays - Unstable.

⇒ One way to make stable condition - To behave like loss system - the queued up traffic is cleared to an acceptable limit - Called Flow Control.

⇒ Quality of service (QOS):

→ new term being used in more recent times & more general term GOS.

→ Includes GOS, speech quality, error-free transmission capability etc.

⇒ Subscriber viewpoint : GOS = call congestion = loss probability

Network viewpoint : Blocking Probability = Time Congestion.

Delay Systems & Queuing System/Model:

⇒ A class of telecommunication network (as Data Network) places the call or message arrivals in a queue in the absence of resources, until services them as & when resources become available.

⇒ Servicing is not taken up until the resources become available.

⇒ Such systems are called Delay Systems or Lost Call Delayed (LCD) Systems.

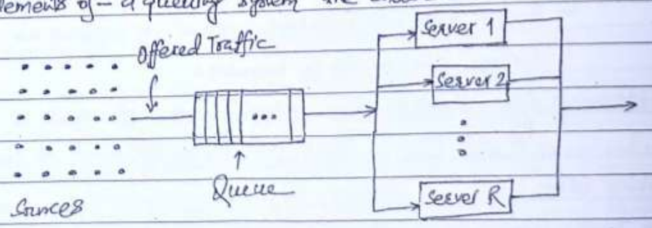
⇒ Delay systems are analyzed by Queuing Theory / Waiting line Theory.

⇒ Examples of delay systems in telecommunication include:

(i) Message switching (ii) Packet switching

(iii) Digit Receiver Access (iv) Automatic Call Distribution, (v) Call Processing

⇒ Elements of a queuing system are shown below:



⇒ When all the servers are busy, a request arriving at the network is placed in a queue until a server becomes available.

⇒ While analyzing queuing systems, we have to deal with a no. of random variables as listed below:

Random Variable	Notation
State of the system : No. of calls present in the system	K
Queue length : No. of calls / requests waiting to be served	K_q
No. of calls in service	K_s
Mean Wait Time	t_q
Mean Service Time	t_h
Mean arrival Rate	λ
Mean Service Rate	μ
Mean interarrival Time	τ
Prob. that there are K calls in the queuing system	P_K
Traffic Intensity : λ/μ	ρ
Server utilization : ρ/R	u

⇒ The no. of requests present in system = requests in queue + request being serviced

or, $K = K_q + K_s$

⇒ Mean time of a call or request spends = Mean Wait Time + Mean Service time (or holding)

$= t_q + t_h$

⇒ Queuing → enables better utilization of servers than does a loss system.

↳ has the effect of smoothing out the traffic flow as far as servers are concerned.

⇒ Need of infinite queuing capacity if there were to be no loss of traffic.

⇒ In a practical system, finite queue capacities are possible and hence, there is a probability (however small) of blocking in delay systems.

⇒ Assume : Infinite queue capacity, condition for its stable operation is

$$\frac{\text{Mean arrival rate}}{\text{Mean service rate}} < 1 \Rightarrow \frac{\text{offered traffic}}{\text{no. of servers}} < 1$$

⇒ If this condition is not satisfied, queue length - infinite - Unstable.

⇒ A queue system is characterised by six parameters due to D.G. Kendall in notation as A/B/C/K/m/Z :

A = Arrival Process Specification $\quad B$ = Service Time Distribution

c = No. of servers $\quad K$ = Queue Capacity

m = no. of sources (input population) $\quad Z$ = Service Descipline

⇒ Value of K & m → finite or infinite numbers.

⇒ Z → rule used for choosing the next customer to be serviced from queue.

→ Example:– First-come, First-service (FCFS), Random, Priority based etc.

⇒ K, m & Z → can be omitted from model – assume some default values.

→ Infinite for K & m and FCFS for Z as default.

⇒ c – nonzero +ve number

⇒ A & B – may assume any one of values shown below:

values for A & B	Meaning	Remark
GI	Arrival process with general independent distribution for interarrival time	For A only
G	General (no assumption) service time distribution	For B only
E_K	Erlang-K interarrival or " " "	
M	Poisson arrival & exponential " " "	M ≡ Markov
D	Deterministic interarrival or " " "	
H_K	Hyperexponential (with K stage) interarrival or service time distribution	

⇒ Eg: → M/D/4 = Poisson arrival, deterministic service time distribution, 4 servers, infinite sources & queue capacity & FCFS descipline.

⇒ Assumptions for analysis of delay systems:

(1.) Infinite source exists

(2.) " queuing capabilities

(3.) Queue is serviced on FCFS basic.

(4.) Poisson arrival process & Exponentially distribution of service time.

→ Start with a B-D (Birth – Death) process:

$\lambda_K = \lambda$ for $K = 0, 1, 2, \cdots$ ⇒ Call arrival rate in state K

$\mu_K = \begin{cases} K\mu & \text{for } K = 0, 1, 2, \cdots, R \\ R\mu & \text{for } K > R \end{cases}$ ⇒ Call termination rate " " "

⇒ R: no. of servers in the system

⇒ The stability condition of the system demands that

$$\frac{\lambda}{R\mu} < 1 \Rightarrow \frac{A}{R} < 1$$

& $P_0 + P_1 + \cdots + P_R + P_{R+1} + \cdots = 1$

⇒ $P_0 = 1 - \sum_{K=1}^{\infty} P_K$

→ From B-D process: $P_{K-1}\lambda_{K-1} + P_{K+1}\mu_{K+1} - (\lambda_K + \mu_K)P_K = 0$, $K \geq 1$

& $P_1\mu_1 - \lambda_0 P_0 = 0$ for $K = 0$

⇒ for $R = K$ ⇒ $R\mu P_{R+1} = (\lambda + R\mu)P_R - \lambda P_{R-1}$

$= (\lambda + R\mu)\frac{A^R}{R!}P_0 - \lambda\frac{A^{R-1}}{(R-1)!}P_0$

or, $R\mu P_{R+1} = \frac{\lambda A^R}{R!}P_0$

→ $P_{R+1} = \lambda/\mu R \cdot A^R/R!\, P_0 = A/R \cdot A^R/R!\, P_0 = A/R \cdot P_R$

⇒ for $K = R+1$ we get,

$P_{R+2} = (A/R)^2 P_R$

⇒ for $K > R$ ⇒ $P_K = (A/R)^{K-R} \cdot P_R$

⇒ We know, $A/R + (A/R)^2 + (A/R)^3 + \cdots = \frac{A/R}{1-(A/R)} = \frac{A}{R-A}$

