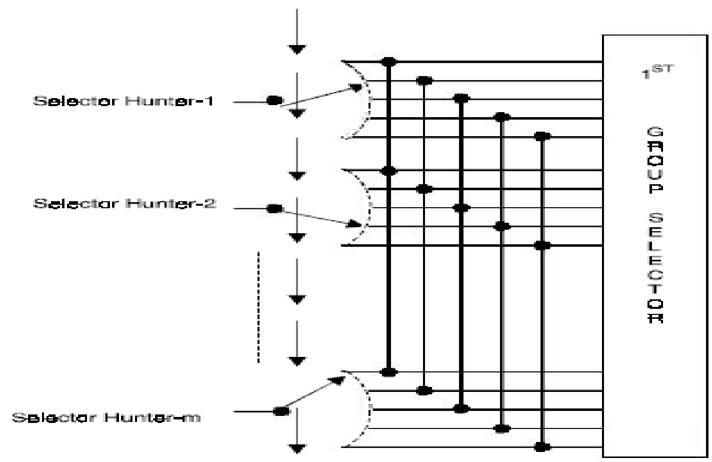
Q.2 a. What are the differences between selector hunter and line finder from the operating principle point of view with the help of proper diagram? (8)

Answer:

➢ SELECTOR HUNTER



The steps followed for connecting a subscriber to a selector hunter are as follows:

The calling subscriber lifts his handset

↓

Interrupt mechanism in his selector hunter is activated

↓

Wiper starts scanning the first selector stages until it gets a free group selector

↓

The free group selector is sensed

↓

The interrupter is disabled

↓

The corresponding first selector is marked 'Busy'

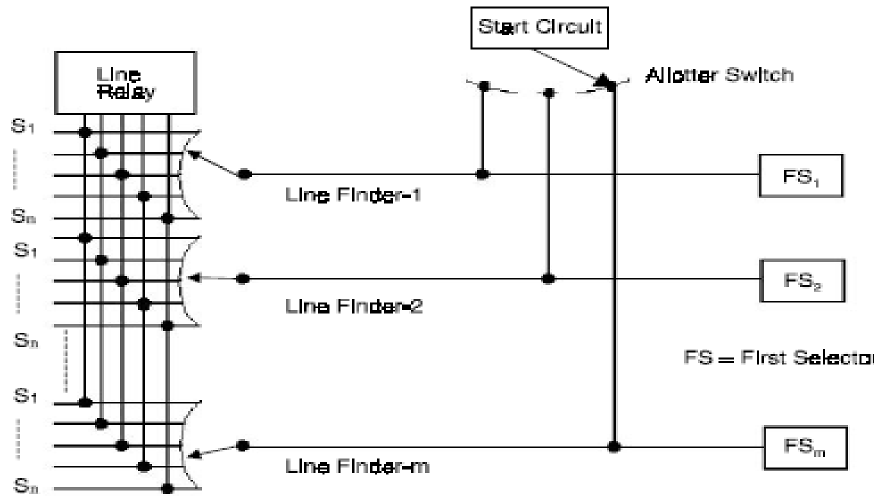
↓

The first selector sends a dial-tone to the corresponding subscriber via a selector hunter stage. Actually the selector hunter stage provides only the electrical path

↓

The first selector is now ready to receive the dialing pulses

➢ LINE FINDER



The steps involved in selecting a line finder are as follows:

Normally the wiper of the allotter switch is connected to one of the free line finders or free first selector

↓

The calling subscriber lifts his receiver

↓

The start signal is passed through the relays to line receiver

↓

Line finder commences to find free line

↓

As soon as the calling line is found, the allotter switch will jump to the next free line finder or first selector

↓

The line is now ready to receive the next calling subscriber

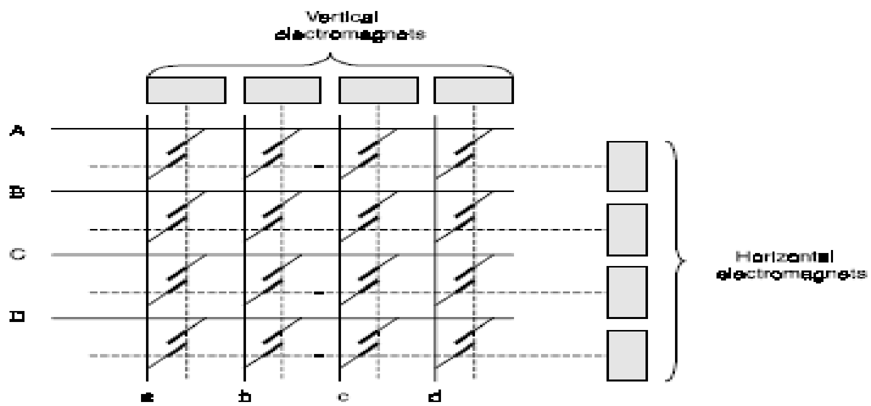
➢ DIFFERENCES BETWEEN LINE FINDER AND SELECTOR HUNTER

1. Line finder finds a free line before lifting of handset and selector hunter finds a free line after lifting of handset.
2. Line finder arrangement is more complex.
3. Delay is less in case of line finder as compared to that in selector hunter.
4. Line relays, Start circuit and Allotter switch concepts are introduced in line finder arrangement.

b. Describe the principle of operation of crossbar switching system. What is crossbarmatrix? What is cross connection problem? How can we overcome the problem of cross communication? (8)

Answer:

As shown in the Fig. , in a crossbar switch an array of vertical and horizontal wires are connected (solid lines) to the separated contact points of the switches. Horizontal and vertical bars (dotted lines) are connected to these contact points. The bars are connected to the electromagnets.



