Lecture Notes Faculty: S. Agrawal

Subject: Communication Networks and Switching (CNS)

2nd Semester, M.Tech (Communication System Engineering)

Module-I: (10 Hours)

Overview of switching systems, Electronic switching and stored program control systems, Centralized SPC, Availability, Distributed SPC, Enhanced services, Digital switching: time switching, space switching, time and space switches, Switching techniques: Circuit Switching, Message and Packet Switching.

Module-II: (10 Hours)

Computer controlled switching systems: Introduction, Call processing, signal exchange diagram, state transition diagram, hardware configuration, switching system software organization, software classification and interfacing, Maintenance software, call processing software, Administration software, Electronic Exchanges in India.

Module-III: (10 Hours)

Traffic engineering: Traffic pattern, Grade of Service and blocking probability, modeling of switching systems: Markov Process, Birth-Death Process.

Telephone network organization: Network management, Network services, various networking plans, types of networks, Routing plan, International numbering plan, National numbering plan, Numbering plan in India, Signaling: in channel signaling, common channel signaling.

Module-IV: (10 Hours)

Overview of ISDN, VPN, VOIP, IP switching

Text books:

1. Telecommunication Switching Systems and Networks, by Thiagarajan Viswanathan, PHI.
2. Telecommunication Systems Engineering, R. L. Freeman, 4/e, Wiley publication, 2010

Reference book:

1. Telecommunication Switching and Networks. By P.Gnanasivam, New Age International.

ACKNOWLEDGMENT

Different sources used in the preparation of this material are:

1. Telecommunication Switching Systems and Networks, by Thiagarajan Viswanathan, PHI.
2. Telecommunication Systems Engineering, R. L. Freeman, 4/e, Wiley publication, 2010.
3. Telecommunication Switching and Networks. By P.Gnanasivam, New Age International.
4. Internet Sources for Module IV.

Module-I: OVERVIEW OF SWITCHING SYSTEMS

Electronic switching and stored program control systems, Centralized SPC, Availability, Distributed SPC, Enhanced services, Digital switching: time switching, space switching, time and space switches, Switching techniques: Circuit Switching, Message and Packet Switching

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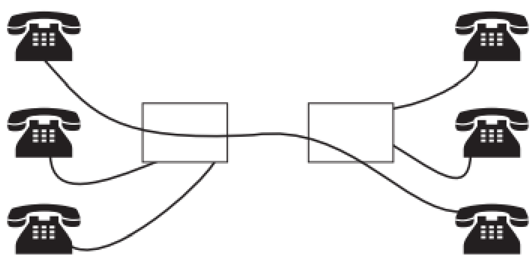
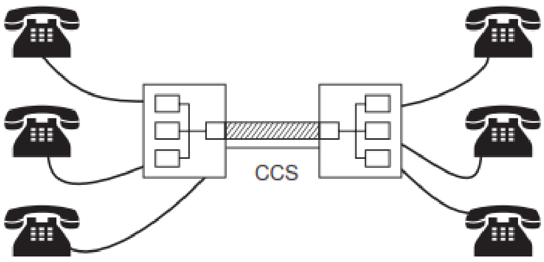
Telecommunication networks carry information signals among entities, which are geographically far apart. An entity may be a computer or human being, a facsimile machine, a teleprinter, a data terminal and so on. The entities are involved in the process of information transfer which may be in the form of a telephone conversation (telephony) or a file transfer between two computers or message transfer between two terminals etc. Today it is almost true to state that telecommunication systems are the symbol of our information age. With the rapidly growing traffic and untargeted growth of cyberspace, telecommunication becomes a fabric of our life. The future challenges are enormous as we anticipate rapid growth items of new services and number of users. What comes with the challenge is a genuine need for more advanced methodology supporting analysis and design of telecommunication architectures. Telecommunication has evaluated and growth at an explosive rate in recent years and will undoubtedly continue to do so. The communication switching system enables the universal connectivity. The universal connectivity is realized when any entity in one part of the world can communicate with any other entity in another part of the world. In many ways telecommunication will acts as a substitute for the increasingly expensive physical transportation. The telecommunication links and switching were mainly designed for voice communication. With the appropriate attachments/equipment, they can be used to transmit data. A modern society, therefore needs new facilities including very high bandwidth switched data networks, and large communication satellites with small, cheap earth antennas.

The use of computers to control the switching led to the designation ‘‘electronic’’ switching system (ESS) or Electronic automatic exchange (EAX). In 1970, first electronic switching system No. 1 ESS or No. 1 EAX was introduced. Digital electronic switching matrices were first introduced into the U.S. Public network in 1976 with AT & T’s No. 4 ESS digital toll switch. By the mid 1980’s the interoffice transmission environment has changed to almost exclusively digital. Fig. 1.1 shows the various telephone networks.



(a) Telephone network around 1890



(b) Telephone network around 1988

Fig. 1.1. Various telephone networks

Telecommunication is mainly concerned with the transmission of messages between two distant points. The signal that contains the messages is usually converted into electrical waves before transmission. Our voice is an analog signal which has amplitude and frequency characteristic.

Voice frequencies. The range of frequencies used by a communication device determines the communication channel, communicating devices, bandwidth or information carrying capacity. The most commonly used parameter that characterizes an electrical signal is its bandwidth of analog signal or bit rate if it is a digital signal. In telephone system, the frequencies it passes are restricted to between 300 to 3400 Hz. Thus the network bandwidth is 3100 Hz. The bandwidth and bit rate for various types of system are shown in Table 1.1.

Table 1.1. Bandwidth requirements of various applications

Type	*Bandwidth*	*Bit Rate*
Telephone (speech)	300—3400 Hz	—
Music	50 Hz—16 kHz	—
Facsimile	40 kHz	—
Broadcast televison	0—55 MHz	—
Personal communication	—	300 to 9600 bits/sec
E—Mail transmission	—	2400 to 9600 bits/sec
Digitized voice phone call	—	6400 bits/sec
Digital audio	—	1 to 2 M bits/sec
Compressed video	—	2 to 10 M bits/sec
Document imaging	—	10 to 100 M bits/sec
Full motion video	—	1 to 2 G bits/sec

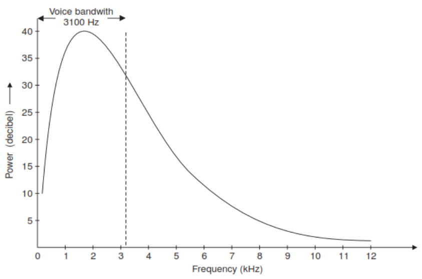


Fig. 1.2. Speech Spectrum.

Speech spectrum. The telephone channel over which we wish to send data are designed to transmit electrical oscillations (microphone converts sound into equivalent number of electrical oscillation) of voice. Fig. 1.2 is described as a speech spectrum diagram. It illustrates human speech strength variations at various frequencies. Most of the energy is concentrated between 300 Hz to 3400 Hz.

ELEMENTS OF COMMUNICATION SWITCHING SYSTEM

The purpose of a telecommunication switching system is to provide the means to pass information from any terminal device to any other terminal device selected by the originator.

Telecommunication system can be divided into four main parts. They are

1. End system or Instruments
2. Transmission system
3. Switching system
4. Signaling.

CRITERIA FOR THE DESIGN OF TELECOMMUNICATION SYSTEM

Traditionally, the design for telephone switching centre or equipment requirement in a telecommunication system are determined on the basis of the traffic intensity of the busy hour. The traffic intensity is defined as the product of the calling rate and the average holding time. The busy hour is defined as that continuous sixty-minute period during which the traffic intensity is highest. The calling rate is the average number of request for connection that are made per unit time. If the instant in time that a call request arises is a random variable, the calling rate maybe stated as the probability that a call request will occur in a certain short interval of time. The holding time is the mean time that calls last. Otherwise the average holding time is the average duration of occupancy of traffic path by a call.

Grade of Service. In telephone field, the so called busy hour traffic are used for planning purposes. Once the statistical properties of the traffic are known, the objective for the performance of a switching system should be stated. This is done by specifying a grade of service (GOS). GOS is a measure of congestion expressed as the probability that a call will be blocked or delayed. Thus when dealing with GOS in traffic engineering, the clear understanding of blocking criteria, delay criteria and congestion are essential.

Blocking criteria. If the design of a system is based on the fraction of calls blocked (the blocking probability), then the system is said to be engineered on a blocking basis or call loss basis. Blocking can occur if all devices are occupied when a demand of service is initiated. Blocking criteria are often used for the dimensioning of switching networks and interoffice trunk groups. For a system designed on a loss basis, a suitable GOS is the percentage of calls which are lost because no equipment is available at the instant of call request.

Delay criteria. If the design of a system is based on the fraction of calls delayed longer than a specified length of time (the delay probability), the system is said to be a waiting system or engineered on a delay basis. Delay criteria are used in telephone systems for the dimensioning of registers. In waiting system, a GOS objective could be either the percentage of calls which are delayed or the percentage which are delayed more than a certain length of time.

Congestion. It is the condition in a switching centre when a subscriber cannot obtain a connection to the wanted subscriber immediately. In a circuit switching system, there will be a period of congestion during which no new calls can be accepted. There are two ways of specifying congestion.

1. **Time congestion.** It is the probability that all servers are busy. It is also called the probability of blocking.

2. **Call congestion.** It is the proportion of calls arising that do not find a free server. Call congestion is a loss system and also known as the probability of loss while in a delay system it is referred to as the probability of waiting. If the number of sources is equal to the number of servers, the time congestion is finite, but the call congestion is zero. When the number of sources is large in comparison with servers, the probability of a new call arising is independent of the number already in progress and therefore, the call congestion is equal to the time congestion. In general, time and call congestions are different but in most practical cases, the discrepancies are small.

Measure of GOS. GOS is expressed as a probability. The GOS of 2% (0.02) mean that98% of the calls will reach a called instrument if it is free. Generally, GOS is quoted as P.02 or simply P02 to represent a network busy probability of 0.02. GOS is applied to a terminal-to terminal connection. For the system connection many switching centres, the system is generally broken into following components.(*i*) an internal call (calling subscriber to switching office)(*ii*) an outgoing call to the trunk network (switching office to trunk)(*iii*) The trunk network (trunk to trunk)(*iv*) A terminating call (switching office to called subscriber)The GOS of each component is called component GOS. The GOS for internal calls is 3 to 5%, for trunk calls 1-3%, for outgoing calls 2% and for terminating calls 2%. The overall GOS of a system is approximately the sum of the component grade of service. In practice, in order to ensure that the GOS does not deteriorate disastrously if the actual busy hour traffic exceeds the mean, GOS are specified 10% or 20% more of the mean.

A telephone network is composed of a variety of all processing equipment, interstate switching links and inters office trunks. Because of the random nature of the call request, the design of equipment switching links and trunks are quite difficult. Thus, the traffic analysis is the fundamental request for the design of cost effective, efficient and effective configuration of networks. The effectiveness of a network can be evaluated in terms of how much traffic it carries under normal or average loads and how often the traffic volume exceeds the capacity of the network. Fundamental problem in the design of telecommunication networks concerns the dimensioning of a route. To dimension the route, volume of traffic required grade of service and capacity (in bits per sec) must be known.

Traffic. In telecommunication system, traffic is defined as the occupancy of the server in the network. There are two types of traffic viz. voice traffic and data traffic. For voice traffic, the calling rate is defined as the number of calls per traffic path during the busy hour. In a day, the 60 minutes interval in which the traffic is highest is called busy hour (BH).

Average occupancy. If the average number of calls to and from a terminal during a period T second is ‘*n*’ and the average holding time is ‘*h*’ seconds, the average occupancy of the terminal is given by

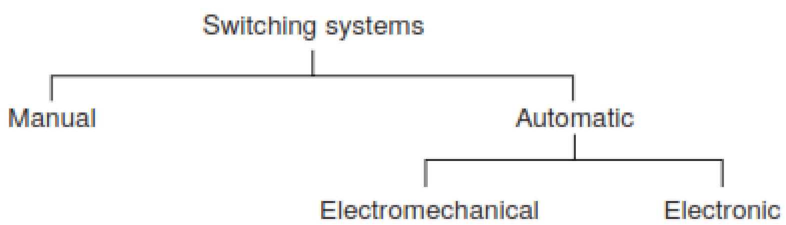
$$\mathrm{A} = \frac{nh}{\mathrm{T}}$$

The average occupancy is also referred as traffic flow of traffic intensity. The international unit of telephone traffic is the Erlang.