$\therefore \sum_{K=0}^{R} A^K/K!\, P_0 + A/(R-A) \cdot A^R/R!\, P_0 = 1$

$$\therefore P_0 = \frac{1}{\sum_{K=0}^{R}\frac{A^K}{K!} + \frac{A^R}{R!}\cdot\frac{A}{R-A}}$$

$$\therefore P_R = \frac{A^R}{R!}\cdot\frac{1}{\sum_{K=0}^{R}\frac{A^K}{K!} + \frac{A^R}{R!}\cdot\frac{A}{R-A}}$$

Switching Hierarchy & Routing:

- Telephone networks requires some form of interconnection of switching exchanges to route traffic effectively & economically.
- Exchanges are interconnected by Trunk lines/Groups that carry traffic in one direction – so 2 trunk groups between any two exchanges.
- Three Basic topologies for interconnecting exchanges:
  - (1.) Mesh
  - (2.) Star or Double or higher order star
  - (3.) Hierarchical

  Refer chapter-1 for detail. !!!
- 3 methods are commonly used for deciding on the route for a particular connection:

(1.) Right - Through Routing:

- The originating exchange determines the complete route from source to destination.
- No routing decisions are taken at the intermediate routes.
- In absence of a computer, only a predetermined route can be choosen.
- There may be more than one predetermined routes & originating exchange may choose one considering time of day, distribution of traffic etc.

(2.) Own - Exchange Routing: OR Distributed Routing:

- Allows alternative routes to be choosen at the intermediate nodes.
- The strategy is capable of responding to changes in traffic loads and network configurations.
- Another merit is that when new exchanges are added, modifications required in the switch is minimal.

(3.) Computer - Controlled Routing:

- Computers are used in networks with CCS features – SS7.
- With computers in positions, a no. of sophisticated route selection methods can be implemented.
- Computer based routing is a standard feature of data networks.
- [Detail – chapter 8].

Numbering Plans:

Transmission Plan:

- CCITT lays down certain guidelines in its recommendation Q.40:

(1.) Max. no. of circuits to be connected/used in international call is 12.

(2.) No more than 4 international circuits be used in Tandem between the originating & the terminating international switching centres.

(3.) In exceptional cases & for a low no. of calls, the total no. of circuits may be 14, but even in this case, the international circuits are limited to a max. of 4.

Numbering Plan:

- Objective – To uniquely identify every subscriber connected to a telecommunication network.
- In early stage – numbering scheme was confined to a single local exchange & exchanges were identifies by the names of resident cities.
- The introduction of STD (Subscriber Trunk Dialing) or DDD (Direct Distance Dialling) for intercity & intertown long distance connections called for a National Numbering Plan where multiexchanges areas are identified uniquely by numbers.
- Subsequent development of ISD (International Subscriber Dialling) makes it necessary to have an International Numbering Plan.
- 3 types of National numbering plan:

(1.) Open or Non-uniform Numbering Plan:

- Permits wide variation in the no. of digits to be used to identify a subscriber within a multiexchange area or within a country.
- Used in countries equipped extensively with non-Director Strowger Switching.
- This scheme is usually an exact images of the network structure - almost extinct.

(2.) Semi-open Plan:

- Permits number lengths to differ by almost one or two digits.
- Most common used in India, Sweden, Switzerland & UK.

(3.) Closed or Uniform Numbering Plan :

- The no. of digits in a subscriber number is fixed.
- Used by few countries as France, Belgium & North America (USA, Canada, Hawaii etc).

- An international/World numbering plan has been defined by CCITT in its recomendations E.160 – E.163.
- The world has been divided into nine geographical – each global regin within each country code starts with the same digits as shown below:

Zone Code	Zone	Example Country	Country Code
1	North America, caribean (23 Countries)	Canada	1
		USA	1
2	African continent (61 countries)	Egypt	20
		South Africa	27
3	Europe (34 Countries)	Italy	39
		France	33
4	Europe (15 Countries)	Germany	49
		UK	44
		Czech Republic	420
5	Central & South America (28 countries)	Costarica	506
		Brazil	55
6	Oceania, South Pacific (32 countries)	Malasia	60
		Singapore	65
7	Former USSR (2 countries)	Khzakhstan	7
		Russian Federation	7
8	East Asia (21 countries)	Japan	81
		International free phone service	800
		Bangladesh	880
9	Middle east, South-West Asia	India	91
		Pakistan	92
		Iraq	964
		Nepal	977

(6.12)

- An international telephone no. starts with 1-3 digits country code followed by 9-12 subscriber no. with dialling procedure with prefix '00' as:

← 15 digits max →

Country Code	National Telephone No.
← 1-3 digits →	← 9-12 (max) →

- National Numbering Plan :

- The dialling procedure are of two categories: Closed & open
(1) Closed Plane :→ Refers to a numbering plan which requires users to dial all numbers at all times. This means, that local-local calling also requires the area code to be dialled as well as trunk prefix.
(2) Open Plan :→ Local calls can be placed without the trunk prefix and area code.

← Subscriber Number →

Area/Trunk Code	Exchange Code	Line Number

← National Significance Number (NSN) →
↳ 12 digits max.

- Customer Local Number = Exchange + Line Number
National Distribution Code (NDC) = Area Code + Exchange Code
- Prefix for national – '0'.

Charging Plan :

- The cost of providing a telecommunication network consists of 2 costs:
(1.) Capital costs :→ includes Switching Systems, Building, lines & ~~fields~~ lands.
(2.) Operating costs :→ " Staff Salaries, Maintenance costs, Water & electricity charges & Miscellaneous Expenses.
- All these costs must be met by the income obtained by the telecom operator from its subscribers.
- The telecom operator charges its subscribers for its services by 3 ways:
(1) An initial charge for providing a network connection (Installation charge)
(2) A rental or leasing charge
(3) Call charges.
- The initial costs are covered partly from installation charges & partly from rental.

(6.13)

The operating costs are recovered through rental & call charges.

- The quantity of equipments used, routing exchanges, switching systems, lines carrying voice / data & human involvement in establishing a connection between subscribers differs wrt distance between subscribers, the time of which the call is made (busy hour or off peak hour), the area etc.
- The charging methods for individual calls — 2 categories:
  (1.) Duration Dependent charging: — Periodic Pulse Metering
  (2.) Duration Independent charging: — One or More Pulse Metering
- Traditionally, charges for long distance calls have been proportional to distance multiplied with duration.
- The local calls within a numbering area are usually charged on a duration independent basis.
- A meter for each subscriber counts the number of charging units based on the service providers policy decision.
- As the billing procedure changes time to time according to the Govt. policies & to meet

8: Data Communication [10 Hours]

Switching Techniques in Data Communication:

- Communication between two computers at a distance is achieved by transmitting data from source to destination through a network of intermediate nodes.
- The nodes may be telecom exchange, connection/ ~~the~~ switching devices, etc.
- Nodes are not concerned ~~with~~ with content, rather their purpose is to provide a switching facility that will move the data from node to node until the content reaches at its destination.
- Terminal → which sends/ receives information or station
  → eg:→ Computers, Telephones or other communication Devices.
- Collection of nodes - Communication Network.
- Two types of switching techniques:
  (1.) Circuit Switching
  (2.) Store & Forward (S&F) Switching $\xrightarrow{2}$ Message Switching
  → Packet Switching → Datagram / Virtual Circuit.