When a vertical bar (*i.e.* calling subscriber) is energized to face its corresponding bar (*i.e.* called for subscriber), an excitation of the magnet causes a rotation of the bar. This much motion of the contact unit is not sufficient to cause a complete contact. If now an electromagnet connected to the horizontal bar along with the electromagnet connected to a vertical bar energizes simultaneously, then only the contact is completed. There is an arrangement of mechanical latching which latches (holds) the contact even after one bar (say horizontal) de-energizes. If and only if the other bar de-energizes, the contact is broken up

In the Fig. . . *a*, *b*, *c* and *d* are calling subscribers and *A*, *B*, *C* and *D* are treated as called for subscriber. But actually '*a*' is not different from '*A*'. The notations are taken for the ease of understanding.

Let's assume that subscriber '*a*' is calling for subscriber '*D*'. The steps of making connection are as follows.

Lower most horizontal electromagnet energizes as it corresponds to subscriber 'a'

↓

The lowermost electromagnet bends.

↓

The bending is not sufficient to complete the contact

↓

The left most vertical electromagnet energizes as it corresponds to subscriber 'D'

↓

The leftmost electromagnet bends

↓

The contact between 'a' and 'D' subscribers made

↓

If one of these two electromagnets de-energizes, the connection does not break for the mechanical latching arrangement

↓

When both of the magnet s de-energizes, the connection is broken up

Cross connection problem

aA	bA	cA	dA
aB	bB	cB	dB
aC	bC	cC	dC
aD	bD	cD	dD

Say at time instant $t = t_1$, *a* is calling for *D* and *c* is calling for *B*. When (for the 1st case) magnet corresponding to *a* and *D* energize, *aD* connection made. But when (for the 2nd case) *c* energizes, *c*l connection will be made as *D* was being energized before. But this is not a desired connection. And *1* subscriber will be connected with *a* and *c*. This is called as cross-connection problem.

We can overcome the problem very easily by following a particular sequence for a successfı call make up. That is

Horizontal magnet energize

⇓

Vertical magnet energize

⇓

Horizontal magnet de-energize

Or we can follow another sequence as below :

Vertical magnet energize

⇓

Horizontal magnet energize

⇓

Vertical magnet de-energize

Say the 1st sequence is being followed. Taking into account the same example, if the horizontal magnet is de-energized, no chance of cross-connection is there as *c* will not find *D* corresponding magnet energized to make a contact.

Q.3 a. Compare LCC, LCR and LCH system? (4)

Answer:

In LCC system, it is assumed that unserviceable requests leave the system and never return This assumption is appropriate where traffic overflow occurs and the other routes are in other calls service. If the repeated calls not exist, LCC system is used. But in many cases, blocked calls return to the system in the form of retries. Some examples are subscriber concentrator systems, corporate tie lines and PBX trunks, calls to busy telephone numbers and access to WATS lines. Including the retried calls, the offered traffic now comprise two components *viz.*, new traffic and retry traffic. The model used for this analysis is known as lost calls returned
(LCR) model. The following assumptions are made to analyse the CLR model.

1. All blocked calls return to the system and eventually get serviced, even if multiple retries are required.
2. Time between call blocking and regeneration is random statistically independent of each other. This assumption avoid complications arrising when retries are correlated to each other and tend to cause recurring traffic peaks at a particular waiting time interval.
3. Time between call blocking and retry is somewhat longer than average holding time of a connection. If retries are immediate, congestion may occur or the network operation becomes delay system.

Consider a system with first attempt call arrival ratio of λ (say 100). If a percentage B (say 8%) of the calls blocked, B times ⌊ retries (*i.e.* 8 calls retries). Of these retries, however a percentage B will be blocked again.
Hence by infinite series, total arrival rate λ^2 is given as

$$\lambda' = \lambda + B\lambda + B^2\lambda + B^3\lambda + ...$$

$$\lambda' = \frac{\lambda}{1 - B}$$

where B is the blocking probability from a lost calls cleared (LCC) analysis.
In a lost calls held system, blocked calls are held by the system and serviced when the necessary facilities become available. The total time spend by a call is the sum of waiting time and the service time. Each arrival requires service for a continuous period of time and terminates its request independently of its being serviced or not. If number of calls blocked, a portion of it is lost until a server becomes free to service a call. An example of LCH system is the time assigned speech interpolation (TASI) system.
LCH systems generally arise in real time applications in which the sources are continuously in need of service, whether or not the facilities are available. Normally, telephone network does not operate in a lost call held manner. The LCH analysis produces a conservative design that helps account for retries and day to day variations in the busy horn calling intensities. A TASI system concentrates some number of voice sources onto a smaller number of transmission channels. A source receives service only when it is active. If a source becomes
active when all channels are busy, it is blocked and speech clipping occurs. Each speech segment starts and stops independently of whether it is served or not. Digital circuit multiplication (DCM) systems in contrast with original TASI, can delay speech for a small amount of time, when necessary to minimize the clipping.
LCH are easily analysed to determine the probability of the total number of calls in the system at any one time. The number of active calls in the system at any time is identical to the number of active sources in a system capable of carrying all traffic as it arises. Thus the distribution of the number in the system is the poisson distribution. The poisson distribution given as

$$P(x) = \frac{\mu^x}{x!} e^{-\mu}.$$

The probability that k sources requesting service are being blocked is simply the probability that $k + N$ sources are active when N is the number of servers.

b. Derive an expression to obtain the Erlang's second formula of delay system. (6)

Answer:

1. Waiting calls are selected on of first-come, first served (FCFS) or first-in-first-out (FIFO) service.
2. The second aspect of the service discipline is the length of the queue. Under heavy loads, blocking occurs. The blocking probability or delay probability in the system is based on the queue size in comparison with number of effective sources. We can model the Erlang delay system by the birth and death process with the following birth and death rates respectively

$$\lambda_k = \lambda, \; k = 0, 1, ...$$

$$\mu_k = \begin{cases} k\mu, \; k = 0, 1, ... N - 1 \\ N\mu \; k \geq S \end{cases}$$

$$P(k) = \frac{\lambda_0 \lambda_1 \ldots\ldots \lambda_{k-1}}{\mu_1 \mu_2 \ldots\ldots \mu_k} P(0) \; k = 1, 2, \ldots\ldots$$

we set
$$P(k) = \begin{cases} \dfrac{1}{k!}\left(\dfrac{\lambda}{\mu}\right)^k P(0) & 0 \le k \le S \\ \dfrac{\left(\dfrac{\lambda}{\mu}\right)^k}{N!\,N^{k-N}} P(0) & k \ge N. \end{cases}$$

As $A = \dfrac{\lambda}{\mu}$, we get
$$P(k) = \begin{cases} \dfrac{A^k}{k!} P(0) & 0 \le k \le S \\ \dfrac{A^k}{N!\,N^{k-N}} P(0); & k > N. \end{cases}$$

Under normalised condition,

$$\sum_{k=0}^{\infty} P(k) = 1 \qquad \text{or} \qquad \sum_{k=0}^{N-1} \frac{A^k}{k!} P(0) + \sum_{k=N}^{\infty} \frac{A^k}{N!\,N^{k-N}} P(0) = 1$$

$$\frac{1}{P(0)} = \sum_{k=0}^{N-1} \frac{A^k}{k!} + \frac{N^N}{N!} \sum_{k=N}^{\infty} \left(\frac{A}{N}\right)^k$$

$$= \sum_{k=0}^{N-1} \frac{A^k}{k!} + \frac{N^N}{N!}\left[\left(\frac{A}{N}\right)^N + \left(\frac{A}{N}\right)^{N+1} + \left(\frac{A}{N}\right)^{N+2} + \ldots\ldots\right]$$

$$= \sum_{k=0}^{N-1} \frac{A^k}{k!} + \frac{N^N}{N!}\left(\frac{A}{N}\right)^N \left[1 + \frac{A}{N} + \left(\frac{A}{N}\right)^2 + \ldots\ldots\right]$$

$$= \sum_{k=0}^{N-1} \frac{A^k}{k!} + \frac{A^N}{N!}\left[\frac{1}{1 - A/N}\right] = \sum_{k=0}^{N-1} \frac{A^k}{k!} + \left[\frac{A^N}{N!} + \frac{A^N}{N!}\left(\frac{A}{N-A}\right)\right]$$

$$\frac{1}{P(0)} = \sum_{k=0}^{N} \frac{A^k}{k!} + \frac{A^N}{N!}\left(\frac{A}{N-A}\right)$$

$$P(0) = \frac{1}{\sum_{k=0}^{N} \frac{A^k}{k!} + \frac{A^N}{N!}\left(\frac{A}{N-A}\right)}$$

$$P(k) = \frac{A^k}{k!} P(0), \; k = 1, 2, \ldots\ldots, N$$

$$C(N, A) = \frac{A^N / N!}{\sum_{k=0}^{N} \frac{A^k}{k!} + \frac{A^N}{N!}\left(\frac{A}{N-A}\right)}$$

$$\frac{1}{C(N, A)} = \frac{\sum_{k=0}^{N} \frac{A^k}{k!}}{\frac{A^N}{N!}} + \frac{\frac{A^N}{N!}\left(\frac{A}{N-A}\right)}{\frac{A^N}{N!}!}$$

$$\frac{1}{C(N, A)} = \frac{1}{B} + \frac{A}{N-A}.$$

$$\text{Prob. (delay)} = P(>0)\ C(N, A) = \frac{BN}{N - A(1-B)}$$

where B = Blocking probability for a LCC system

N = Number of servers

A = Offered load (Erlangs)

c. A PBX has 4 operators and receives 300 calls during a busy hour. The average holding time is 36 seconds. Assume that call arrivals are poisonian and service time is negative exponential distribution. Calculate (i) the percentage of calls on queue (ii) average delay (iii) percentage of calls delayed for more than 45 seconds, 30 seconds and 20 sec. (6)

Answer:

Given data : *N = 3, n = 300 calls, h = 36 sec.*

(i)

$$A = \frac{nh}{T} = \frac{300 \times 36}{3600} = 3 \text{ Erlangs.}$$

$$P(0) = 0.038$$

$$B = 0.0285$$

$$C(4, 3) = 0.105$$

i.e. 10.5% of calls have delay on answer.

(ii)

Average delay of a call

$$W(t)_{avg.} = \frac{C(N, A)h}{N - A} = 3.78 \text{ seconds.}$$

(iii)

Percentage of calls delayed for more than 45 seconds

$$P(t \geq 45) = C(N, A)\, e^{-(N-A)t/h} = 0.03$$

that is 3% calls delayed for more than 45 sec

$$P(t \geq 30) = 0.045$$

that is 4.5% calls delayed for more than 30 sec

$$P(t \geq 20) = 0.06$$

that is 6% calls delayed for more than 20 sec.

Q4 a. Using Lee graph, obtain the expression of blocking probability of a three stage switching network. (6)

Answer:

Lee graphics. C.Y. Lee's approach of determing the blocking probabilities of various switching system is based on the use of utilization percentage or loadings of individual links.