Telecommunication system is an important and integral part of modern society. In addition to public switched telephone network (PSTN), it plays vital role in radio and television networks, internet and Asynchronous transfer mode (ATM) networks. The switching system provides various services to the subscribers. The switching system is a collection of switching elements arranged and controlled in such a way as to setup a communication path between any two distant points. This chapter demonstrates the switching systems of manual exchanges to the electronic switching systems. The process of transferring message from one place to another (or line to line) is called switching related to outside the switching plant or systems. There are three types of switching namely a circuit switching, message switching and packet switching. In telecommunication switching, the circuit switching and message switching are used. The switching technique used in computer communication network or data transfer is packet switching. Telecommunication is the communication of voice or data over long distances using public switched telephone network (PSTN). PSTN consists of transmission component, switching components and facilities for maintaining equipment, billing system and other internal components. PSTN also referred to as plain old telephone system (POTS). The switching technique used in PSTN is circuit switching in general. To setup connection between subscribers, the PSTN consists of the transmission systems, switching system and signalling systems.

Classification of Switching System

In early days, the human exchange provided switching facilities. In manual exchanges, a human operator and the elements like switches, plugs and sacks were used to connect two subscribers. Around 1890's many electromechanical switching devices were introduced. Till 1940, different electromechanical switching system were invented, of which strowger switching system and cross bar switching system were still popular. The later invention of electronic switching system (ESS) which uses stored program control (SPC) and computer controlled switching systems are presently dominating the worldwide exchanges. Fig. shows the classification of switching system.



The electronic switching system (ESS) uses stored program control. The further classification of ESS are space division switching and time division switching. The time division switching is divided into digital and analog switching systems. The digital switching system is classified into space switch, time switch and combination switch.

Requirements of Switching System

All practical switching system should satisfy the following requirements for the economic use of the equipment of the system and to provide efficient service to the subscribers. Depends on the place (Rural or town, big town, city or big cities). The local exchange located, the service provided to the subscriber may vary. Some important requirements are discussed briefly.

High availability. The telephone system must be very reliable. System reliability can be expressed mathematically as the ratio of uptime to sum of the uptime and down time. The uptime is the total time that the system is operating satisfactorily and the down time is the total time that is not. In telephone switching networks, the availability or full accessibility is possible if all of the lines are equally accessible to all incoming calls. The full accessibility is also defined as the capacity or number of outlets of a switch to access a given route. If each incoming trunk has access to a sufficient number of trunks on each route to give the required grade of service is known as limited availability. The availability is defined as

$$A = \frac{\text{Uptime}}{\text{Uptime} + \text{down time}} \qquad A = \frac{\text{MTBF}}{\text{MTBF} + \text{MTTR}}$$

MTBF = Mean time between failure

MTTR = Mean time to repair.

The unavailability of the system is given by

$$U = 1 - A < \frac{\text{MTTR}}{\text{MTBF} + \text{MTTR}}$$

High speed. The switching speed should be high enough to make use of the switching system efficiently. The speed of switching depends on how quickly the control signals are transmitted. For instance, the seize signal from the calling terminal must be identified quickly by the system to realise the need of path setup by the subscriber. The common control should be used effectively to identify the called terminal or the free trunks to setup a path. Thus the switching system must have the facility of quick access of the switching equipment and networks.

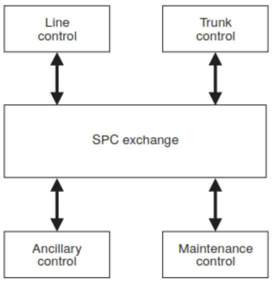
Low down time. The down time is the total time the switching system is not operating satisfactorily. The down time is low enough to have high availability. The unavailability of switching system may be due to failure of equipment, troubles in transmission media, and human errors in switching etc.

Good facilities. A switching system must have various facilities to serve the subscriber. For example wake up calls, address identification on phone number or phone number identification on address, recording facilities, quick service for the emergency numbers, good accessibility etc. Also it should have good servicing facilities in case of repair of equipment, skilled technicians, standby systems, etc. Good facilities is possible any switching system whether it is at rural or town or in cities, if that exchange is not overloaded.

High security. To ensure satisfied or correct operation (*i.e.* providing path and supervising the entire calls to pass necessary control signals) provision should be provided in the switching system.

Duplicated common control circuits, registers, processors and standby systems are used provide high security.

There are two classes of switching system based on the division of information in space, time. They are (*i*) Space division switch (*ii*) Time division switch. The space division provides fixed path for the entire duration of a call. Simply, unlimited bandwidth, cross talk limitations are the advantages of space division switches. But these space switches are slow to operate, bulky, and involves large amount of wiring. In time division switching all inlets and outlet one connected to a common switch mechanism. The switch is connected to the required inlet and outlet for short durations. Each input is sampled to change the connecting pattern. Thus switch is fast and compact. This technique may only be used where the signal is not affected by the sampling process. Time division switches of analog signals have limited applications. Thus time division switches have more practical value only when the signal is already in digital form.



In 1965, Bell system installed the first computer controlled switching system which uses a stored program digital computer for its control functions. The SPC concepts permits the features like abbreviated dialling, call forwarding, call waiting etc. The SPC provides significant advantages to end users. The SPC enables easier number changes, automated call tracing message unit accounting (for billing) etc.

In SPC, a programme or a set of instructions are stored in its memory and executed automatically one by one by the processor. Carrying out the exchange control functions through programs stored in the memory of a computer led to the name stored program control. A computer can be programmed to test the conditions of the inputs and last states and decide on new outputs and states. The decisions are expressed as programs which can be rewritten to modify or extend the functions of control system. All switching systems manufactured for use as public switching systems now use computers and software programming to control the switching of calls. Using SPC, 20 mA transmitter (old transmitter need 23 mA) with 52 V battery feed and longer subscriber loop can be achieved.

The SPC uses processors designed to meet the various requirements of the exchange. More than one processors are used for the reliability. Normally these processors are duplicated. Also the SPC system uses distributed software and hardware architectures. To carry over the maintenance functions of the switching system, a separate processor is used. Using the above setup, the SPC performs trunk routing to other control or tandem offices. Special features and functions are also enabled with sophisticated equipment’s and in compact form. There are two types in SPC exchanges, namely centralised SPC and distributed SPC.

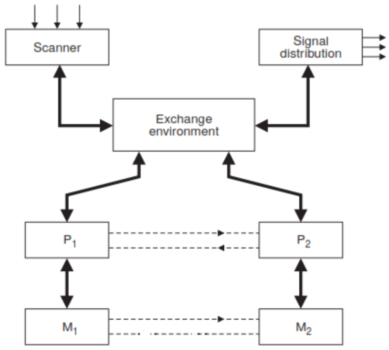


Fig. Centralized SPC.

Early electronic switching systems are centralised SPC exchanges and used a single processor to perform the exchange functions. Presently centralised exchanges uses dual processor for high reliability. All the control equipment are replaced by the processors. A dual processor architecture may be configured to operate in (*a*) standby mode (*b*) synchronous duplex mode and (*c*) Load sharing mode.

Standby mode. In this mode, any one of the processors will be active and the rest is standby. The standby processor is brought online only when the active processor fail. This mode of exchange uses a secondary storage common to both processors. The active processor copies the status of the system periodically and stores in axis secondary storage. In this mode the processors are not connected directly. In secondary storage, programs and instructions related to the control functions, routine programs and other required information are stored.

Synchronous duplex mode. In this mode, the processors *p*1 and *p2* are connected together to exchange instructions and controls. Instead of a secondary storage common to P1 and P2, separate memory M1 and M2 are used. These processors are coupled to exchange stored data. This mode of operation also uses a comparator in between *p2*. The comparator compares the result of the processors. During normal operation, both processors receives all the information from the exchange and receives related data from their memories. Although only one processor actually controls the exchange and remaining is in synchronism with first one. If a mismatch occurs, the fault is identified by the comparator, and the faulty processor is identified by operating both individually. After the rectification of fault, the processor is brought into service.

Load sharing mode. In this mode, the comparator is removed and alternatively an exclusion device (ED) is used. The processors calls for ED to share the resources, so that both the processors do not seek the same resource at the same time. In this mode, both the processor are active simultaneously and share the resources of exchange and the load dynamically. If one processor fails, with the help of ED, the other processor takes over the entire load of the exchange. Under normal operation, each processor handles one half of the calls on a statistical basis. However the exchange operator can vary the processor

Single processor.

Availability where MTBF = Mean time between failures, MTTR = Mean time to repair.

Unavailability = 1 – A $\qquad A = \frac{MTBF}{MTBF + MTTR}$

$$U = 1 - \frac{MTBF}{MTBF + MTTR} \; ; \; U = \frac{MTTR}{MTBF + MTTR}$$

If MTBF >> MTTR,

Dual Processor. A dual processor system is said to have failed only when both processor fails and the total system is unavailable. The MTBF of dual processor is given by

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

where $(MTBF)_D$ = MTBF of dual processor, MTBF = MTBF single processor

Availability $$A_D = \frac{(MTBF)_D}{MTTR + (MTBF)_D}$$

Substituting $(MTBF)_D$ in the above equation, we have

$$A_D = \frac{(MTBF)^2 / 2MTTR}{MTTR + \dfrac{(MTBF)^2}{2MTTR}}$$

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

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

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

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

For example problems, please refer text books.

The introduction of **distributed SPC** enabled customers to be provided with a wider range of services than those available with centralised and electromechanical switching system. Instead of all processing being performed by a one or two processor in centralised switching, functions are delegated to separate small processors (referred as regional processors). But central processors is still required to direct the regional processors and to perform more complex tasks.

$$U = \frac{MTTR}{MTBF}$$

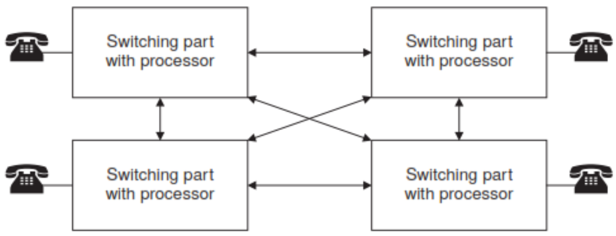


Fig. Distributed SPC

SWITCHING TECHNIQUES

This section describes various techniques used to establish connections between users' exchanges. Switches are hardware and/or software devices used to connect two or more users' temporarily. Message switching, circuit switching and packet switching are the most important switching methods. The terminals of the message switching systems are usually tele printers. In this switching, delays are incurred but no calls are lost as each messages are queued for each link. Thus, much higher link utilisation is achieved. The reason for the delay is that the system is designed to maximise the utilisation of transmission links by queueing message awaiting the use of a line. This switching is also called store and forward switching. The circuit switching sets up connection between the telephones, telex networks etc. which interchange information directly. If a subscriber or system to which connection to be made as engaged with other connection, path setup cannot be made. Thus circuit switching is also referred as lost call system. The modified form of message switching is called packet switching. Packet switching system carries data from a terminal or computer as a short packets of information to the required destination. This system is midway between message switching and circuit switching.

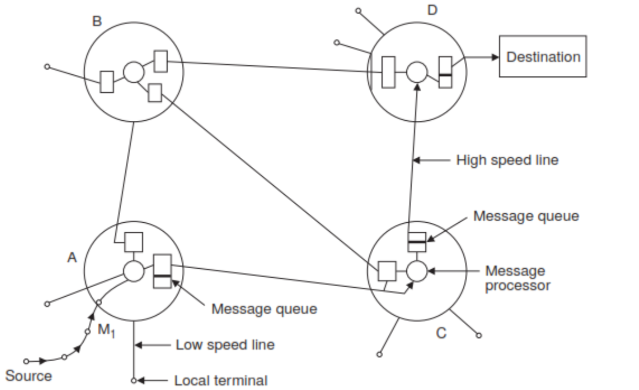
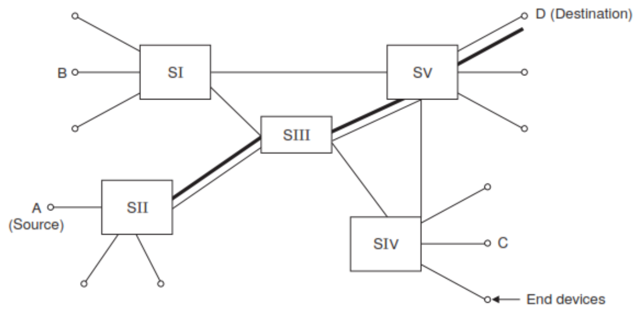


Fig. Message switching

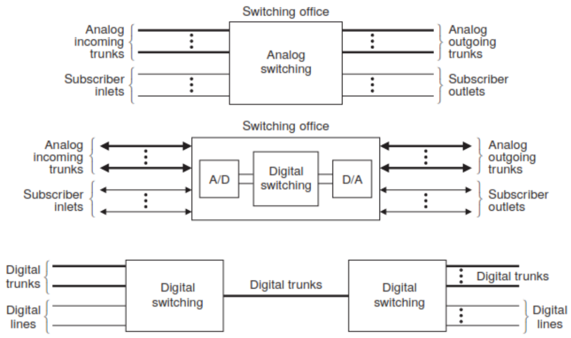


S1-SN → Switches, —Path between switches, ━Path setup between end device A&D,

Fig. Circuit switching

Message switching	Cirtuit switching
The source and destination do not interact in real time	The source and destination are connected temporarily during data transfer.
Message delivery is on delayed basis if destination node is busy or otherwise unable to accept traffic.	Before path setup delay, may be there due to busy destination node. Once the connection is made, the data transfer takes place with negligible propagation time.
Destination node status is not required before sending message.	Destination node status is necessary before setting up a path for data transfer.
Message switching network normally accepts all traffic but provides longer delivery time because of increased queue length.	A circuit switching network rejects excess traffic, if all the lines are busy.
In message switching network, the transmission links are never idle.	In circuit switching, after path setup, if the users denied service, the line will be idle. Thus, the transmission capacity will be less, if the lines are idle.