Circuit Switching:

- An electrical path is established between source & destination before any data transfer takes place. through the nodes of the network.
- The electrical path may be realized by physical wires or coaxial cables or radio or satellite links.
- It remains dedicated to the communicating pair for entire duration of transmission, irrespective of whether data is actually transferred or not.
- No other user can use the path even if it is idle.
- The connection is released only when ~~a~~ specifically signalled so by either of the communicating entities.
- eg:→ Data transmission using a PSTN connection.
- 3 explicit phases involved in the circuit switching:
  (i) Connection Establishment → (T_e)
  (ii) Data Transmission (T_t)
  (iii) Connection Release. (T_r)

  $\therefore \; T_{cs} = T_e + T_t + T_r$

Disadvantages:

(i) The path set-up time (20-30 s) is an excessive overhead for bursty computer

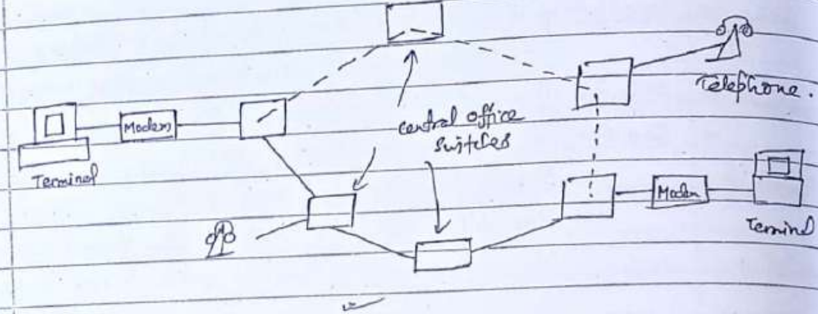
(ii) Subscriber end links are poor quality link whereas other links in PSTN are high quality (> 95%) causing entire circuit of poor quality - last mile.

(iii) Speed of operation is limited by the slowest link in the circuit.

(iv) No error control facilities which are to be handled by the end systems.

(v) Reserved bandwidth — wastage of unused bandwidth.

⇒



(2.) Store & Forward (S&F) Switching:

⇒ The switching nodes have the ability to store user messages & forward the same towards the destination as & when the links available.

⇒ Thus, each node is equipped with a processor & some buffer storage.

⇒ No end-to-end link is set up prior to data transmission.

⇒ The user deposits his/her message to the nearest switching node and then on, the network takes the responsibility for delivering the message to the destination user or host.

⇒ The network moves the user information from node to node — called Hop.

⇒ Since the comm. links are used one at a time between any two nodes, line speeds can be utilized efficiently.

⇒ Two techniques:

(i) Message Switching:

⇒ Once the transmission is initiated, a message is transmitted in its entirety without a break from one node to another.

⇒ The node processor performs the following functions:

(a) Receive the full user message & store the same.

(b) Check for transmission errors & perform recovery if necessary.

(c) Determine the destination address from the user message.

(d) Choose appropriate link towards destination based on routing criterion.

(e) Forward the message to the next node on the chosen link.

⇒ Drawbacks:

(a) Long message requires adequate storage space on the receiving node before initiation of transmission. Otherwise, part of message may not be stored, thus requiring retransmission of the entire message.

(b) Retransmission of entire message in case of transmission errors.

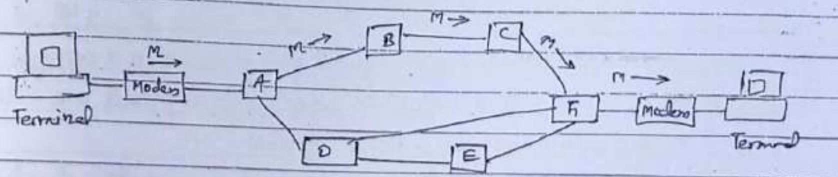
(c) High priority short message have to wait if a longer message is in transmission.

→ Advantages:

(a) Links are shared among communication devices — better b/w u[illegible]

(b) Priorities may be used to manage network traffic.

(e)



(ii) Packet Switching:

⇒ Messages are split into a no. of packets, often in fixed size and the packets are transmitted in a S&F fashion.

⇒ Messages are split at the source host & reassembled at the destination host.

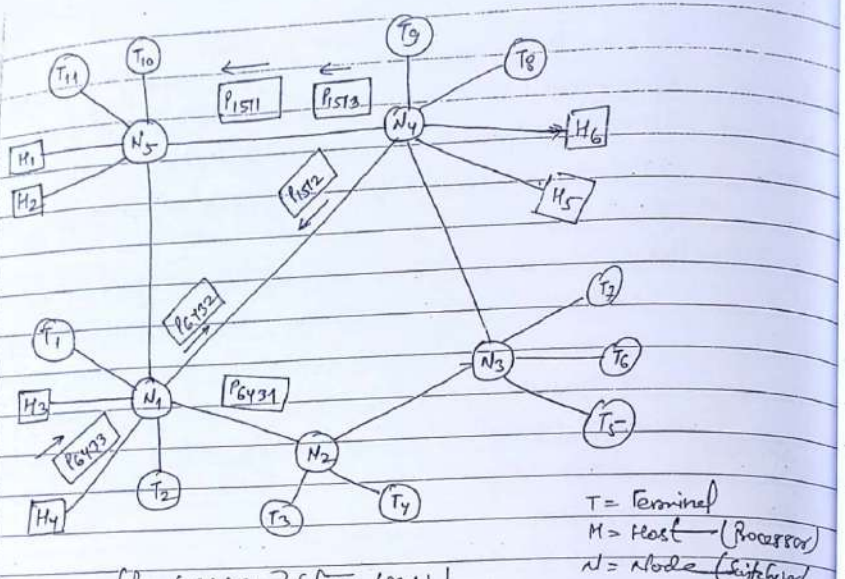
⇒ Each packet transmission is independent of the others.

⇒ The packets of a single message may travel via different routes and arrive at the destination with different delays.

⇒ This may lead to the situation where the packets of same message arrive out of sequence at the destination node.

⇒ Every packet needs to carry the complete address information, such as destination identifier, source id, message id & packet id along with the actual user data.

⇒ See the figure on the next page:



T = Terminal
H = Host (Processor)
N = Node (Switching)

(Packet Switching Network (PSN))

⇒ Eg:→ P6433 : Packet no. 3 of 3rd message destined to host 6 from host 4.
P1513 : " " 3 " 1st " " " " 1 " " 5.