Let p be the probability that a link is busy. The probability that a link is idle is denoted by $q = 1 - p$. When any one of n parallel links can be used to complete a connection, the blocking probability B is the probability that all links are busy is given by

$$B = p^n$$

when a series of n links are all needed to complete a connection,

$$B = 1 - q^n$$

$$B = (1 - q'^2)^k$$

where q' = probability that an interstage link is idle = $1 - p'$

p' = probability that any particular intersatge link is busy

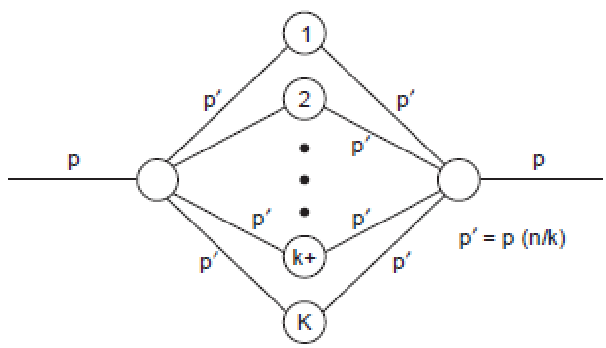
k = number of centre stage arrays.

If p is known, the probability that an interstage link is busy is given by

$$p' = \frac{p}{\beta}$$

where $$\beta = k/n$$

β is the factor by which the percentage of interstage links that are busy is reduced.



$$B = \left[1 - \left[1 - \frac{p}{\beta}\right]^2\right]^k$$

b. Compare single stage networks and multistage networks. (6)

Answer:

SINGLE STAGE	MULTI STAGE
Inlet to outlet connection is through a single crosspoint	Inlet to outlet connection is through a multiple crosspoint
Use of single crosspoint per connection result in better quality link	Use of multiple crosspoints may degrade the quality of a connection
Each individual crosspoint can be used for only	Same crosspoint can be used to establish

one inlet/outlet pair connection	connection between a number of inlets/outlets pairs
A specific crosspoint is needed for each specific connection	A specific connection may be established by using different sets of crosspoints
If a crosspoint fails, associated connection cannot be established. Ther is no redundancy	Alternative cross-points and paths are available
Number of crosspoints is prohibitive.	Number of crosspoints is reduced significantly
The network is non blocking in character	The network is blocking in character
Time for establishing a call is less	Time for establishing a call is more

c. A three stage switching structure is to accommodate N = 128 input and 128 output terminals. For 16 first stage and 16 last stage, determine the number of cross points for nonblocking. (4)

Answer:

The number of matrices at first and last stage is given by $\alpha = \frac{N}{n}$.

Hence $n = \frac{N}{\alpha} = \frac{128}{16} = 8$

To avoid blocking $k = 2n - 1 = 2 \times 8 - 1 = 15.$

Number of crosspoints is calculated by

$$N_x = k\left[2N + \left(\frac{N}{n}\right)^2\right] = 15\left[2 \times 128 + \left(\frac{128}{8}\right)^2\right]$$

$$N_x = 7680 \text{ cross points.}$$

Q.5 a. Explain basic Time division space switching with a diagram of switching structure. (5)

Answer: Refer section 6.1 of Text Book-II

b. Compare TST and STS networks. (5)

Answer:

STS	TST
Input space block is interlaced to output space block using a time block in between	Input time block interlaced to output time block by using a space block in between
Not preferred now since size of 's' block being a matrix increases as square of inputs	Preferred now as size of time switch increases linearity with number of input and output buses.
Now costly due to cost of switching hardware because two space switch blocks are required	Now economical due to availability of low cost higj speed memories required for T-blocks
For large size, size can be limited by splitting S blocks S-S-T-S-S	Small size can be further reduced by T-S-S-S-T
Peripheral functions cannot be incorporated into S block	Peripheral functions such as super MUX alignment of PCM with exchange frame can be incorporated into T-S block
Design is very simple	Design is complicated
Memory size is large	Memory size is small

c. In an STS switch, blocking probability is 0.002 and loading is 0.2 erlang per channel. How many time-slot interchange (TSI) are needed? What is the cost of switch? Given M_1=4128 primary TDM signals and 30 voice channels per input. (6)

Answer:

P_B=0.002, ρ=0.2 E, M_1=4, N=128, M=30

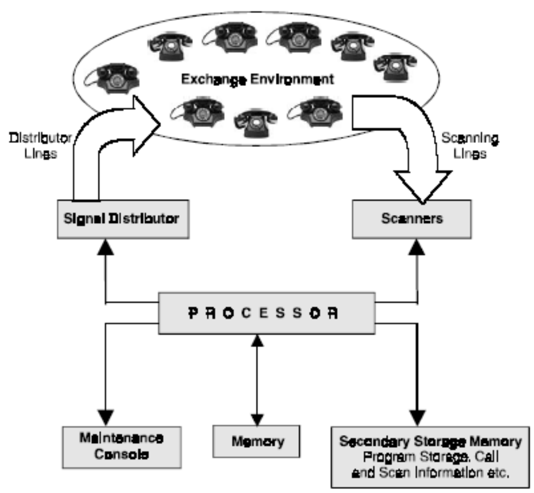
Blocking probability is given by $P_B = [1-(1-\frac{\rho}{k})^2]^{M_1}$

$0.02 = [1-(1-\frac{0.2}{k})^2]^4$. Hence, k=41.

Cost of STS switch is given by C_{STS}=2Nk+4MN=(2x128x41)+(4x30x128)=26880 units.