DIGITAL SWITCHING

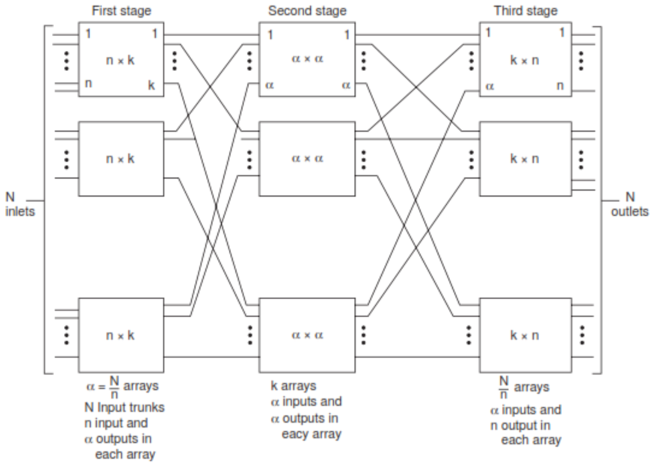


SPACE DIVISION SWITCHING

The fundamental operation of a switch is to setup and release connection between subscribers. It involves direct connection between subscriber loops at an end office or between station loops at a PBX. The switches are hardware and/or software devices capable of creating temporary connections between two or more subscribers. In space division switching, the paths in the circuit are separated from each other spatially. It was originally designed for analog networks, but is used currently in both digital and analog switching. A cross point switch is referred to as a space division switch because it moves a bit stream from one circuit/bus to another. For large group of outlets, considerable savings in total cross points can be achieved if each inlet can access only a limited number of outlets. Such situation is called limited availability. By overlapping the available outlet groups for various inlet groups, a technique called ‘‘grading’’ as established. Rectangular cross point array is an example of grading. For longer trunk groups, large cross points were expensive and not used now-a-days. The number of crosspoints required are M × N, where M is number of inlets and N is number of outlets.

Multistage Switching

It is inefficient to build complete exchanges in single stages. Single stage can only be used to interconnect one particular inlet outlet pair. Also the number of cross points grows as the square of the inputs for grading, N (N–1)/2 for a triangular array and N (N–1) for a square array. Also the large number of cross points on each inlet and outlet line imply a large amount of capacitive loading on the message paths. Therefore, it is usual to build exchanges in two or three stages to reduce the number of cross points and to provide alternative paths. The sharing of cross points for potential paths through the switch is accomplished by multiple stage switching. Fig. shows the three stage switching structure to accommodate 128 input and 128 output terminals with 16 first stage and last stage.



The structure shown in Fig. provides path for N inlets and N outlets. The N input lines are divided into N/*n* groups of *n* lines each. Each group of *n* inputs is accommodated by an *n*-input, *k* output matrix. The output matrices are identical to the input matrices except they are reversed. The intermediate stages are *k* in number and N/*n* inputs and N/*n* outputs. The interstage connections are often called junctors. Each of the *k* paths utilizes a separate center stage array. An arbitrary input can find *k* alternate output. Thus multistage structure provides alternate paths. Also the switching link is connected to a limited number of crosspoints. This enables the minimized capacitive loading.

The total number of crosspoints N_X for three stage is

$$N_X = 2NK + K\left(\frac{N}{n}\right)^2$$

where N = Number of inlets-outlets

n = size of each inlet-outlet group

k = number of second stage.

$2Nk$ = number of cross points in 1st and 2nd stage

$\left(\frac{N}{n}\right)^2$ = number of cross points in each array of second stage

$k\left(\frac{N}{n}\right)^2$ = number of cross points in second stage.

Also $$N_X = K\left[2N + \left(\frac{N}{n}\right)^2\right]$$

The three stage switching matrix require that $k > 2n - 1$ to generate no blocking.

$k = 2n - 1$

Substituting we get

$$N_X = 2N(2n-1) + (2n-1)\left(\frac{N}{n}\right)^2$$

For large N, $n = \sqrt{\frac{N}{2}}$

Substitute $\sqrt{\frac{N}{2}}$ for n

Thus $$N_x = 2N\,(2\sqrt{n/2} - 1) + (2\sqrt{N/2} - 1)\left(\frac{N}{\sqrt{\frac{N}{2}}}\right)^2$$

$$= 2N\,(\sqrt{2n} - 1) + (\sqrt{2n} - 1)\ 2N$$

$$N_{x(\min)} = 4N\,(\sqrt{2n} - 1)$$

Number of cross points for a single stage switching matrix to connect N inlets to N outlets is Nx (SS)=N^2

$$\lambda \equiv \frac{N_x\ (\text{min, 3 stage})}{N_x(\text{SS})} = \frac{4\sqrt{2}\ N^{3/2}}{N^2}.$$

$$\lambda = \frac{4\sqrt{2}}{\sqrt{N}}$$

where Nx (SS) = Number of cross points in single stage. In practice, this gives reasonable scaling.

Blocking Probability Evaluation Techniques

All the switching systems are designed to provide a certain maximum probability of blocking for the busiest hour of the day. It is one of the aspects of the grade of service of the telephone company. There are variety of techniques to evaluate the blocking probability of a switching matrix. Depends on the accuracy, required availability, geographical area, priority, complexity and applicability of different network structures, the techniques are varying. Here, two techniques are described.

1. **Lee graphs.** It was proposed by C.Y. Lee. It is a most versatile and straight forward approaches of calculating probabilities with the use of probability graphs.

2. **Jacobaeus method.** It was presented in 1950 by C. Jacobaeus. It is more accurate than Lee graph method.

Lee graphics. C.Y. Lee's approach of determining 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, $B = 1 – q^n$For a probability graph of three stage network, shown in Fig. the probability of blocking is given by$B = (1 – q^2)^k$where q' = probability that an interstage link is idle = $1 – p'$ p' = probability that any particular interstage link is busy k = number of centre stage arrays.If p is known, the probability that an interstage link is busy is given by$p' = p^ß$ where ß = k/nß is the factor by which the percentage of interstage links that are busy is reduced.

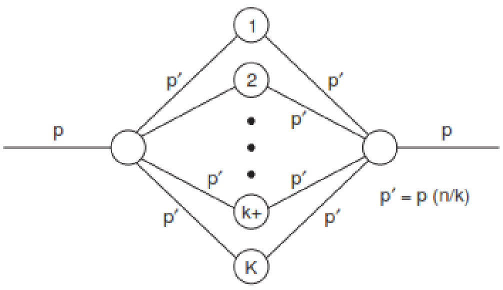


Fig. Probability graph of three stage network. Substituting we set $q' = 1 – p/ß$

Substituting again we get complete expression for the blocking probability of a three stage switch in terms p as

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

with inlets of 10% busy, the switch size of N with $n = 8$, $h = 5$, ß = 0.625 requires 2560 crosspoints. The merits of this method are(*i*) It provide accurate results(*ii*) Its formulas are directly relate to the network structures(*iii*) It provides insight of the network and thus provides ideas to change the structure for high performance.

Jacobaeus. The Lee's graph approach is not much accurate. Because the probability graphs entail several simplifying assumptions. The important one which gives erroneous values of blocking is the assumption that the individual probabilities are independent. In fact the probabilities not independent and highly dependent when significant amounts of expansion are not present. According to C. Jacobaeus the blocking probability of a three stage switch is

$$\mathrm{B} = \frac{(n\,!)^2}{k\,!\,(2n-k)\,!}\, p^k\,(2-p)^{2n-k}$$

where n = number of inlets (outlets) per first (third) stage array

k = number of second stage array

p = inlet utilization.

More accurate techniques can be used for systems with high concentrations and high blocking. As the high blocking probabilities not having much practical value, those techniques are not considered.

TIME DIVISION SWITCHING

In space division switching, crosspoints are used to establish a specific connection between two subscribers. The crosspoints of multistage space switches assigned to a particular connection is dedicated to that connection for its duration. Thus the crosspoints cannot be shared. Time division switching involves the sharing of crosspoints for shorter periods of time. This paves way for the reassign of crosspoints and its associated circuits for other needed connections. Therefore, in time division switching, greater savings in crosspoints can be achieved. Hence, by using a dynamic control mechanisms, a switching element can be assigned to many inlet-outlet pairs for few microseconds. This is the principle of time division switching. Time division switching uses time division multiplexing to achieve switching. Two popular methods that are used in time division multiplexing are (*a*) the time slot interchange (TSI) and (*b*) the TDM bus. In ordinary time division multiplexing, the data reaches the output in the same order as they sent. But TSI changes the ordering of slots based on the desired connections. The demultiplexer separates the stots and passes them to the proper outputs. The TDM uses a control unit. The control unit opens and closes the gates according to the switching need. The principle of time division switching can be equally applied to analog and digital signals. For interfacing sampled analog signals but not digitized, the analog time division switches are attractive. But for larger switches, there are some limitations due to noise, distortion and crosstalk which normally occurs in PAM signals. Thus analog switching is now used only in smaller switching systems.

Analog Time Division Switching

Fig. shows a simple analog time division switching structure. The speech is carried as PAM analog samples or PCM digital samples, occurring at 125 μs intervals. When PAM samples are switched in a time division manner, the switching is known as analog time division switching. If PCM binary samples are switched, then the switching is known as digital time division switching. A single switching bus supports a multiple number of connections by interleaving PAM samples from receive line interfaces to transmit line interfaces. There are two cyclic control stores. The first control store controls gating of inputs onto the bus one sample at a time. The second control store operates in synchronism with the first and selects the appropriate output line for each input sample.

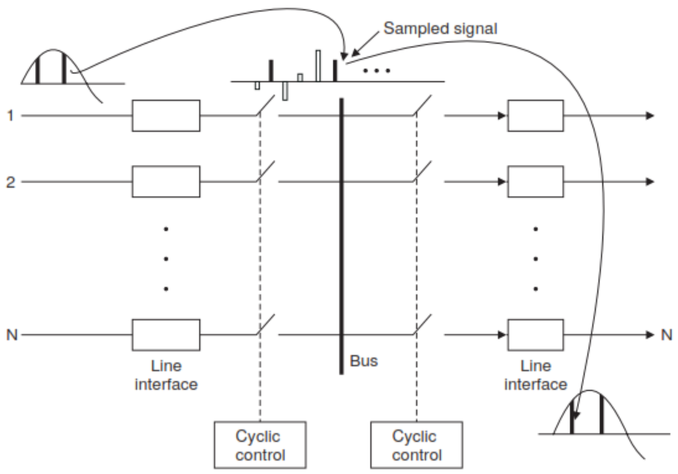
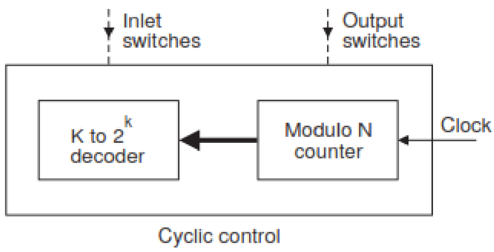


Fig. Analog Time Division Switching structure.

The selection of inlet/outlet is controlled by various ways. The (*a*) cyclic control and (*b*) memory based control are the important controls.



The cyclic control is organised by using Modulo-N counter and k to 2^k decoder. The k and N are related by

$$\lceil \log_2 N \rceil = k$$

where N = number of inlets/outlets

k = decoder size. It means k may be assumed lowest integer or more than that.