⇒ The source host delivers the packets of a message in a sequence to the network node & it is natural to expect that the packets are delivered to the destination host in a proper sequence.

⇒ However, as the packets may arrive out of sequence at destination node, it becomes the responsibility of the network to resequence the packets before delivery to the destination host.

⇒ So, considerable overhead in terms of buffer storage & processing power at the network nodes — Expensive service.

⇒ To be cost-effective, packet networks offers 2 forms of services:

(a) Datagram service:

→ also called Connection-less packet switching, no need of resequencing by n/w.

⇒ normally used for transmitting messages of one or two packet lengths.

⇒ Each packet includes complete routing or addressing information.

⇒ The packets are routed individually & may be out of order of sequence.

(b) Virtual circuit service:

→ also called Connection-oriented packet switching, no need of resequencing by n/w.

Routing or Packet Routing:

⇒ [See chapter 6 also for definition & classification of Routing]

⇒ The main function of network layer (3rd layer of OSI model) is routing packets from source to destination host.

⇒ The routing algorithm is that part of the network layer software which is responsible for deciding which output line an incomming packet should be transmitted on.

⇒ In datagram, routing decision must be made a new one for every arriving data packets since the best route may have changed since last time.

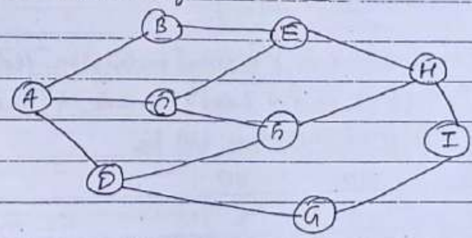
⇒ In virtual circuit, routing decisions are made only when a new virtual circuit is being set-up — also called Session Routing.

⇒ Can be grouped in two major classes:

(i) Static / Non-adapting Algorithm:

⇒ In this algorithm, the routing table is fixed and does not vary with the traffic pattern.

⇒ Consider the following network:

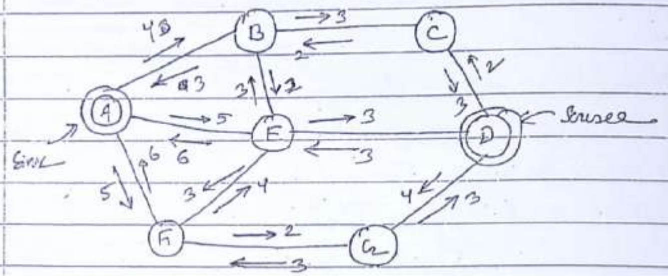


⇒ The static routing table for node C can be generated as:

Packet Destn.	First Choice Node	2nd Choice Node
A	A	F
B	A	E
D	A	F
E	E	A
F	F	A
G	F	A
H	E	F
I	E	F

⇒ No need of 3rd choice node as no node has more than 3 paths going from it.
⇒ If the first choice path is blocked, a node will use the second choice node.

(2.) Dynamic / Adaptive Routing :
⇒ This algorithm uses a table which changes as the traffic pattern / loads change, attempting to give the best route at different times.
⇒ So called because network adapts its behaviour to changing traffic load.
⇒ Powerful, sophisticated and most common for several n/w as ARPA.
⇒ Consider the following configuration

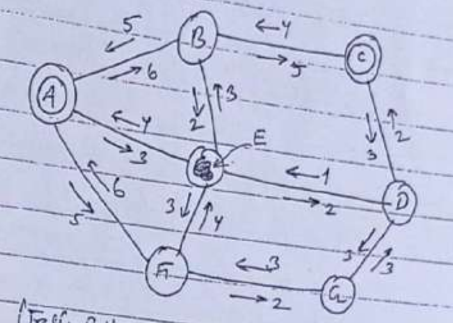


⇒ The figures on each path is proportional to delays on that link.
⇒ Let node D wants to send a packet to node A (final destination).
⇒ The table for node D at this time will be

FD	NN	OD
A	C	7
B	C	4
C	C	2
E	E	2
F	E	6
G	G	4

FD : Final Destination
NN = Next Node
OD = Overall Delay

⇒ The best route is : D → C → B → A with min. delay of 7.
⇒ The packet from D reaches node C and now the traffic pattern on the network changes as shown in the next page :



(Traffic Pattern as packet arrives C)

FD	NN	OD
A	D	8
B	B	4
D	D	3
E	D	4
F	D	7
G	D	7

(Node C Routing Table)

⇒ Now, the optimum route is C → D → E → A with OD = 7
⇒ The packet returns to node (original) D without making any progress towards reaching the final destination node A.

⇒ Thus, this technique is refined enough to handle this kind of problem requiring complex system.
⇒ There may be : Duplication, lost & looping of packets.
⇒ To overcome this problem, a hop count is used in control field or protocol field of packet format as shown earlier in packet switching.

Flow Control :

⇒ Routing & flow control are two closely related operational requirements of any communication networks.

⇒ Same basic principle for controlling flow in both circuit- & packet- switching.

⇒ Flow control in a packet network, is primarily concerned with the buffer management.

⇒ If all store-&-forward buffers in adjacent nodes become filled with packets destined to each other, the nodes are unable to receive additional packets, and a deadlock exits.

⇒ A node does not release a buffer as soon as it transmits a packet.

⇒ The buffer is released when an acknowledgment is returned from the adjacent node.

⇒ If a receiving node has no available buffers, it cannot accept a new packet & therefore cannot acknowlege it.

⇒ Flow control requirements emply that interface nodes in a packet-switched network are aware of overload conditions and refuse any new requests for services until the congestion is relieved.

→ Virtual circuits are an attractive/effective means of controlling the flow of multiple-packet messages, but they represent too much overhead for a single packet or datagram flow control.

⇒ The packet might as well be sent immediately and be considered its own circuit setup message.

→ In store-&-forward buffers, however, a single-packet message is much different from a call establishment packet.

→ A call establishment packet requires a certain amount of storage and processing by the call processor of each network node it reaches, but it does not compete for store-&-forward buffers as does a message packet.

⇒ A conventional circuit-switched network is unconcerned with flow control between the endpoints of a connection, since once the circuit is established, the activity or inactivity of endpoints has no effect on other connections or on the network as a whole. (8.9)

→ The route from source to destination is fixed for all packets of a message.

→ The packets of a message need not carry the full address information as the packets follow the same path/route and are delivered in sequence.

→ The transmission of packets may not start until & unless a route/circuit is chosen & finalised between the source & destination.

→ The circuit so choosen is not dedicated to a particular ~~station~~ connection, & the same route & the circuit may be used for transmitting packets from different sources — so called Virtual Circuit.