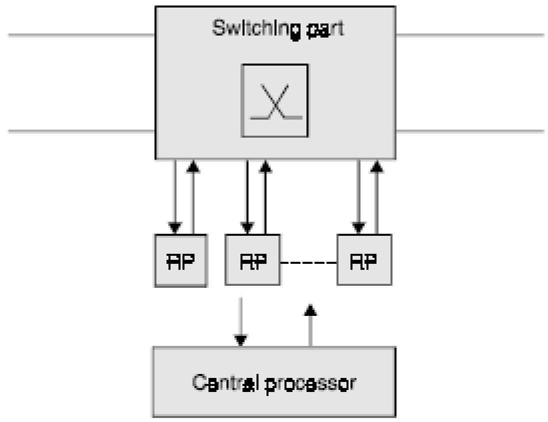
Q.6 a. With the help of block diagram explain the working of centralized stored program control. (8)

Answer:



The above figure shows the general organization of a centralized SPC. Today, digital technology is used both in the control part (processors) and in the switching part (in the form of specialized hardware). There is a distinct division between the processor part and the switching part - an arrangement called centralized control. However, a modern alternative is distributed control: by using microprocessors, processor capacity can be located close to the object to be controlled. Processor capacity can also be located far away from the equipment, provided the control logic is centralized.

In the most refined form, only one processor is used (a single-processor system) to perform both routine work and advanced operations. The processor must be dimensioned according to the most difficult tasks. At the same time, however, because the routine tasks are the most time-consuming, the processor may have difficulty getting all things done. One solution to this kind of problem is to let several processors share the work load (multiprocessor system). Another model for sharing work is based on a hierarchically designed processor structure, in which the routine-like work is handled by several regional processors (RPs) and their coordination and more complex tasks are handled by a central processor.



-

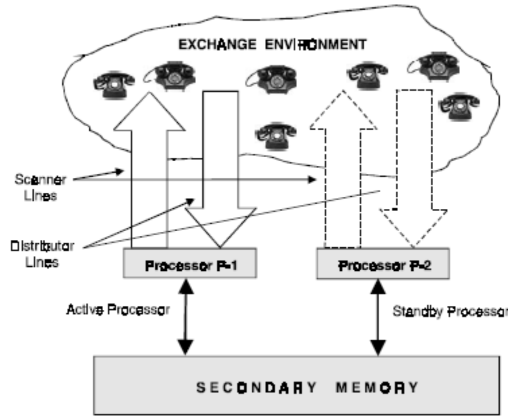
Here, too, we distinguish between single- and multiprocessor systems. In the hierarchically designed multiprocessor system, the central processor consists of several processor units working in parallel. A hierarchical single-processor system has only a central processor.

Centralized SPC using two processor configuration is generally classified into three categories on the basis of their mode of operation. These are as follows:

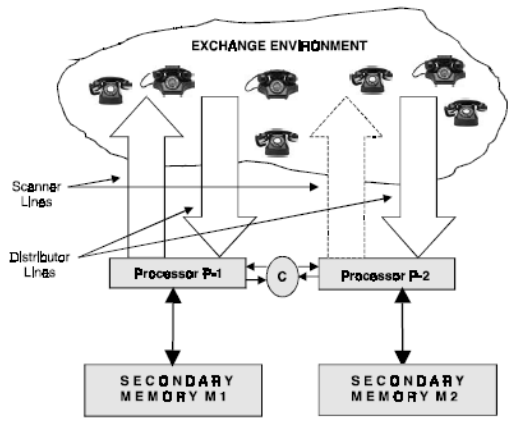
1. Standby Mode
2. Synchronous Duplex Mode
3. Load Sharing Mode

1. Standby Mode

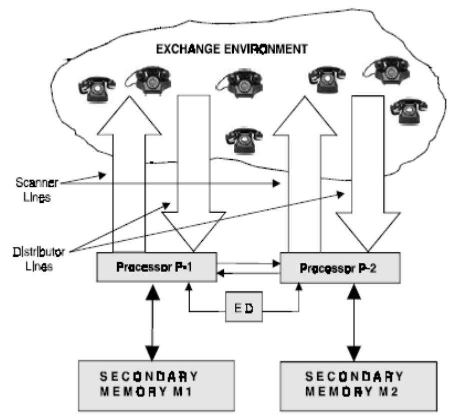
The entire exchange environment is scanned through the scanner by the processor P1 and the signals are distributed accordingly through the distributor. The scan result is stored in the secondary memory. For large exchanges it is difficult to store the scan results at every instant as the data is very bulky. So, the results are stored after a fixed amount of time. The processor P2 is the stand-by processor and remains idle as long as P1 function properly. If somehow processor P1 fails, the processor P2 becomes active. The data of the scanned results are retrieved by P2 from the secondary memory. The calls which were being processed in between the last saved result and the failing of P1, are lost.



2. Synchronous Duplex Mode



3. Load Sharing Mode



Here the entire exchange environment is scanned synchronously through the individual scanners of the processors P1 and P2. Distribution of signals is done only by P1. Both P1 and P2 have independent dedicated memories. C is a comparator which checks the state of health of both the processors. Here the comparator C actually functions according to XOR decision making. We all know that from the XOR truth table that if two different states occur, output is '1'. In errorless operation, P1 and P2 both will give the same output. When error occurs, C comparator indicates the error and asks for a *check out program*. If it finds P1 as faulty, it gives the entire control of the exchange to P2. M1 and M2 have check-out programs in them which are run separately to detect the faulty processor.

However, if both P1 & P2 fail, the comparator can't detect the fault. Also, transient errors can't be detected by check-out programs. In that case one may ignore the transient error and continue to work with the same processor. It may also be possible to run the exchange by the other processor leaving the faulty one.

b. A central processor system contains 2 identical units each of which can carry the full load. The mean time to failure (MTTF) of each unit is 1000 hrs. It can be assumed that failures of the units are independent random event. Estimate the MTTF of the system if the mean time to repair (MTTR) for a unit is
(i) 10 hrs (ii) 1 Hrs. (8)

Answer:

Let the units be A and B then $MTTF_A = MTTF_B = 1000$ hrs

i) $MTTR_A = MTTR_B = 10$ hrs

Thus $(10/(10+x)) = (10/(1000+10)) x (10/(1000+10))$

Hence x=11.64 years

ii) $(1/(1+x)) = (1/(1000+1)) x (1/(1000+1))$

Hence, x=114.38 years.