This kind of switching is non-blocking but lack of full availability as it is not possible to connect inlet to any outlet. The switching capacity or number of channel supported by cyclic controlled system is

$$C = \frac{125\,\mu\,\text{sec}}{t_s}$$

Memory based control. Full availability can be achieved if any one control is made memory based. If the input side is cyclically switched and the outlets are connected based on the addresses of the outlets stored in contiguous location is referred as input controlled or input driven. If the outlets are cyclically switched, the switch is referred as output controlled or output driven. As the physical connection is established between the inlet and the outlet through the common bus for the duration of one sample transfer, the switching technique is known as time division space multiplexing. For this system,

$$C = \frac{125\,\mu\,\text{sec}}{t_i + t_m + t_d + t_t}$$

t_m= time to read the control memory

t_d= time to decode address and select the inlet and outlet.

t_i= time to increment the modulo N counter.

t_t= time to transfer the sample.

The capacity equations are valid only for a 8 kHz sampling and non folded network (can be used for folded network with certain changes in network). The switching capacity in the memory controlled is equal to N. The use of cyclic control in input or output controlled switches restricts the number of subscribers on the system rather than the switching capacity since all the lines are scanned whether it is active or not. No restrictions on subscriber number and full availability of the switching system can be achieved by designing a switching configuration with control memory for controlling both inlets and outlets. This configuration referred to as memory controlled time division space switch is shown in Fig. As each word of the control memory has inlet address and an outlet address, the control memory width is $2^{\lceil \log_2 N \rceil}$. modulo counter is updated at the clock rate. For the path setup of addresses are entered in control memory and path is made. Then the location is marked busy. When conversation is terminated, the addresses are replaced by null values and location is marked free. Hence

$$C = \frac{125}{t_s}\,\mu\,\text{sec, where } t_s = t_i + t_m + t_d + t_t$$

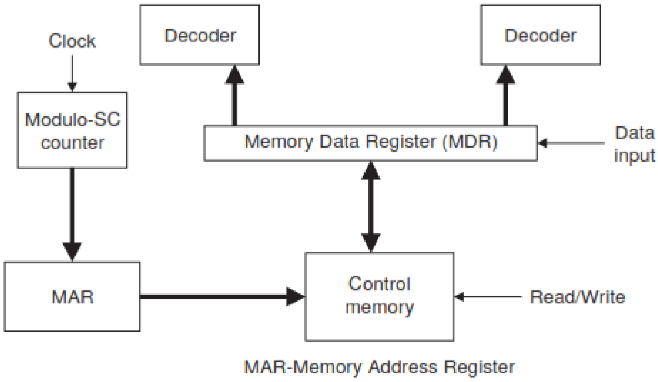


Fig. memory controlled for both input and output.

Digital Time Division Switching

The analog time division switching is useful for both analog and digital signals. The digital time division multiplexed signals usually requires switching between time slots as well as between physical lines. The switching between time slots are usually referred as time switching. Similar to analog time division switching the switching structure can be organised expect the use of memory block in place of the bus. This adds the serial to parallel and parallel to serial bit conversion circuitry's as the input to the memory block should be in parallel form. The time division switch can be controlled in any of the following three ways.

The basic requirement of time division switching is that the transfer of information arriving at in a time slot of one input link to other time slot of any one of output link. A complete set of pulses, arriving at each active input line is referred to as a frame. The frame rate is equal to the sample rate of each line. A time switch operates by writing data into and reading data out of a single memory. In the process the information in selected time slots is interchanged as shown in Fig.

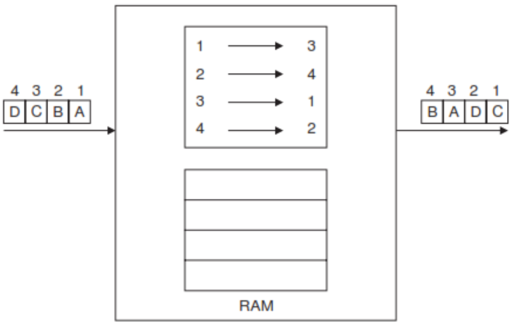


Fig. TSI operation

In TSI operation, inputs are sequentially controlled and outputs are selectively controlled. The RAM have several memory locations, each size is the same as of single time slot. Fig. shows the general arrangement of the time division time switching.

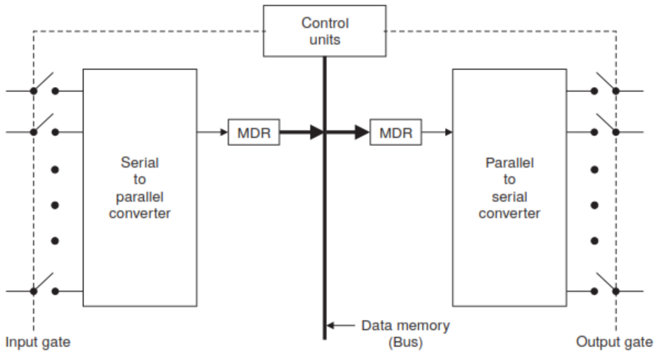
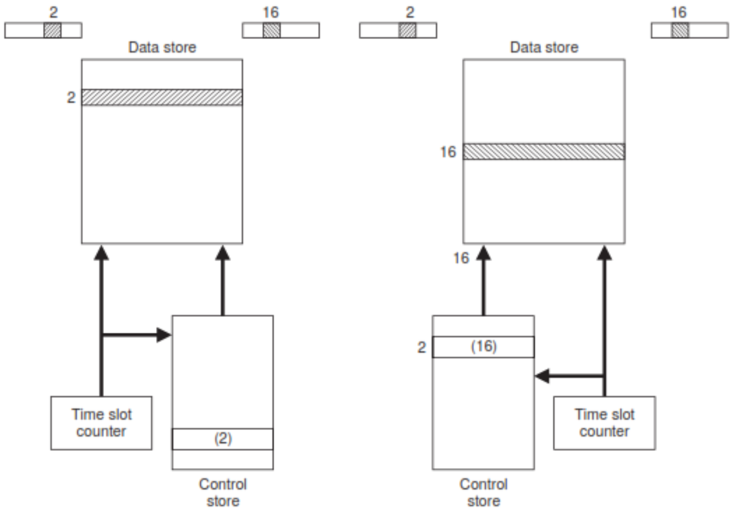


Fig. TDTS diagram

The serial to parallel and parallel to serial converter are used to write the data into the memory and read the data out of memory. For convenience, two MDR are shown, but MDR is a single register. Gating mechanism is used to connect the inlet/outlet to MDR. The input and output lines are connected to a high speed bus through input and output gates. Each input gate is closed during one of the four time slots. During the same time slot, only one output gate closed. This pair of gates allows a burst of data to be transferred from one input line to a specific output line through the bus. The control unit opens and closes the gates according to switching need. The time division time switch may be controlled by sequential write/random read or random write/ sequential read. Fig. depicts both modes of operation and indicates how the memories are accessed to translate information from time slot 2 to time slot 16. Both methods use a cyclic control. Fig. (*a*) implies that specific memory locations are dedicated to respective channels of the incoming TDM link. Data are stored in sequential locations in memory by incrementing modulo N counter with every time slot. Thus incoming data during time slot 2 is stored in the second location within the memory. On output, information retrieved from the control store specifies which address is to be accessed for that particular time slot. Thus sixteenth word of control store contains the number 2, implying that the contents of data store address 2 is transferred to the output link during outgoing slot 16. Random write/sequential read mode of operation is opposite to that of sequential write/random read. Incoming data are written into the memory locations as specified by the control store, but outgoing data are retrieved sequentially under control of an outgoing time slot counter. The data received during time slot 2 is written directly into data store address 16 and it is retrieved during outgoing TDM channel number 16.



(a) (b)

TWO DIMENSIONAL DIGITAL SWITCHING

Combination of the time and space switches leads to a configuration that achieved both timeslot interchange and sample switching across trunks. These structures also permit a large number of simultaneous connections to be supported for a given technology. Large digital switches require switching operations in both a space dimension and a time dimension. There are a large variety of network configurations that can be used to accomplish these requirements. The incoming and outgoing PCM highways are spatially separate. So the connection of one line of local exchange obviously requires space switching to connect to the channel of outgoing highways. Thus the switching network must be able to receive PCM samples from one time slot and retransmit them in a different time-slot. This is known as time slot interchange, or simply as time switching. Thus the switching network must perform both space and time switching. The space switching and time switching may be accomplished in many ways. A two stage combination switch may be organised with time switch as first stage and the space switch as the second stage or vice versa. The resulting configurations are referred as time space (TS) or space time (ST) switches respectively. Three stage time and space combinations of TST and STS configurations are more popular and flexible. Very large division switches includes many combinations of time and space switches. Typical configurations are TSST, TSSSST, and TSTSTSTS. These switches support 40000 lines or more economically. The general block diagram involving time and space switching is shown in Fig.

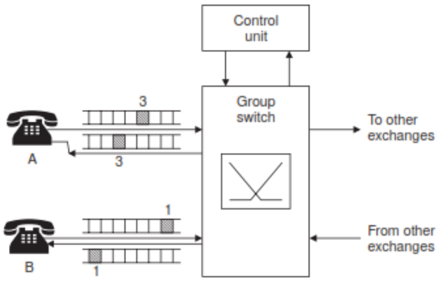


Fig. Combination switching

The main task of the switching part is to interconnect an incoming time slot and an outgoing time slot. The unit responsible for this function is group switch. There are two types of building block in the digital group switch. They are time switch and space switch. In Fig. the subscriber makes a local call to B. The control unit has assigned timeslot 3 to the call on its way into the group switch, and time slot 1 on its way out of the group switch (to B). This is maintained during the entire call. Similarly B to A also carried out. The fundamental design and structure of the two switches viz. time switch and space switch are described in the following sections.

Space and Time Switches

Space switch. Fig. shows a typical space switch. It uses a space array to provide switching generally the space switch consists of a matrix of M × N switching points where M is number of inlets and N is number of outlets. A connection between an inlet and an outlet is made by the simple logic gates (AND gates). As logic gates are unidirectional, two paths through switching matrix must be established to accommodate a two way conversation. The logic gate array can serve for concentration, expansion or distribution depending on M is larger, smaller or equal to N. Fig. shows only one voice direction. However, the corresponding components are available for the opposite direction too. A number of M, of X slot multiplexers, provide the inputs and the outlets are connected to N, X slot demultiplexers. The gate select memory has X locations. The word containing information about which cross point is to be enabled is decoded by the translator. During each internal time slot, one cross point is activated. In the shift to the next interval time slot, the control memory is incremented by one step, and a new crosspoint pattern is formed in the matrix.

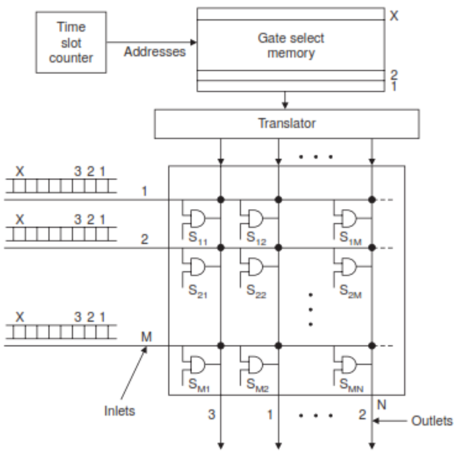


Fig. space switch

Time switch. The time-slot interchange (TSI) system is referred to as time switching (T-switching). Fig. shows the block diagram of time switch. Each incoming time slot is stored in sequence in a speech memory (SM). The control memory (CM) determines in which order the time slots are to be read from SM. This means that a voice sample may be moved from say incoming time slot 3 to outgoing time slot 1.

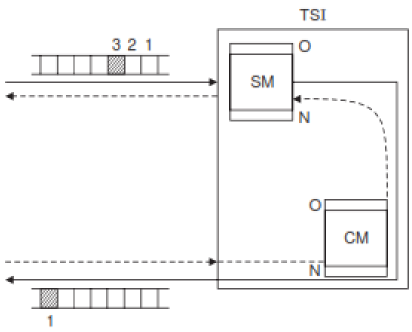


Fig. Time switch

Time-space (TS) Switching

This switch consists of only two stages. This structure contains a time stage T followed by a space stage S as shown in Fig. Thus this structure is referred to as time-space (TS) switch. The space array have N inlets and N outlets. For each inlet line, a time slot interchange with T slots is introduced. Each TSI is provided with a time slot memories (not shown). Similarly a gate select memory needs to be provided for the space array (not shown).