Packet Format:

byte count ↓

Start Frame	Destination Address	Source Address	Sequence Number	Protocol or Control Field	Information (User-Data)	Error control	End Frame

|← Header →| |← Trailer →|

→ Start Frame : To identify the beginning of a packet.

→ End Frame : To signify the end of a packet.

→ Header : To maintain control of the packets over the network.

→ Information : contains a segment of a total text/message to be transmitted.

→ Sequence No. : To ensure that packets can be reassembled in the proper order at the destination.

→ Control Field : To prevent duplication, loss or looping (in which packets is routed back & forth in an endless circle) of the packet.

→ Error Control : Use to verify packet integrity.
To clear any transmission error, generally CRC is used.

→ Source Address : To identify the host originating the packet.

→ Destination " : To " " " destined to receive the packet.

ISDN (Integrated Services Digital Network):

→ ISDN is a set of digital transmission standards which are used for end-to-end digital connectivity.

→ Integrated Services = ability to sustain numerous applications.

→ Extends from local telephone exchanges to the remote user and include all the telecommunications & switching equipments in between.

→ In past, video, audio, voice & data services required atleast four separate networks but ISDN integrates all fours over the same network.

→ Developed originally to address the problem of how to transport digital services across a telephony infrastructure based on copper wiring originally intended to carry analog signals only during 1970's.

→ Initially, IDN (Integrated Digital Networks) then ISDN - all customers will become digital rather than analog.

→ Standards:

→ ISDN is standardized according to recommendations of CCITT, now ITU, which describes the protocols & architectures to implement a world wide digital communications networks as listed below:

(i) SG VII : Public Data Network (X.25) X-series Standards
(ii) SG VIII : Terminal Equipment for telematic services
(iii) SG XI : ISDN & telephone network switching & signalling
(iv) SG XII : Transmission performance of telephone n/ws & terminals
(v) SG XV : Transmission systems
(vi) SG & XVII : Data Transmission over public telephone networks
(vii) SG XVIII : Digital network including ISDN

[SG = Study Groups = comprised by CCITT]

→ Types of ISDN : ⇒ 2 types of ISDN

(i) Narrow Band ISDN (N-ISDN):

→ Carry data rating upto 64 kbps, ranging upto T1 rates.

→ Sometimes used to refer a regular telephone & nonvideo capable systems.

(ii) Wide-band ISDN (W-ISDN):

→ The communication standards being developed by the ITU to handle the high bandwidth applications such as video.

→ It will use ATM technology over SONET based transmission units to provide data rates of 155 Mbps to 622 Mbps and beyond.

ISDN Services:

⇒ ISDN services generally fall into 3 categories:

(1.) Bearer Services:

⇒ ISDN works on the principle of transport services called bearer services.

⇒ The bearer services offers the capability to transport digital voice or non-voice services using this standard.

⇒ The basic operation of the bearer service is the 64 Kbps channel capacity.

⇒ Bearer services provide the means to transfer information (voice, data & video) between users.

⇒ The network does not need to process the information.

⇒ Bearer service belongs to the first three layers of OSI model.

⇒ These services can be provided with circuit switched, packet switched, frame switched or cell switched networks.

(2.) Tele Services:

→ In this service, the network may change or process the contents.

→ This service corresponds to layers 4-7 of the OSI model.

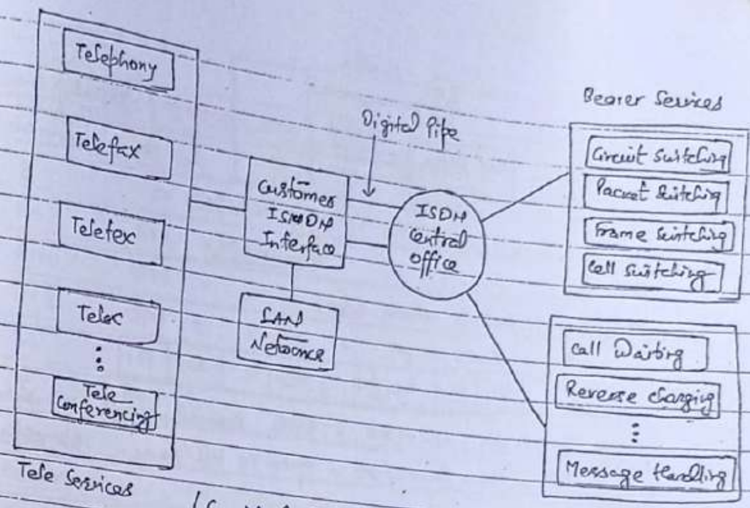
→ They include telephony, telefax, videofax, telex & teleconferencing.

(3.) Supplimentry services:

→ It provides additional functionality to the bearer services & teleservices.

→ They include call waiting, Reverse charging & message handling.

The conceptual view of ISDN is shown on the next page:



(Conceptual View of ISDN)

Advantages of ISDN:

(1) High speed service

(2.) Cost advantage

(3) High quality transmission

(4.) Simultaneous transmission

(5) Multiple device connection

(6) Conferencing, (7.) Call Identification feature

(8.) Call Management features ⇒ (i) Call forwarding (ii) Call Pickup (iii) Directed call Pick up (iv) Directed Dial (v) Ringing options (vi) Additional call offering (vii) Message Waiting indicator.

ISDN Interfaces:

⇒ Two main types of interfaces: BRI & PRI

(1) Basic Rate Interface (BRI):

⇒ made up of 2 B-channels & 1 D-channels with total rate of 2B+D.

⇒ B channels are 64 Kbps & can be used for voice & data communications.

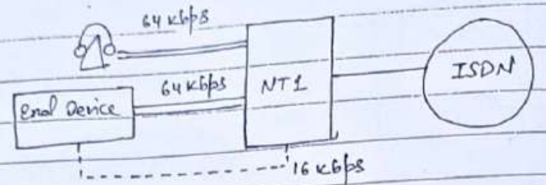
⇒ D channels is 16 Kbps & is used for call initialization & signalling connection.

⇒ Most appropriate type of ISDN service for computer connections & individuals.

⇒ Each of 3 channels (2B+D) can be used simultaneously.

⇒ Designed to carry the most data possible to the home through existing copper wires.