Q.7 a. (i) What advantage does common channel signalling have over channel associated signalling?

(ii) With the help of block diagram, explain out of band signalling. (8)

Answer:

i)

In associated CCS signalling mode, there is a direct link between two exchanges. In this mode, the signalling path passes through the same set of switches as does the speech path. Network topologies of the signalling network and the speech network are the same. This mode of operation is simple, economic and easy to control. This involves in delayed operation for long distance communication.

In non-associated CCS signalling, there are separate control of the networks from the switching machines themselves. In multiexchange network, signal message passing through several intermediate nodes is referred as non-associated signalling. The network topologies for the signalling and the speech networks are different. Between exchanges, many STP’s are placed. This approach is flexible as far as the routing is concerned. It demands more comprehensive scheme for message addressing than is needed for channel associated signalling.

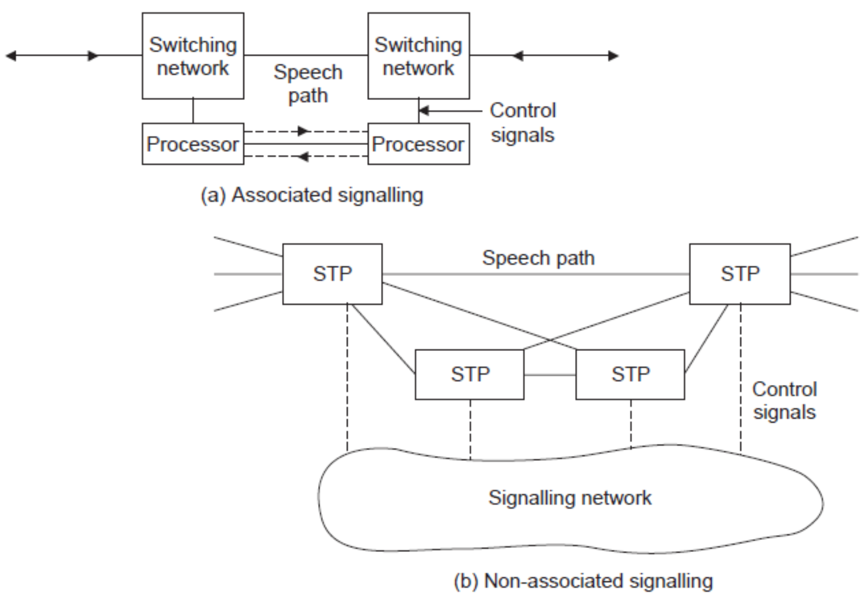


Fig. . Modes of CCS signalling.

In practice, CCS messages are routed through one intermediate node for short distance communication. This is known as quasi-associated signalling. It establishes simplified predetermined paths between exchanges. The signalling paths are not associated but are fixed for given speech connections.

(ii) Refer article 9.9 page 382 of Text Book-II

Outband signalling

This signalling has frequenies above the voice band but below the upper limit of 4 kHz.The CCITT recommended frequency for outband signalling is 3825 Hz, but 3700 Hz and 3850 Hz are also used. The general layout of an outband signalling is shown in Fig. .

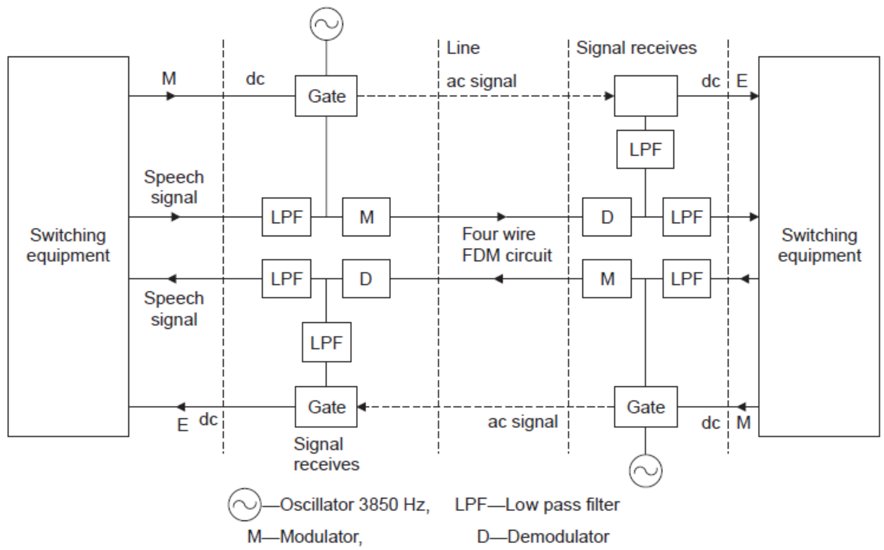


Fig. General layout of an outband signalling.

The system shown above uses 4 wire E and M trunk (two wire voice path, an E-lead, and an M-lead with earth ground returns). Other types of E and M trunk is defined with 8 wires 4 wire voice path, an E-lead, with an associated return lead (SGD), and an M lead with an associated return (SB). In any type of E and M interface, supervision signalling is always conveyed on the E and M leads and not on the the voice pair. The E lead always carries signal from the signalling apparatus to the switching equipment and the M lead carries signals from
the switching equipment to the signalling aparatus.

Advantages:

1. The requirement of line splits are not necessary to avoid sigal limitation.
2. Signals and speech can be transmitted simultaneously without disturbing the conversation.
3. Simple and consequently cheap.