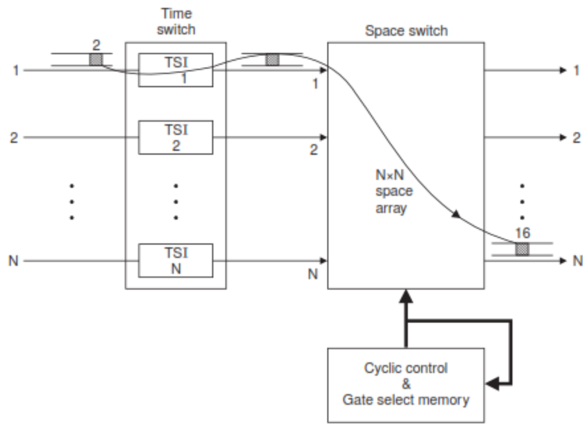


Fig. TS switch

The transmission of signals carried out from sender to receiver through multiplexer input and demultiplexer output. The reverse communication also similar. Thus a hybrid arrangement is needed to isolate the transmitted signal from the received signal. The basic function of the time switch is to delay information in arriving time slots until the desired output time slot occurs. Let the communication is to take place between subscriber A and B. Let A is assigned time slot 2 and line 7 and subscriber B is assigned time slot 16 and line 11. Then the signal moved from time slot 2 to time slot 16 by the time-slot exchanger and is transferred from line 7to line 11 in the space array. Similarly, the signal originated by B is moved from slot 16 to slot2 through line 11 to 8.The cyclic control and gate select memory contains the information needed to specify the space stage configuration for each individual time slot of a frame. The time stage have to provide decays ranging from one time slot to a full frame. During each outgoing time slot, control information is accessed that specifies interstage link number to output link. During other time slots, the space switch is completely reconfigured to support other connections.

Let each time slot interchanger have T slots. If the space array is a N × N, then the simultaneous connections possible is NT. If T = 128 and N = 16, 2048 connections can be supported. This structure is not free of blocking. The control store is a parallel end around shift resister. If space array is at the inlet side and time switch is at the output side, the structure is referred as space time (ST) switching. Both TS and ST arrangements are equally effective.

Blocking probability :

The blocking probability of TS switching is calculated as follows.

The probability that a subscriber A is active = $\frac{\rho}{T}$

where ρ = fraction of time that a particular link is busy measured in Erlangs

T = number of time slots in a frame.

The probability that any other subscriber is active on the same link

$$= \frac{(T-1)\rho}{T}$$

The probability that a particular called subscriber is chosen by A

$$= \frac{1}{NT} \times \frac{1}{T}$$

where N = Number of inlets (or outlets) for N × N space array.

NT = Simultaneous connection

The probability that the same time slot on a different outlet is chosen by the other subscribers on the same inlet

$$= \frac{(T-1)(N-1)\rho}{T(NT-1)}$$

From Blocking probability $= B = \left(\frac{\rho}{T \times NT}\right)\left(\frac{(T-1)(N-1)}{T(NT-1)}\right)$

As T >> 1 and N >> 1, & NT >> 1

$$B = \frac{P}{NT^3}$$

The TS switch can be made non-blocking by using an expanding time switch (T to T slots) and a concentrating space switch (which is complex).

Implementation complexity. In general the complexity of the switching is represented in terms of number of cross points (N) and its associated cost. The number of cross points in space stage can be easily calculated which is based on the array size. The time stage uses significant amount of memory which adds the cost of the whole system. To take this into account the cost of memory bit is assumed one hundredth of the cost of cross point. Thus,

Implementation complexity $= N_X + \frac{N_B}{100}$

where N_X = Number of space stage cross points

N_B = Number of bits of memory.

The N_B not only includes the time stage memory arrays, but also the control memory (store) of the time stage and space stage. Thus,

$$N_B = N_{BX} + N_{BT}$$

where N_{BX} = Number of memory bits for the space stage control store

= N × (Number of control words) (number of bits per control word)

N_{BT} = Number of memory bits in the time stage equal to sum of time slot interchange and the control store bits.

= N × number of channels × number of bits per channel + N × number of control words × number of bits per control world.

For example problems please refer text books.

STS and TST Switching

The TS structure is of blocking nature. Let A and B are the subscribers using different timeslot on the same line want to connect to two subscribers C and D using same time slot on different lines. A and B can be moved to the same time slot but during that time slot, the inlet line can be connected to C's line or D's line but not both. This is the significant limitation of the structure. Moreover, time stage switching is generally less expensive than space stage switching as digital memory is much cheaper than digital cross points (AND gates).The multiple stages overcomes the limitations of the individual switches and cost savings can also be achieved. TST, STS, TSST, TSSSST and TSTSTSTSTSTSTS are the switching system configurations used in digital switching system. However, the TST structure is the most common.

STS Switching: In STS switching, the time stage is sandwiched between two space arrays. The digital switching system ITS 4/5 of USA (1976) uses the STS switching configuration. It handles 3000 trunks and accommodates 1500 Erlangs of traffic. Fig. shows the space-time-space (S-T-S) switching network for M incoming and outgoing PCM highways. Establishing a path through an STS switch requires finding a time switch array with unavailable units' access during the incoming time slot and an available read access during the desired outgoing time slot. The input side space stage as well as the output side space stage is free to utilise any free time switch modules. In the diagram shown in Fig. the time slot 2 is connected to the TSM 2 where the time slot allotted is 16 and passed to the (M – 1)th line of output space array. Thus the path is provided. This structure is of non-blocking nature.

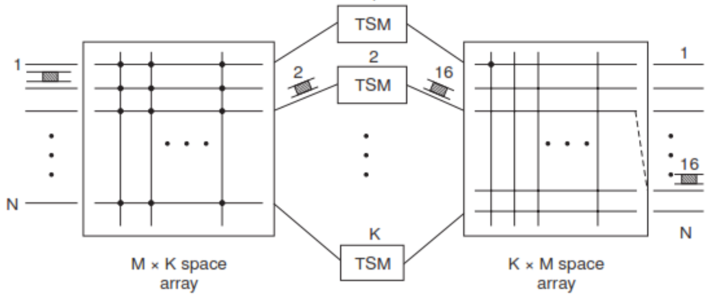


Fig. STS switching structure.

Blocking probability. The STS switch is identical to the probability graph of three stage space switches (Fig. 5.12). Similar to that, the blocking probability of an STS switch is

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

where p = probability that a link is busy

$\beta = \frac{K}{N}$ = is the factor by which the percentage of links that are busy is reduced. ($\beta < 1$)

K = number of center stage TSM.

Implementation capacity (IC). While calculating IC, the total number of two space stage cross points, total number of two space stage control bits, number of time stage memory bits and number of time stage control bits are to be considered. Thus,

$$IC = 2KN + \frac{2KC \log_2 N + KC(8) + KC \log_2 C}{100}$$

where K = The minimum number of centre stage TSM to provide desired grade of service, calculated from

C = number of channel.

TST Switching. In TST switching the space stage is sandwiched between two time stage switches. Of all the multistage switching, TST is a popular one. Some important features of TST switches are:

(*i*) **Low blocking probability.** An incoming channel time slot may be connected to an outgoing channel time slot using any possible space array time slot. Thus there are many alternative paths between two subscribers. This concept reduces the blocking probability of a three stage combination switch.

(*ii*) **Stage independency.** The space stage operates in a time-divided fashion, independently of the external TDM links. The number of space stage time slots L does not coincide with the number of external TDM time slots T.

(*iii*) **Implementation advantage.** The factors to be considered for switching design and implementation are traffic loads, modularity, testability, expandability and simple control requirements. For large switches with heavy traffic loads, the TST have good implementation advantage.

(*iv*) **More cost effective.** If the input channel loading is high, the time expansion of TST and space expansion of STS are required. Time expansion of TST can be achieved at less cost than space expansion of STS. In comparison with STS, the TST have certain limitations. For small switches, the STS architectures are less complex to implement than TST. The control requirements of STS is simpler than TST.

The principle of operation of TST switching is shown in Fig. In figure, two flows of time slots, one for each direction are connected together.

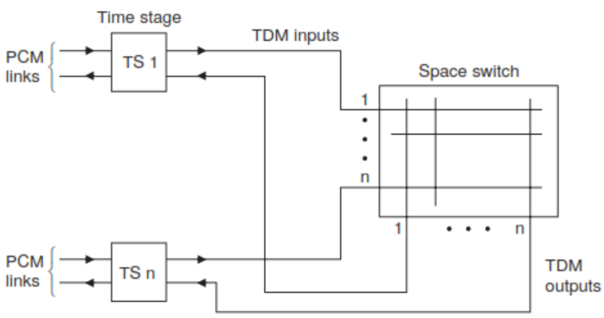


Fig. TST switching

The functional block diagram which explains the transfer of signals from inlet to outlet is shown in Fig. The information arriving at the incoming link of TDM channel is delayed in the· inlet times stage until an appropriate path through the space stage is available. Then the information is transferred through the space stage to the appropriate outlet time stage. Here the information is held until the desired outgoing time slot occurs. Any space stage time slot can be used to establish a connection. The space stage operates in a time divided fashion independently of the external TDM links. There are many alternative paths between a prescribed input and output unlike a two stage network which has only one fixed path.

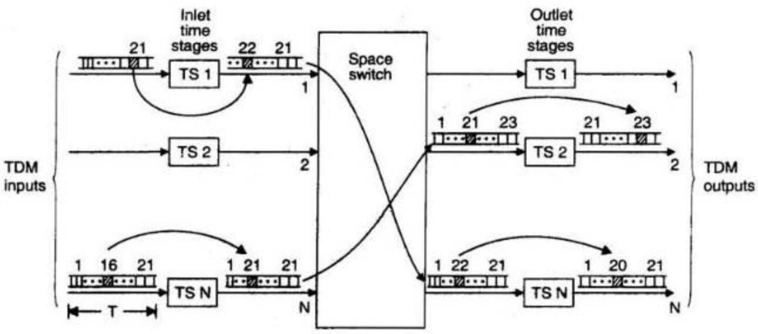


Fig. TST switching structure.

Blocking probability. The blocking probability is minimised if the number of space stage time slots L is made to be large. By direct analogy of three stage space switches, the TST switch is strictly non-blocking if

$$L = 2T - 1$$

where T = number of time slot of time switch.

L = number of space slot of space switch.

The probability graph of TST switch with non-blocking stage is shown in Fig.

Module-II: COMPUTER CONTROLLED SWITCHING SYSTEMS

Introduction, Call processing, signal exchange diagram, state transition diagram, hardware configuration, switching system software organization, software classification and interfacing, Maintenance software, call processing software, Administration software, Electronic Exchanges in India.

INTRODUCTION

Most digital switching systems have a quasi-distributed hardware architecture, since they maintain control of the switching functions through an intermediate processors. All digital switching systems employ multiprocessor subsystem for the best understanding of communication and control process. The architecture of a working digital switching system is very complex with many subsystems. All present day digital switching system includes minimum software which are necessary for implementation of call processing for all the levels of control structure. In modern digital switching systems, many call processing functions are performed by using interface controllers. Some of the call processing are call identification, call routing, path setup between subscribers, digital translation, call status, billing etc.

CALL PROCESSING

In this section, the basic steps involved in processing a call is discussed. Most digital system follow a similar scheme. For any switching system design, the range of signals that has to be interchanged between a terminal and system is considered. These signals described in signal exchange diagram. The sequence of operation between subscribers and system are shown in state transition diagram (s.t.d.).

Basic Steps to Process a Call

The sequence of processing between subscribers are described below:

1. **Idle state.** At this state, the subscriber handset is in ‘on-hook’ condition. The exchange is ready to detect the call request from the subscriber.

2. **Call request identification.** The exchange identifies a line requiring for a service. When the handset is lifted, current flows in the line called seize signal indicates the call request.

3. **Providing dial tone.** Once the seize signal is received, an exchange sends a dial tone to the calling subscriber to dial the numbers.

4. **Address analysis.** Once the first digit received, the exchange removes the dial tone and collect all numbers. Then the address is analysed for the validity of the number, local, STD or ISD etc. If the number is invalid, a recorded message may be sent to the calling subscriber and terminates call request.

5. **Called line identification.** The exchange determines the required outgoing line termination from the address that it has received.

6. **Status of called subscriber.** The called line may be busy or free or unavailable or even out of service. In the case of PBX, where the customer have a group of lines, the exchange tests each termination until either it finds a free one or all one found busy. For busy, number unobtainable or the handset off hook, a status signal or call progress signal is sent to the calling subscribers for line termination. Now the exchange resumes idle state.

7. **Ringing.** Once, the exchange finds the called subscriber is free, power ringing is provided to the called subscriber and audible ringing to the calling subscriber.

8. **Path setup.** When the called subscriber lifts his handset, the line is looped and ringing is removed. Once the conversation started, the exchange completes the connections between the subscribers.

9. **Supervision.** The exchange supervises the connection to detect the end of the call for charging.