⇒ The concept is shown on the next page:



(BRI Concept)

⇒ BRI format is shown below:

bits	8	1	8	1	8	1	8	1
	B1	D	B2	D	B1	D	B2	D

⇒ B1 : 16 bits, B2 : 16 bits, D : 4 bits, Overhead : 12 bits (not shown)

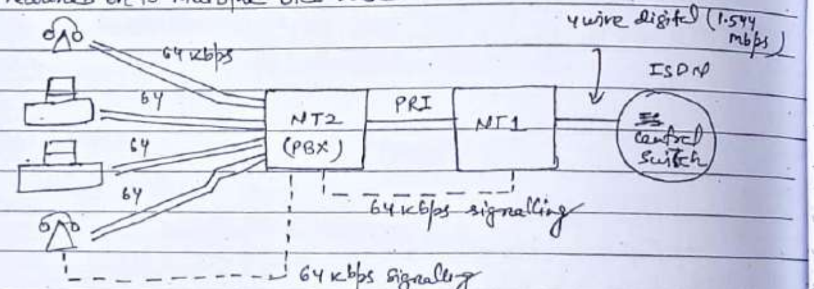
⇒ BRI = 4000 frames/sec = 4000 × 48 bits/frame = 192 kbps.

(2) Primary Rate Interface (PRI):

⇒ PRI in North America has 23 B channels & one 64K D channels or the total rate a is 23 B + 1 D, total b/w of 1.544 Mbps.

⇒ PRI in rest of world uses 30 B channels & 1 D channel or 30B + D with a total rate of 2.048 Mbps.

⇒ Unlike BRI, PRI does not support a bus configuration and only one device can be connected to a PRI line, however, a PBX can reallocate ISDN PRI resources on to multiple BRI buses.



(PRI Concept)

⇒ PRI frame format is shown below:

1 bit	8 bits	8 bits	8 bits	8 bits	…	8 bits
					…	

1 frame < 193 bits, 1D = 8 bits, 23B = 23×8 = 184 bits

Overhead = 1 bit.

PRI = 8000 frames/sec = 8000 × 193 bits/frame = 1.544 Mbps

(8-13)

ISDN Channels: 3 types – B, D & H

Channels	Bit Rate	Interface	Purpose
B	64 kbps	BRI	Bearer Services
H0	384 "	PRI	6 B channels
H11	1536 "	"	24 B channels
H12	1920 "	"	30 B channels
D	16 "	BRI	Administrative & Control Signalling
D	64 "	PRI	" & " "

(1) B channels:

⇒ Bearer channels – logical channels digital pipes which exist on a single ISDN.

⇒ Carry data & voice services at 64 kbps.

⇒ full duplex mode, end-to-end physical circuit that is temporarily dedicated to transferring data between two devices.

(2) D channels:

⇒ Delta or Signalling channels, can be either 16 or 64 kbps

⇒ To carry control signalling & administrative informations for B channel to set up & tear down the calls.

⇒ use packets-switched connection.

⇒ associated with higher level protocols at layers 2 + 3 of OSI model.

⇒ provides the signalling information that is required for caller identification, low-data rate transfer & applications as telemetry & alarm transmission.

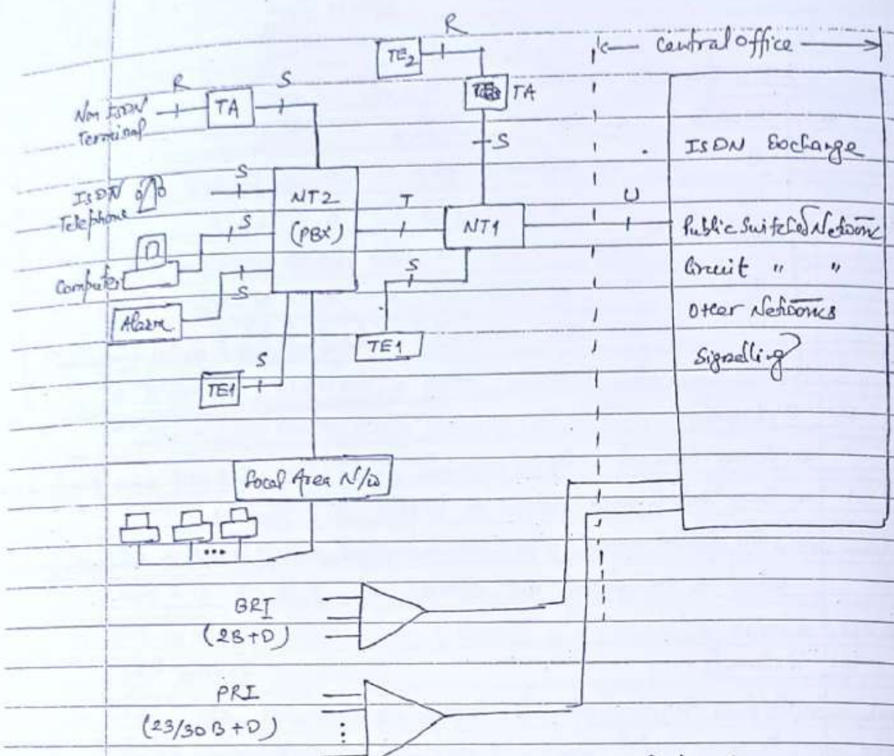
(3) H channels:

⇒ Hybrid channels, suitable for high data rate applications such as video, teleconferencing & so on.

Architecture of ISDN Network:

⇒ shown on the next page:

(8-14)



(ISDN Architecture)

⇒ NT1 (Network Terminator Type 1):

- Digital interface point between network & customer's equipment.
- Provides physical & electrical terminal and acts as a multiplexer & demultiplexer.
- can connect upto 8 ISDN devices (telephone & computers).

⇒ NT2 (Network Terminator Type 2):

- Handles switching & multiplexing (such as PBX)
- Performs functions at the physical, data link & network layers of OSI model.
- Provides intermediate signal processing between data generating devices & an NT1.
- Coordinates transmission from a no. of incoming links (user phone lines) & multiplexes them transmittable by an NT1.

⇒ TE1 (Terminal Equipment 1):

⇒ TE1 are ISDN terminals like video conferencing equipment, Group-4 fax, feature telephone which are digital & can be directly connected to NT through 'S' bus interface.

⇒ TE2 (Terminal Equipment 2):

- They are non ISDN terminals such as analog phone, PC, G3, FAX, which are non-digital & can not be directly connected to NT1.
- They require another interface called TA.

⇒ TA (Terminal Adapter):

- TA enables analog to digital conversion & vice-versa.
- Otherwise, the purpose of TA is to ~~connect~~ convert the bipolar signalling from the public network to the unipolar signalling used by computers.
- The most ~~prevalent~~ prevalent use of TA is to connect a computer to an ISDN line.

⇒ CCITT has defined 4 interface points: R, S, T & U.

(1.) U-interface ⇒ connection between ISDN exchange & NT1.
(2.) T-interface ⇒ " " NT1 & NT2
(3.) S-interface ⇒ " " NT1 with TE1 or TA
⇒ " " NT2 with TE1
(4.) R-interface ⇒ " " TA & TE2 (non-ISDN Terminals).