Disadvantages:

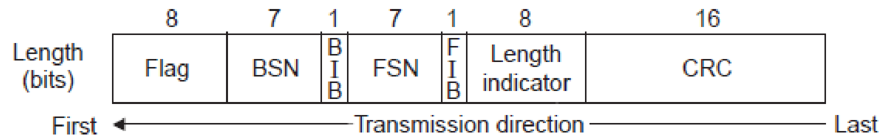
1. Very narrow bandwidth is available for signalling.
2. Filtering circuits are needed to handle the signalling bands.
3. More dependent on the transmission system.

b. Name three types of signalling units used in SS7. With neat diagrams explain each fields associated with the signalling units. (8)

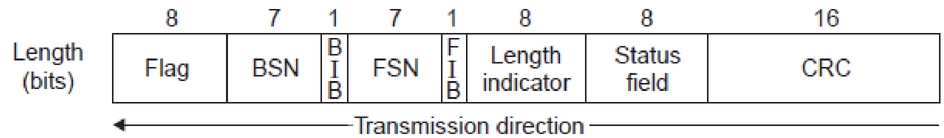
Answer: Refer article 9.2 page 384 of Text Book-II

There are three types of signalling units (SU) defined in SS7. They are message signal unit (MSU), Link status signal unit (LSSU) and Fill in signal unit (FISU). SUs of each type follow a format unique to that type. A high level view of those formats is shown in Fig. All three SU types have a set of common fields that are used by MTP level 2. The SU is based on the high level data link control (HDLC) protocol (described in chapter 11).

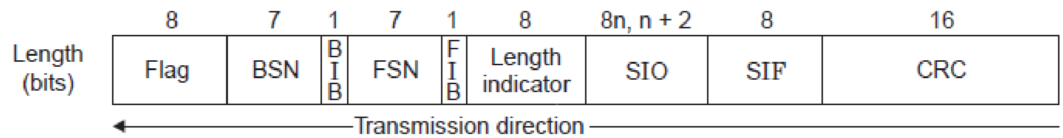
The MSU transfers information supplied by a user part (level 4) via the signalling network level (level 3). The LSSU is used for link initialization and flow control. The FISU is sent to maintain alignment when there is no signal traffic.



(a) Fill-insignal unit (FISU)



(b) Line status signal unit (LSSU)



(c) Message signal unit (MSU)

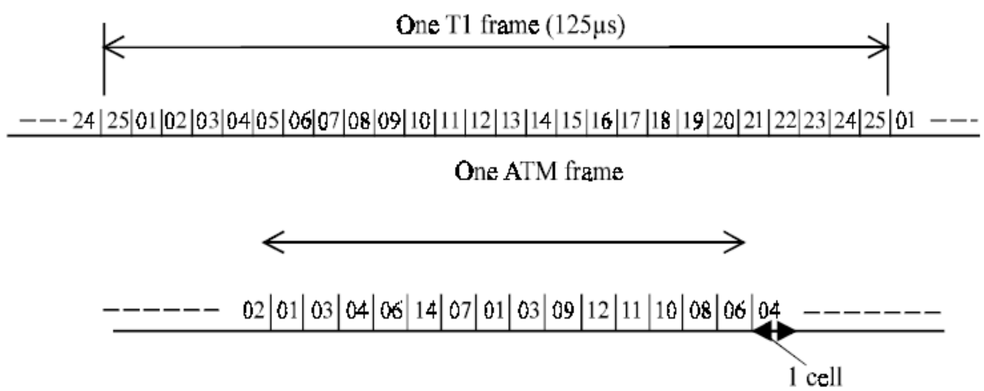
Q.8 a. What is ATM? How ISDN data is transmitted through ATM network? (8)

Answer:

Transmission in ATM Network

Asynchronous transfer mode can be contrasted with the synchronous T1 system. One T1 frame is generated precisely every 125 μs. This rate is generated by a master clock. Slot *k* of each frame contains 1 byte of data from the same source and arranged alternately one after another. Here channel one (1) gets exactly one byte at the start of each frame.

In ATM, in contrary, has no requirement that cells rigidly alternate among the various sources. Cells arrive randomly from different sources *i.e.,* it does not follow a particular pattern. No requirement of cell ordering is there.



ATM link must be situated between two switches or one switch and one computer.

The ATM *Physical Medium Dependent (PMD)* sub-layer is concerned with getting the bits IN and OUT the wire. Different hardwares are needed for different cables and fibers, depending on the speed and line coding technique. The purpose of *transmission convergence (TC)* sub-layer is to provide a uniform interface to the ATM layer in both directions. Outbound, ATM layer provides a sequence of cells, and the PMD sub-layer encodes according to necessity and pushes them out of the door as a bit stream.

Inbound the PMD sub-layer takes the incoming bits from the network and delivers a bit stream to the TC sub-layer. It is upto TC sub-layer to somehow figure out how to tell the ending and beginning on one cell. This job is theoretically impossible. Thus TC sub-layer has its work cut out for it.

b. Describe the operation of star, bus, ring and hybrid network topology. (8)

Answer:

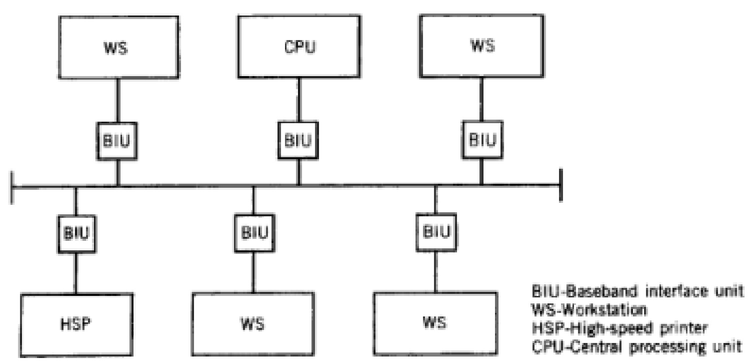


Figure 1A. A bus network.