10. **Clear signal.** Once the need for connection is over, either customer may replace his handset. It causes the line current seize and provides a clear signal to exchange. If the calling subscriber replaces his phone set, the clear signal sent to the exchange is called clear forward signal. If called subscriber do first, the clear signal is called clear backward signal.

Signal Exchange Diagram

There are two types of diagrams used to represent the sequence of events between the subscriber and exchanges. They are signal exchange diagram and state transition diagram. Both diagrams

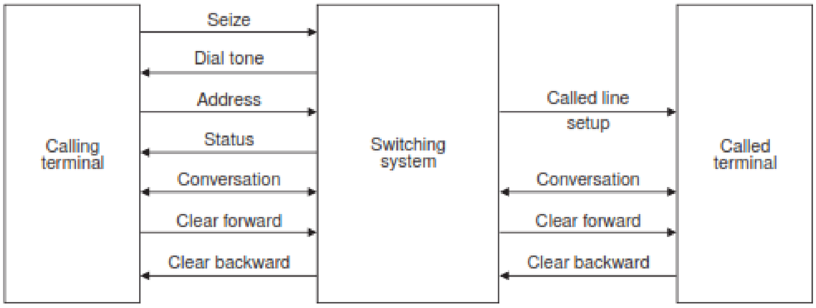


Fig. Signal exchange diagram

can be used to specify the behaviour of different control units in switching centre. For the local call, the steps involved in processing a call is shown in Fig. 6.1.Normally, once the conversation is over, the exchange will be at idle state. But in general, there are two types difficulties arises.

1. **Called subscriber held (CSH).** This condition arises when the called subscriber replaces the hand set but the caller does not. In this case, the caller does not originate a call or receiver a call.

2. **Permanent loop condition (PL).** This condition occurs when the caller replaces the phone but the called subscriber does not. Now, a loop present between called and exchange and it results in busy tone to another call to the same called subscriber. In strowger system, this condition is called permanent glow condition. In electromechanical system, the above conditions are removed by manual disconnection. In modern ESS systems, a time out process is used. If the call setup between two subscribers are made through many exchanges and trunks, the originating exchange where calling subscriber is connected sends the seize and then address to the terminating exchange where the called subscriber is connected. Remaining signalling are similar to the local call, but through the originating and terminating exchanges. In electromechanical system, the signalling between exchanges are sent through same interexchange circuits referred as channel associated signalling. In SPC controlled exchanges, interexchange signals are generated at originating exchange, but processed at terminating exchange. The signals are transferred over high speed data like instead of speech connections are referred as common channel signalling.

State Transition Diagram

The state transition diagram (s.t.d.) specifies the response of a control unit to any sequence of events. s.t.d. is a powerful design tool. It helps the designer to consider all possibilities of occurrence of events. Fig. shows the basic symbols used in a state transition diagram.

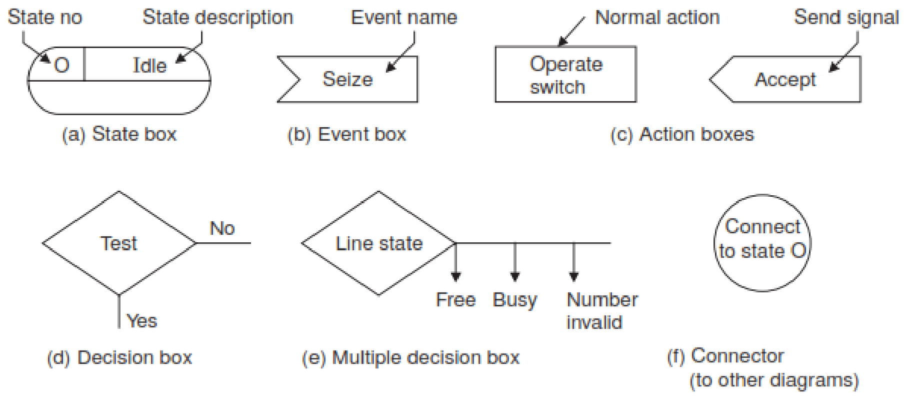


Fig. basic symbols of s.t.d.

The basic symbols are defined as follows:

State boxes. The state boxes are labelled with state number and state description. If necessary, additional information can also be included. The combination of the present state and a new event defines a task and performing this results in next state. Sometimes more than one state occurs, the choice depending on external information.

Event boxes. The intended arrow of the symbol indicate whether the event corresponds to the receipt of forward or backward signal. The forward signal and backward signal refers to the flow of signal from calling to called and called to calling subscriber through exchange respectively.

Action boxes. The rectangular box represents the action taken on the event. The protruding arrow indicates whether the signal is sent forward or backward.

Decision boxes. The diamond shaped box is used for the cases where two divisions are possible. For multiple decisions, another symbol shown in Fig. (*e*) is used.

Connectors. This symbols are used to connect one flow chart to another diagram.

Fig. shows the s.t.d. diagram for a typical local call. Let the calling subscriber is A and the called subscriber is B.

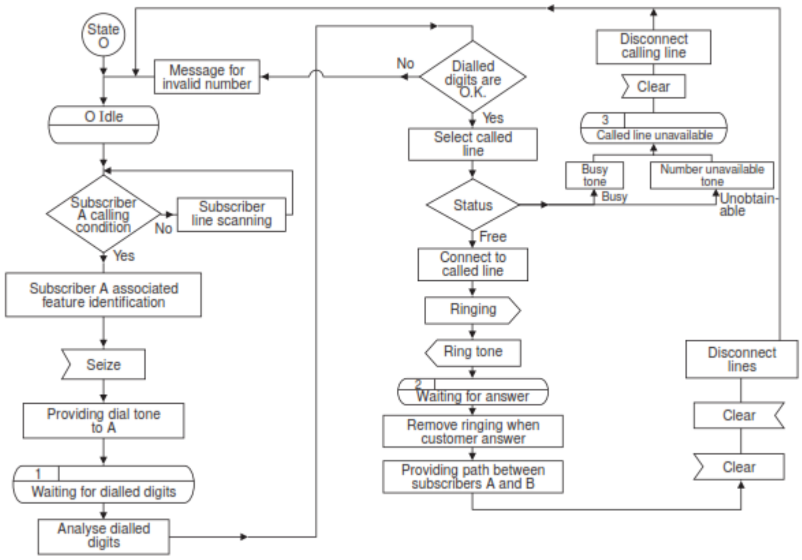


Fig. State transition diagram.

The computer controlled switching is in general referred as electronic switching system (ESS). ESS offers the greatest potential for both voice and data communications. An ESS consists of 1.computer

2. Memory or storage

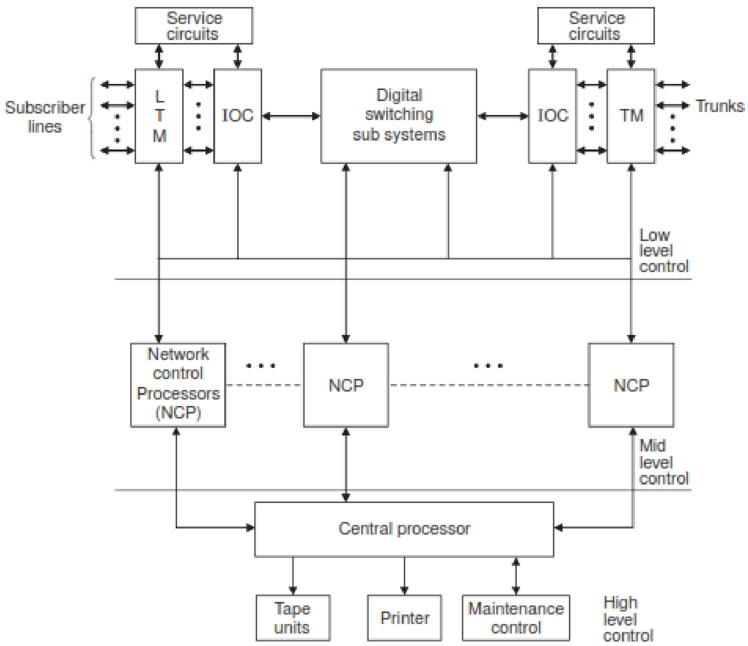
3. Programming capability

4. An extremely rapid switching component.

A computer based common control switching equipment implies two distinct type of units. They are 1. Control unit 2. Switching network. The common control receives, stores and interprets dial pulses and then selects an available path through the switching hardware to complete connection. Efficient high speed common control equipment can complete many calling connections during the time of an average phone call. Thus it saves a lot of time and money. The switching network can be used to connect many lines by one common group of control devices referred as control unit. Thus the control unit is the brain of a switching system, a control unit completes its function in a small fraction of a second for a single call. The hardware of digital switching system are broadly divided by their functions into many subsystems. The functions performed by the subsystem includes line and trunk access, line scanning, message interpretation, switching communications, path setup between subscribers, line supervision, line termination, billing providing advanced features and system maintenance. These subsystems are classified into various levels of control. Each level of control and its subsystems are tabulated in table.

Low level control	Mid level control	High level control
1. Line Terminating module 2. Trunk module 3. Input/output controller 4. Service circuits	1. Network control Processors	1. Central processors 2. Tape units 3. Printers 4. Maintenance control

A general hardware configuration is shown in Fig. However, various switching system may have different kind of arrangements of the subsystems. Most digital switching systems have a quasi-distributed hardware architecture, as the control of the switching functions are made through an intermediate processors. All digital switching systems employ multiprocessor subsystems as shown in Fig. A similar architecture is used by most of the digital telephone exchange systems. Some popular systems are AXE – 10 systems (Sweden), DMS – 10 (Canada),E – 10 system (France), No. 5 ESS system (USA) EWS D system (Germany) and the NEAX system (Japan). Fig. illustrates the hardware architecture of the digital switching system.



LTM : Line terminating modules, TM : Trunk modules,
IOC : Input/Output controller, NCP : Network control processors.

Low level control. This level associated with subscriber lines, trunks, selective circuits, Input/output controller and digital subsystems. The line terminating module and trunk modules are microprocessor based and communicate with subsystems through the input/output controllers. The input/output controllers interpret the incoming messages and takes necessary actions and communicate to the network control processors. All subscriber lines connected to digital switching system through the main distributing frame (MDF) are continuously scanned to detect the state of the subscriber.

When the customer lifts his handset, the line scanning program detects this state and reports to the input/output controller. The IOC is the primary peripheral controller and it controls all peripherals associated with call or trunk processing. At this level, all the requests of incoming and outgoing trunks are handled. Any advanced features to be incorporated in a digital switching system also handled at this level using IOC.

Mid-level control. This level is associated with network control processors and associated circuits. The IOC is controlled by the network control processors (NCP). Many NCP’s are used depends on the size of the digital switching system. A dedicated bus system is usually required for the processors to communicate with one another. Specific messaging protocols are used to communicate between processors. For messaging between the peripherals and external systems, many digital switching systems utilize standard protocols such as signalling system 7(SS7); X.25 and X.75. Thus this is the most important level of control any digital switching system. Distributed processing are performed at this level.

High level control. This level associated with central processor which organizes the entire network control sub processors. In includes many subsystems like call accounting subsystems (CAS), call processing subsystems (CPS). Digital switching subsystems (DSS).Digital subscriber’s switching subsystem (DSSS), Local administration (LA), maintenance control subsystems (MCS); management statistics subsystems (MSS), message transmission subsystems (MTS), signal interworking subsystems etc. This central processor is normally a main frame type computers. Thus all basic controls of a digital switching system are incorporated at this level. In real time operation, the processor determines the state of a call by reading data from memory. The store areas (not shown) include,

Line store. In this memory, the status of the line is stored. The status may be busy, free or disconnected.

Call record. All the call processing data’s such as origin of a call, path of a call, and duration of a call and clearing of a call are stored.

Translation tables. Most switching system require a look-up table in order to decode routing digits into suitable routings. For electromechanical system, such tables are realized by distribution frame. Hundreds of translation tables are built for a switching system which stores data for equipment number (EN) to directory number (DN) and for DN–to–EN translation. Also it consists of, features related to a particular subscriber, data to route the call based on the first 3 digits dialled, area code translation, international call translators etc.

Map of the switching network. There are two techniques for selection junctors.

1. **Map-in-memory.** In this technique, the memory contain a bit for each link. If it is set to 1 the link is free and if this bit is set to zero, the line is busy.

2. **Map-in-network.** In this technique, the junctor itself contains a one bit memory element, which is read by the path setup program to check whether it is free. The map-in-network consumes more time, but more advantages when several processors controlling the system.