Broadband ISDN (B-ISDN):

- provides needs of the next generation technology.
- Digital service with speed > 1.544 Mbps
- Two types of services: Interactive & Distributive
  - Interactive ⇒ Conversational & message or retrieval, Bidirectional
  - Distributive ⇒ Unidirectional with/without user control.
- Several forms of B-ISDN:

(1.) Frame Relay service.
(2.) Switched Multimegabit Digital Service (SMDS)
(3.) ATM

Digital Subscriber Line (DSL) Technology :

- xDSL – a generic abbreviation for many variations DSL technology.
- DSL refers to the technology used between a customer premises and the telephone company enabling more bandwidth over the already installed copper cabling.
- Thus xDSL is a technology backed by telephone companies to provide next generation high bandwidth service to the home & business using the existing telephone cabling infrastructure.
- This technology accomplishes high speed delivery of data, voice, video and multimedia.
- Various DSL technologies are listed below:

DSL Technology	Speed Down Stream	Speed Upstream	Operation
ADSL	1.5–1.92 Mbps	16–640 kbps	one pair of wire
RADSL	64 kbps – 8.192 mbps	16–768 kbps	" " " "
CDSL	1 mbps	16–128 kbps	Now ratified as DSL-lite No splitter, one pair of wire
HDSL	1.544 mbps (NA) 2.048 mbps (Rest)	1.544 mbps 2.048 "	Symmetrical Service Two pair of wires
VDSL	13–52.6 mbps	1.5–6 "	Fiber needed & ATM N/W probably used
IDSL or ISDN DSL	14.4 kbps (as BRI)	144 kbps (as BRI)	Symmetrical operation one pair of wires
SDSL	1.544 mbps 2.048 "	1.544 mbps 2.048 "	one pair of wires one pair of wires targeted
SHDSL	2.312 "	192–384 "	the residential subscribers

ADSL = ~~Asynchronous~~ ~~Asy~~ Asymmetric DSL
RADSL = Rate Adaptive DSL
CDSL = Consumer DSL | SHDSL – Single Pair High bit rate DSL
HDSL = High bit rate
VDSL = Very high speed DSL
SDSL = Single DSL

Principle of operation of xDSL :

- This technique uses greater range of frequencies over telephone cable than the traditional telephone services used.
- By using freq. range higher/above the telephone bandwidth (300–3400 Hz), xDSL can encode more data to achieve higher data rates.
- For this, xDSL equipment must be installed on both ends.
- The loading coils used in copper wires between ~~of~~ customer premises & local exchange must be removed ~~to~~ or avoided to enable the copper wires to pass higher frequencies over the entire wire.
- The loading coils are inductors added in series with the copper wires to compensate the parallel capacitance of the line.
- This coils limits the bandwidth in the copper line.

- The bit rate may differ for upstream & downstream.
- Upstream → Transfer of information from subscriber to local exchange.
- Downstream → " " " to " from " " .
- Equal upstream & downstream → Symmetric.
- Unequal " & " → Asymmetric or Rate Adaptive or Uneven.
- Most xDSL – Asymmetric – higher downstream than upstream.

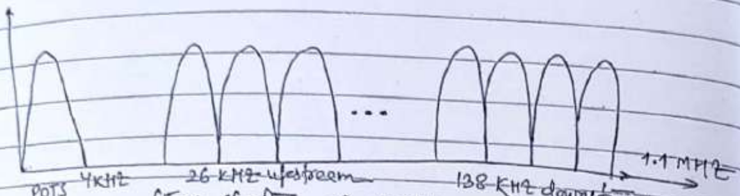
Asymmetric DSL (ADSL) :

- The DSL Forum was formed in December 1994 to promote the DSL concept & facilitate development of DSL system architecture, protocols and interfaces for major DSL applications.
- The ANSI approved the first ADSL in 1995 with data rate upto 6.1 mbps.
- ADSL, a modern technology, converts existing twisted pair telephone lines (subscriber loop) into access paths for multimedia and high-speed data communications.
- Can transmit upto 6 mbps to a subscriber & as much as 832 kbps or more in both directions.

~~ADSL~~

ADSL Frequency Spectrum :

⇒ ADSL divides the bandwidth of a twisted pair cable into 3 bands as:



(Freq. spectrum of ADSL)

⇒ Earlier, FDM or Echo cancellation are used to divide the available channels.

⇒ Presently, ADSL uses DMT (Discrete Multitone) encoding methods, which use QAM to divide the bandwidth of the channel into multiple sub-channels with each channel transmitting information using QAM.

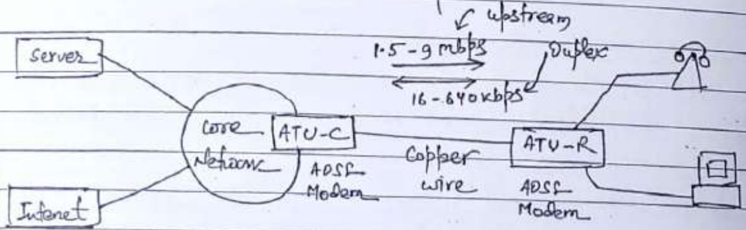
⇒ DMT uses freq. spectrum from 26 KHz to 1.1 MHz for broad band data.

⇒ 0-4 KHz → used by POTS or PSTN to carry voice.

⇒ 26-138 KHz → upstream = 25 channels } Total channels = 249
138-1100 KHz → downstream = 224 " } Bandwidth of a channel = 4.3 KHz

Topology for ADSL System :

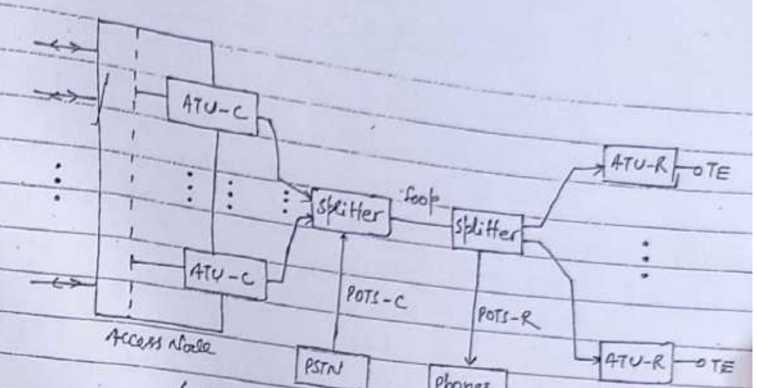
⇒ ADSL modem is connected to each of end of twisted-pair, one at subscriber end & other at the central office as shown



(ADSL Modem Connection)

⇒ ATU-C ⇒ Modem at exchange = ADSL-Terminal Unit Central Office
ATU-R ⇒ " " subscriber = ADSL - Remote.