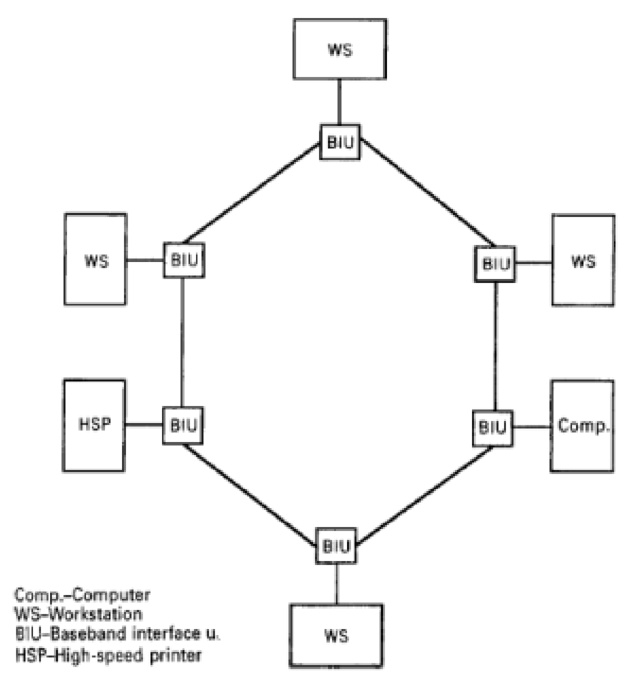


Figure .1B. A typical ring network.

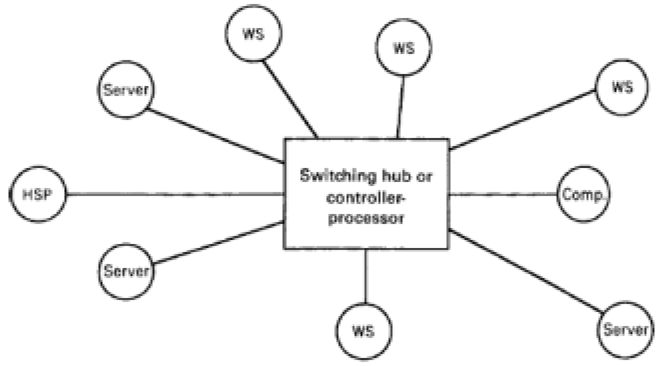


Figure .1C. A star network.

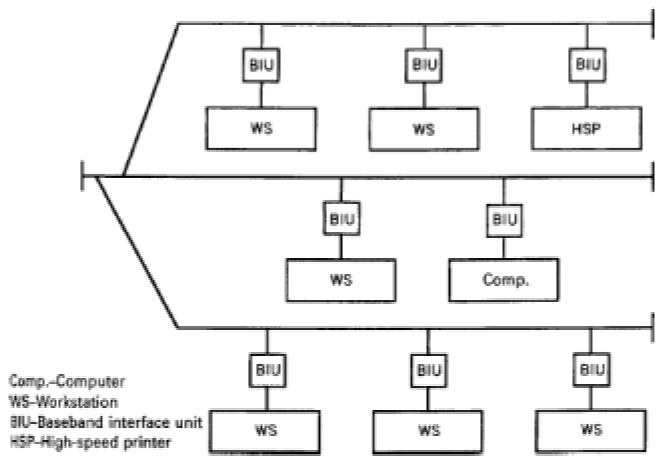


Figure 1D. A typical tree configuration.

Q.9 a. Tabulate the PSTN numbering format followed in India. (8)

Answer:

Trunk code (SDCA code)	+	Telephone exchange code	+	Last n digits of subscriber number
ABCD	+	EF	+	PQRS
ABCD	+	EFG	+	PQR
ABC	+	EF	+	PQRST
ABC	+	EFG	+	PQRS
AB	+	EFG	+	PQRST
AB	+	EFGH	+	PQRS

The trunk code is 2 to 4 digits. The telephone exchange code and last *n* digits of subscriber number together called subscriber number and is from 6 t0 8 digits. Hence national number to call a subscriber is 8 to 12 digits. To call a subscriber in another SDCA, prefix ‘0’ must be dialled first

b. We consider a cellular system in which total available voice channels to handle the traffic are 960. The area of each cell is 6 km^2 and the total coverage area of the system is 2000 km^2. Calculate (i) the system capacity if the cluster size N (reuse factor) is 4 and (ii) the system capacity if the cluster size is 7. How many times would a cluster of size 4 have to be replicated to cover the entire cellular area? Does decreasing the reuse factor N increase the system capacity? Explain. (8)

Answer:

Total available channels=960
Cell area=6km^2
Total coverage area=2000km^2
N=4
Area of cluster with reuse N=4:4x6=24km^2
Number of clusters for covering total area with N equals to 4 is 2000/24=83.33~83.
Number of channels per cell=960/4=240.
System capacity=83x960=79680 channels
N=7
Area of cluster with reuse N=7:7x6=42km^2
Number of clusters for covering total area with N equals to 7 is 2000/4=47.62~48.
Number of channels per cell=960/7=137.15~137.
System capacity=48x960=46080 channels

TEXT BOOKS

I. Telecommunications Switching, Traffic and Networks, J.E.Flood, Pearson Education-2006
II. Telecommunication Switching Systems and Networks, Thiagarajan Viswanathan, Prentice Hall of India Pvt. Ltd, 2007