SWITCHING SYSTEM SOFTWARE ORGANIZATION

In last section, three levels of controls of hardware architectures were discussed for a general digital switching system. For effective processing of a call, to perform various functions of subsystems and to interface with the other subsystems, software plays a vital role. The software programs enables any digital switching system input data, to give outputs in a fraction of seconds, concurrent processing of many calls in real time and performs many features other than simple path set between subscribers for conversation. In this section, the need for software, the software classification, basic software architecture, the involvement of software in various levels of hardware architecture, interfacing between subsystems through software and software presently used in various digital switching system are described.

Need for Software

Other than call processing, any exchange is to serve the subscriber various facilities and many administrative tasks. Fig. shows various activities of a switching system. To carry out these activities efficiently and effectively, the use of software is unavoidable.

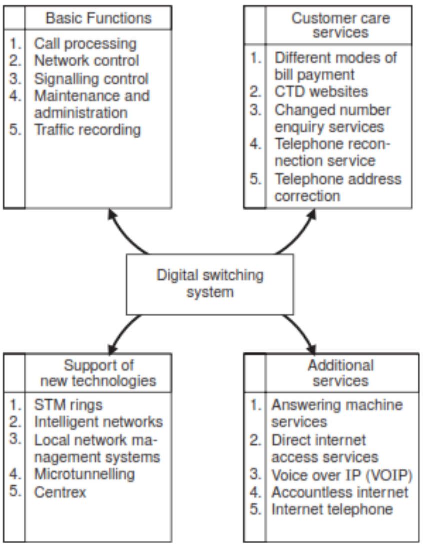


Fig. Activities of switching system.

To perform the above tasks, a large amount of software is required. However, the software for basic functions are must and remaining services are optional and requires software depends on the location of switching systems. Approximately 70% of the total software is used to perform basic functions. Only 0.1% of the total processing time is used by the 30% of the remaining service oriented software packages.

Software Classification and Interfacing

Classification. At various levels of hardware architecture, the software are used. Thus, many digital switching systems employ some system level software. Basic software systems are classified as:

1. Maintenance software

2. Call processing software

3. Database/Administration software

4. Feature software.

Above software packages are divided into program modules. Each module dealing with specific task. Several modules are grouped together to form functional units. Various factors are associated with the development of software product. These factors include the requirements of the business, the location of telephone exchanges, customer needs, internal requirements, and parameterised design. The parameterised design includes hardware parameter and software parameters. The hardware parameter are based on the hardware used in the central office or exchanges. They are number of network control processors, number of line controllers, number of subscribers to be serviced, number of trunks for which the exchange is engineered etc. Some examples of software parameters are the registers associated with number and size of automatic message accounting (AMA) registers, number and size of buffers for various telephony function and various features to be included for that particular exchanges. Thus, the parameterised design helps in designing software common to the similar types of exchanges.

Maintenance software

There are various activities and tests involved to maintain a switching system. Some of them are :

1. Supervision of the proper functioning of the exchange equipment, trunks and subscriber lines.

2. Monitoring the database of line and trunk assignments.

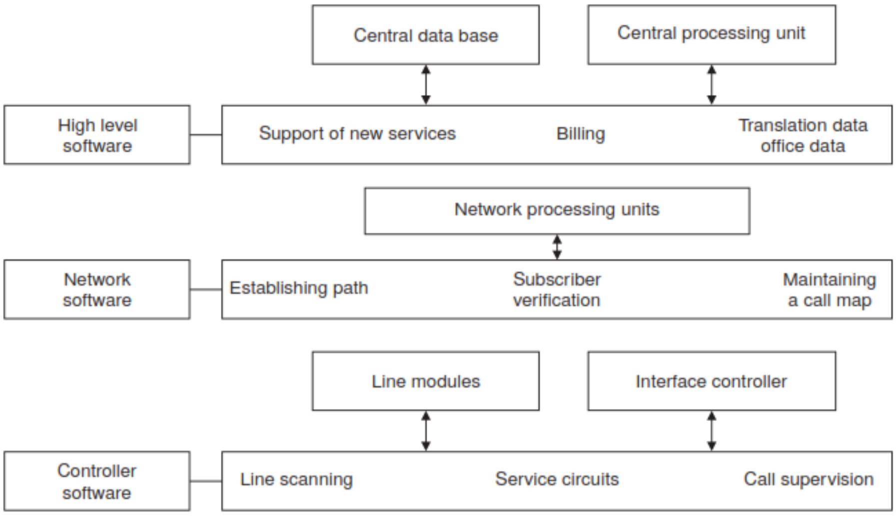
3. Efforts for the system recovery in case of failure.

4. Automatic line tests, which permits maintenance persons to attend several exchanges from one control location.

5. Effective diagnostic programs and maintenance strategies used to reduce the maintenance cost.

The root cause of the failure of any digital switching system is related to the software bugs which affects the memory and program loops, hardware failures, failure to identify the exact problem of failure and at least but not least the human error. Thus, the first step in software build is to select the appropriate program modules which is suitable for the switching system. The points to be considered are types of lines, location of switching system, signalling systems, availability of skilled person. Preventive maintenance programs are activated during the normal traffic. If a fault occurs, the OS activates the maintenance program to recover the system. Effective preventive and maintenance programs and strategies helps in proper maintenance of digital switching system with reduced maintenance cost.

Call processing software. The call processing functions are controlled by a central processor. Other functions carried out by the central processor are maintenance and administration, signalling, network control. Thus, the call processing programs are usually responsible for call processing and to interface with the translation data, office data, and automatic message accounting and maintenance programs. The translation data is the type of data generated by telephone companies related to subscriber. The office data is related to a particular digital switch. The call processing programs can be derived from state-transition diagrams in specification and description language (SDL). The SDL description in text form, is machine read and stored in memory in the form of data structures and linked lists and translation tables. An interpreter programs is written to access the lists and tables and to process the call by interpreting the data within them. Fig. shows three levels of call processing program. But it varies depends on the digital switching system.



Data base/Administration software

For administration and data base management, large amount of software required. But these tasks are performed infrequently, it uses less than 5% of the total processing time. The administration tasks includes

1. Alarm processing

2. Traffic recording

3. Change of numbers or area codes corresponding to the change in subscriber rate and Government policy.

4. Changing routing and routing codes. This decisions made on the traffic intensity of a particular exchange.

5. Generation of exchange management statistics.

Most digital switching system employ a data base system to:

1. Record office information

2. Billing information

3. Software and hardware parameters

4. System recovery parameters

5. System diagnostics.

Feature Software.

Most of the present day digital switching systems uses all packages.

Switching software. Software for digital switching systems are written in high level languages. Early electronic switching systems used assembly language programmes. In 1980, Plenary Assembly, CCITT approved the definition of a high level language as Recommendations–200. This language is known as CCITT high level language (CHILL). It has three major features as data structure, program structure and action statements. It is designed for the various SPC modules discussed earlier. Software codes for digital switching systems are also written in high level programming languages such as C, C ++, PASCAL.

Interfacing. The line control programs scan the status of lines and reports the status to the network status program. The network status programs works with network control programs. To provide dial tone, ringing, message to caller for invalid number, status of the subscriber and to receive dialled digits, and to clear signals from the subscriber, the line control programs interface with the network control programs. The call processing software which is responsible for call processing and in addition interfaces with accounting and maintenance programs for billing, recording and to identify the fault in lines. The call processing software also interfaces with feature programs to serve the customers need. The trunk modules interface different types of trunks to the digital switching system. Most digital switching systems employ special modules to connect ISDN and other digital services to the switch. Some specialized module interfaces are used to provide enhanced services such as **AIN** and packet switching.

ELECTRONIC EXCHANGES IN INDIA

Overview of Telecommunication Organizations

Department of Telecommunication (DOT) is the Government of India department under the ministry of communications. The main role of DOT in coordination with Telecom Regulatory Authority of India (TRAI) are Policy making, licencing and coordination relating to telegraphs, telephones, wireless, data, facsimile and telematics services. It also enforces wireless regulatory measures for wireless transmission by users in the country. The public sector companies under the ministry of communications which plays vital role in the telecommunications in India are

1. Bharat Sanchar Nigam Limited (BSNL)

2. Indian Telephone Industries Ltd (ITI)

3. Telecommunications consultants India (TCIL) Ltd

4. Mahanagar Telephone Nigam Limited (MTNL)

5. Videsh Sanchar Nigam Limited (VSNL)

6. Centre for development of telematics

The details of the BSNL which is the major telecom service provider and ITI, the leading telecom products manufacturer are given below in brief. For detailed information, the reader can refer the related websites. On October 1, 2000 the Department of Telecom operations, Govt. of India become a corporation and was christened Bharat Sanchar Nigam Limited (BSNL). Today, BSNL is the No. 1 telecommunication company and the largest public sector undertaking of India. It has a network of over 45 million lines covering more than 5000 towns and over 35 million telephone connections. The main functions of BSNL includes planning, engineering, installation, maintenance, management and operation of voice and non-voice telecommunication services all over the country. ITI established in 1948 is a Telecom company manufacturing the entire range of telecom equipment which includes telephones, large digital switches, and transmission systems like microwave, Fibre optic systems and satellite communication systems. Its highly satisfied customers in India include BSNL, MTNL, defence services, parliamentary, police and internal security organisations, railways etc. Many African and south Asian nations are its overseas customers. Related to digital switches, ITI in collaboration with ALCATEL, France manufactures large digital switches and with C-DOT India, manufactures small, medium and large digital switches. TCIL is a premier telecommunication consultancy and engineering company under the ministry of communications. TCIL with its number of joint venture Company's manufactures computer hardware, copper and optical fibre cables, developing software packages and providing consultancy and engineering services to other computer, information technology, telecom and software companies.

Switching Systems in India

ITI has contributed to 73% of the installed base of Public switching lines and two thirds of the installed base of large switches in India. ITI provides similar service support for these products outside India also, which will be cost effective. The indigenously develop switching systems used in India are:

• CDOT 256P RAX • CDOT TAX-XL

• CDOT SBM. • CDOT AN-RAX

• CDOT SBM-XL • CDOT RLC

• CDOT MAX-L • CDOT CNMS

• CDOT MAX-XL

The digital switching systems in collaboration with other countries are:

• OCB-283 M/S ITI, M/S Alcatel

• 5 ESS M/S lucent

• EWSD M/S HTL, M/S Siemens

• AXE-10 M/S Ericsson

• FETEX-150 M/S Fujitsu

• NEAX-61E M/S NEC

Module-III: TRAFFIC ENGINEERING

Traffic pattern, Grade of Service and blocking probability, modeling of switching systems: Markov Process, Birth-Death Process.

Telephone network organization: Network management, Network services, various networking plans, types of networks, Routing plan, International numbering plan, National numbering plan, Numbering plan in India, Signaling: in channel signaling, common channel signaling.

The telecommunication system has to service the voice traffic and data traffic. The traffic is defined as the occupancy of the server. The basic purpose of the traffic engineering is to determine the conditions under which adequate service is provided to subscribers while making economical use of the resources providing the service. The functions performed by the telecommunication network depends on the applications it handles. Some major functions are switching, routing, flow control, security, failure monitoring, traffic monitoring, accountability internetworking and network management. To perform the above functions, a telephone network is composed of variety of common equipment such as digit receivers, call processors, inter stage switching links and interoffice links etc. Thus traffic engineering provides the basis for analysis and design of telecommunication networks or model. It provides means to determine the quantum of common equipment required to provide a particular level of service for a given traffic pattern and volume. The developed model is capable to provide best accessibility and greater utilization of their lines and trunks. Also the design is to provide cost effectiveness of various sizes and configuration of networks. The traffic engineering also determines the ability of a telecom network to carry a given traffic at a particular loss probability. Traffic theory and queuing theory are used to estimate the probability of the occurrence of call blocking. Earlier traffic analysis based purely on analytical approach that involved advanced mathematical concepts and complicated operations research techniques. Present day approaches combine the advent of powerful and affordable software tools that aim to implement traffic engineering concepts and automate network engineering tasks. In the study of tele traffic engineering, to model a system and to analyse the change in traffic after designing, the static characteristics of an exchange should be studied. The incoming traffic undergoes variations in many ways. Due to peak hours, business hours, seasons, weekends, festival, location of exchange, tourism area etc., and the traffic is unpredictable and random in nature. So, the traffic pattern/characteristics of an exchange should be analysed for the system design. The grade of service and the blocking probability are also important parameters for the traffic study.

The following statistical information provides answer for the requirement of trunk circuits for a given volume of offered traffic and grade of service to interconnect the end offices. The statistical descriptions of a traffic is important for the analysis and design of any switching network.