⇒ Figure on the next page shows the topology of ADSL system :



(Topology of ADSL system)

⇒ Access Node → connects different types of services as Digital Broadcast, Broadcast N/W, Narrowband N/W, Network Management etc.
→ Provides interfacing of broadband services of ATU-C.

⇒ ATU-C converts the data into ADSL format.

⇒ The ADSL format fed to the splitter multiplexes them onto a single loop line.

⇒ Telephone connection from PSTN enter the system at the splitter level and are added to POTS-C (POTS-Central office) area of ADSL spectrum.

⇒ The splitter in POTS-R (POST - Remote) demultiplexes & transfers phone calls to the phones.

⇒ The ADSL formats are transferred to ~~ADSL~~ ATU-R which in turn converts to the original formats & supplies to the terminal ends (TE)

Advantages of ADSL :

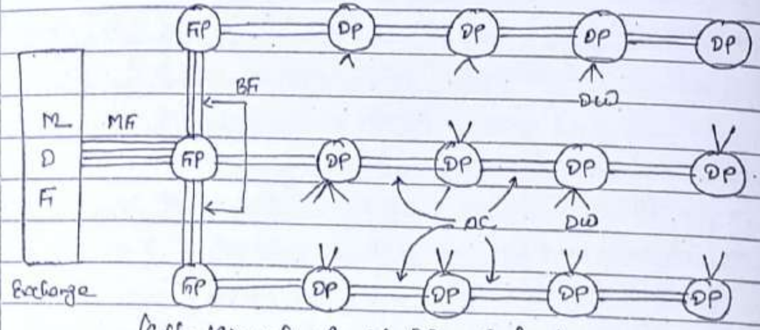
⇒ (1) Simple & affordable mechanism to get more bandwidth to end users, both residential and small to medium businesses.

(2) High speed downstream is increasingly important for internet access, remote access to corporate server, integrated voice/data access and transparent LAN interconnections.

(3) Enables carriers to offer value added, high speed networking services.

Subscriber loop System

- Every subscriber in a telephone network is connected generally to the nearest switching office by means of a dedicated pair of wires – Subscriber loop.
- ~~Unwid~~ It is unwieldy to run physical independent pairs from every subscribers premises to the exchange.
- Far easier to lay cables containing a no. of pairs of wires for different geographical locations & run individual pairs as required by the subscriber.
- Generally, 4 levels of cabling are used as shown below:



(Cable Hierarchy for Subscriber Loops)

MDF = Main Distribution Frame, FP = Feeder Point

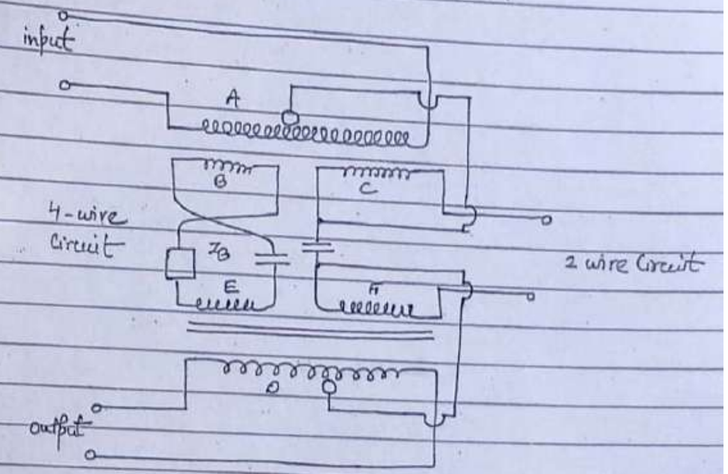
DC = Distribution cable, DP = Distribution Point, MF = Main Feeder

BF = Branch Feeder, DW = Drop wires

- DW ⇒ are the individual wires that run into subscriber premises.
- DP ⇒ Drop wires are connected to wire pairs in the distribution cables (DC).
- FP ⇒ Many distribution cables from nearby geographical locations are terminated, where they are connected to BF cables, which, in turn, are connected to the MF cable.
- MF ⇒ carry a large no. (100–2000) of wire pairs than the distribution cables that carry 100-500 pairs.
- MDF ⇒ The feeder cables are terminated ~~at~~ to MDF at exchange.
- Subscriber pairs & exchange pairs are interconnected ~~by~~ at MDF by means of Jumpers, providing a flexible interconnection mechanism which is very useful in reallocating cable pairs & subscriber numbers.

Transformer Based Hybrid Circuits:

- Digital exchanges require receive & transmit signals on separate 2-wire cir
- This calls for two-wire to four-wire conversion.
- Such a conversion is normally required for trunk transmission in analog exc
- The circuit that performs 2-wire to 4-wire conversion is called Hybrid.
- A transformer based hybrid circuit is shown below:



(Two-wire to Four-wire Transformer Hybrid)

- The main function of hybrid is to ensure that there is no coupling of signal from the input to the output in 4-wire circuit.
- The input signal is coupled to B & F windings equally.
- Through C windings, the input is coupled to the 2-wire circuit.
- The same signal when it flows through the balanced 2-wire couples the signal to winding D through winding C.
- The signal induced in B flows through E & induces a current in D that opposes the current induced by F.
- If the impedance Z_B is exactly matches that of 2-wire circuit, the effect of input signal on the output winding D is completely ~~null~~ nullified.
- In a similar way, the input signal from the subscriber end is completely nullified from coupling into winding A.
- Test function enables forward test of the loop or inward test of the switch.

→ Electrical characteristics – as recommended by CCITT a recommendation in X-21 which also includes the mechanical characteristics.

→ Eg:→ MODEM, RS-232 etc.

TCP/IP Protocol :

⇒ TCP – Transport Control Protocol – Used within a network

⇒ TCP/IP – TCP/ Internet Protocol – Used both within a network and between other networks in Internet.

→ TCP/IP makes data communication between any two computers anywhere in the world.

⇒ OSI Versus TCP/IP Protocols :

OSI		TCP/IP	
Application	→	HTTP	Application
Presentation		SMTP	
Session		DNS, RTP	
Transport	→	TCP, UDP	Transport →
Network	→	IP (Routing)	Network
Data Link	→	ARP, RARP	Link
Physical		cable, wire, o/f, satellite, wireless	

⇒ HTTP → Hyper Text Transfer protocol
→ used to send data on the web.
→ provides some advantages over FTP (File Transfer Protocol).

→ SMTP → Simple Mail Transfer Protocol
→ used for electronic mail.

⇒ DNS → Domain Name System
→ Translation of Host name as ram.tcp.edu.np as into an IP address as 128.135.40.0 via database lookup.

→ RTP → Real Time Protocol
→ used for real time data as on-line video, chat, etc.

→ UDP → Unreliable Datagram Protocol or User

⇒ TCP → Transport Control Protocol

(8.24)