1. **Calling rate.** This is the average number of requests for connection that are made per unit time. If the instant in time that a call request arises is a random variable, the calling rate may be stated as the probability that a call request will occur in a certain short interval of time. If '*n*' is the average number of calls to and from a terminal during a period T seconds, the calling rate is defined as $\lambda = n/T$

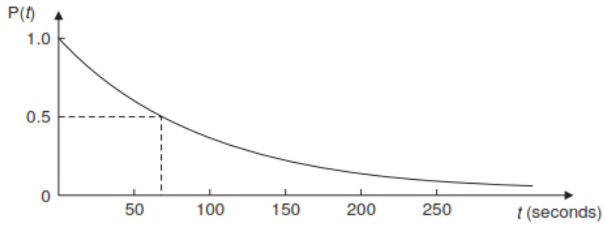
In telecommunication system, voice traffic and data traffic are the two types of traffic. The calling rate (λ) is also referred as average arrival rate. The average calling rate is measured in calls per hour.

2. **Holding time.** The average holding time or service time '*h*' is the average duration of occupancy of a traffic path by a call. For voice traffic, it is the average holding time per call in hours or 100 seconds and for data traffic, average transmission per message in seconds. The reciprocal of the average holding time referred to as service rate (μ) in calls per hour is given as $\mu = 1/h$

Sometimes, the statistical distribution of holding time is needed. The distribution leads to a convenient analytic equation. The most commonly used distribution is the negative exponential distribution. The probability of a call lasting at least *t* seconds is given by

$P(t) = \exp(-t/h)$

For a mean holding time of h = 100 seconds, the negative exponential distribution function is shown in Fig.



3. **Distribution of destinations.** Number of calls receiving at a exchange may be destined to its own exchange or remoted exchange or a foreign exchange. The destination distribution is

described as the probability of a call request being for particular destination. As the hierarchical structure of telecommunication network includes many intermediate exchanges, the knowledge of this parameter helps in determining the number of trunks needed between individual centres.

4. **User behaviour.** The statistical properties of the switching system are a function of the behaviour of users who encounter call blocking. The system behaves differently for different users. The user may abandon the request if his first attempt to make a call is failed. The user may makes repeated attempts to setup a call. Otherwise the user may wait some times to make next attempt to setup a call. These behaviour varies person to person and also depends on the situation.

5. **Average occupancy.** If the average number of calls to and from a terminal during a period T seconds is '*n*' and the average holding time is '*h*' seconds, the average occupancy of the terminal is given by

$$A = nh/T = \lambda h = \lambda/\mu$$

Thus, average occupancy is the ratio of average arrival rate to the average service rate. It is measured in Erlangs. Average occupancy is also referred as traffic flow or traffic intensity or carried traffic.

Traffic Pattern

An understanding of the nature of telephone traffic and its distribution with respect to time (traffic load) which is normally 24 hours is essential. It helps in determining the amount of lines required to serve the subscriber needs. According to the needs of telephone subscribers, the telephone traffic varies greatly. The variations are not uniform and varies season to season, month to month, day to day and hour to hour. But the degree of hourly variations is greater than that of any other period.

Various parameters related to traffic pattern are discussed below:

Busy hour. Traditionally, a telecommunication facility is engineered on the intensity of traffic during the busy hour in the busy session. The busy hour vary from exchange to exchange, month to month and day to day and even season to season. The busy hour can be defined in a variety of ways. In general, the busy hour is defined as the 60 minutes interval in a day, in which the traffic is the highest. Taking into account the fluctuations in traffic, CCITT in its recommendations E.600 defined the busy hour as follows.

1. **Busy hour.** Continuous 60 minutes interval for which the traffic volume or the number of call attempts is greatest.

2. **Peak busy hour.** It is the busy hour each day varies from day to day, over a number 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 number of call attempts is greatest over the days under consideration. In order to simplify the traffic measurement, the busy hour always commences on the hour, half hour, or quarter hour and is the busiest of such hours. The busy hour can also be expressed as a percentage (usually between 10 and 15%) of the traffic occurring in a 24 hour period.

Call completion rate (CCR). Based on the status of the called subscriber or the design of switching system the call attempted may be successful or not. The call completion rate is defined as the ratio of the number of successful calls to the number of call attempts. A CCR value of 0.75 is considered excellent and 0.70 is usually expected.

Busy hour call attempts. It is an important parameter in deciding the processing capacity of an exchange. It is defined as the number of call attempts in a busy hour.

Busy hour calling rate. It is a useful parameter in designing a local office to handle the peak hour traffic. It is defined as the average number of calls originated by a subscriber during the busy hour.

Day-to-day hour traffic ratio. It is defined as the ratio of busy hour calling rate to the average calling rate for that day. It is normally 6 or 7 for rural areas and over 20 for city exchanges.

Units of Telephone Traffic

Traffic intensity is measured in two ways. They are (*a*) Erlangs and (*b*) Cent call seconds (CCS).

Erlangs. The international unit of traffic is the Erlangs. It is named after the Danish Mathematician, A.K. Erlang, who laid the foundation to traffic theory in the work he did for the Copenhagen telephone company starting 1908. A server is said to have 1 erlang of traffic if it is occupied for the entire period of observation. More simply, one erlang represents one circuit occupied for one hour.

The maximum capacity of a single server (or channel) is 1 erlang (server is always busy). Thus the maximum capacity in erlangs of a group of servers is merely equal to the number of servers.

For example problems on traffic engg. Please refer text books.

Cent call seconds (CCS). It is also referred as hundred call seconds. CCS as a measure of traffic intensity is valid only in telephone circuits. CCS represents a call time product. This is used as a measure of the amount of traffic expressed in units of 100 seconds. Sometimes call seconds (CS) and call minutes (CM) are also used as a measure of traffic intensity. The relation between erlang and CCS is given by

1E = 36 CCS = 3600 CS = 60 CM

Grade of Service (GOS)

For non-blocking service of an exchange, it is necessary to provide as many lines as there are subscribers. But it is not economical. So, some calls have to be rejected and retried when the lines are being used by other subscribers. The grade of service refers to the proportion of unsuccessful calls relative to the total number of calls. GOS is defined as the ratio of lost traffic to offered traffic.

$$\text{GOS} = \frac{\text{Blocked Busy Hour calls}}{\text{Offered Busy Hour calls}}$$

$$\text{GOS} = \frac{A - A_0}{A}$$

where A0 = carried traffic

A = offered traffic

A – A0 = lost traffic.

The smaller the value of grade of service, the better is the service. The recommended GOS is 0.002, *i.e.* 2 call per 1000 offered may lost. In a system, with equal no. of servers and subscribers, GOS is equal to zero. GOS is applied to a terminal to terminal connection. But usually a switching centre is broken into following components

(*a*) an internal call (subscriber to switching office)

(*b*) an outgoing call to the trunk network (switching office to trunk)

(*c*) the trunk network (trunk to trunk)

(*d*) a terminating call (switching office to subscriber).

The GOS calculated for each component is called component GOS. The overall GOS is in fact approximately the sum of the component grade of service.

There are two possibilities of call blocking. They are (*a*) Lost system and (*b*) Waiting system. In lost system, a suitable GOS is a percentage of calls which are lost because no equipment is available at the instant of call request. In waiting system, a GOS objective could be either the percentage of calls which are delayed or the percentage which are delayed more than a certain length of time.

Blocking Probability and Congestion

The value of the blocking probability is one aspect of the telephone company's grade of service. The basic difference between GOS and blocking probability is that GOS is a measure from subscriber point of view whereas the blocking probability is a measure from the network or switching point of view. Based on the number of rejected calls, GOS is calculated, whereas by observing the busy servers in the switching system, blocking probability will be calculated. The blocking probabilities can be evaluated by using various techniques. Lee graphs and Jacobaeus methods are popular and accurate methods. The blocking probability B is defined as the probability that all the servers in a system are busy. Congestion theory deals with the probability that the offered traffic load exceeds some value. Thus, during congestion, no new calls can be accepted. There are two ways of specifying congestion. They are time congestion and call congestion. Time congestion is the percentage of time that all servers in a group are busy. The call or demand congestion is the proportion of calls arising that do not find a free server. In general GOS is called call congestion or loss probability and the blocking probability is called time congestion. If the number of sources is equal to the number of servers, the time congestion is finite, but the call congestion is zero. When the number of sources is large, the probability of a new call arising is independent of the number already in progress and therefore the call congestion is equal to time congestion.

MODELLING OF TRAFFIC

To analyse the statistical characteristics of a switching system, traffic flow and service time, itis necessary to have a mathematical model of the traffic offered to telecommunication systems. The model is a mathematical expression of physical quantity to represents the behaviour of the quantity under consideration. Also the model provides an analytical solutions to a tele traffic problems. As the switching system may be represented in different ways, different models are possible. Depending on the particular system and particular circumstance, a suitable model can be selected. In practice, the facilities of the switching systems are shared by many users. This arrangement may introduce the possibility of call setup inability due to lack of available facilities. Also in data transfer, a system has to buffer message while waiting for transmission. Here size of the buffer depends on traffic flow. As serving the number of subscribers subject to fluctuation (due to random generation of subscriber calls, variations in holding time, location of the exchange, limitation in servers etc.), modelling of traffic is studied using the concepts and methods of the theory of probability. If a subscriber finds no available server for his call attempt, he will wait in a line (queue) or leave immediately. This phenomenon may be regarded as a queueing system. The mathematical description of the queueing system characteristics is called a queueing model.

The random process may be discrete or continuous. Similarly the time index of random variables can be discrete or continuous. Thus, there are four different types of process namely (*a*) continuous time continuous state (*b*) continuous time discrete state (*c*) discrete time continuous state and (*d*) discrete time discrete state. In telecommunication switching system, our interest is discrete random process and therefore for modelling a switching system, we use discrete state stochastic process. A discrete state stochastic process is often called a chain. A statistical properties of a random process may be obtained in two ways:

(*i*) Observing the behaviour of the system to be modelled over a period of time repeatedly. The data obtained is called a single sample. The average determined by measurements on a single sample function at successive times will yield a **time average**.

(*ii*) Simultaneous measurements of the output of a large number of statistically identical random sources. Such a collection of sources is called an **ensemble** and the individual noise waveforms is called the **sample function**. The statistical average made at some fixed time $t = t1$ on all the sample functions of the ensample is the **ensemble average.**

The above two ways are analogous to obtaining the statistics from tossing a die repeatedly (large number) or tossing one time the large number of dice. In general, time average and ensemble average are not the same due to various reasons. When the statistical characteristics of the sample functions do not change with time, the random process is described as being **stationary**. The random process which have identical time and ensemble average are known as **ergodic processes.** An ergodic process is stationary, but a stationary process is not necessarily ergodic.

Telephone traffic is nonstationary. But the traffic obtained during busy hour may be considered as stationary (which is important for modelling) as modelling non-stationary is difficult.

Pure Chance Traffic

Here, the call arrivals and call terminations are independent random events. If call arrivals are independent random events, their occurrence is not affected by previous calls. This traffic is therefore sometimes called **memory less traffic.** A.A. Markov in 1907, defined properties and proposed a simple and highly useful form of dependency. This class of processes is of great interest to our modelling of switching systems. A discrete time Markov chain *i.e.* discrete time discrete state Markov process is defined as one which has the following property.

$$P\,[\{X(t_{n+1}) = x_{n+1}\}/\{X(t_n) = x_n,\ X(t_{n-1}) = x_{n-1},\ \ldots\ldots\ X(t_1) = x_1\}]$$
$$= P\,[\{X(t_{n+1}) = x_{n+1}\}/\{X(t_n) = x_n\}]$$

where $t_1 < t_2 \ldots\ldots < t_n < t_{n+1}$ and x_i is the ith discrete state space value.

Equation states that the probability that the random variable X takes on the value x_{n+1}

at time step $n + 1$ is entirely determined by its state value in the previous time step n and is independent of its state values in earlier time steps ; $n - 1$, $n - 2$, $n - 3$ etc.

The Birth and Death Process

The birth and death process is a special case of the discrete state continuous time Markov process, which is often called a continuous-time Markov chain. The number of calls in progress is always between 0 and N. It thus has N + 1 states. If the Markov chain can occur only to adjacent states (*i.e.* probability change from each state to the one above and one below it) the process is known as birth-death (B–D) process. The basic feature of the method of Markov chains is the kolmogorov differential-difference equation, for the limiting case, can provide a solution to the state probability distribution for the Erlang systems and Engest systems.

Let N(t) be a random variable specifying the size of the population at time t. For a complete description of a birth and death process, we assume that N(t) is in state k at time t and has the following properties:

1. P(k) is the probability of state k and P(k + 1) is the probability of state k + 1.

2. The probability of transition from state k to state k + 1 in short duration Δt is $\lambda\,\Delta t$, where λ is called the birth rate in state k.

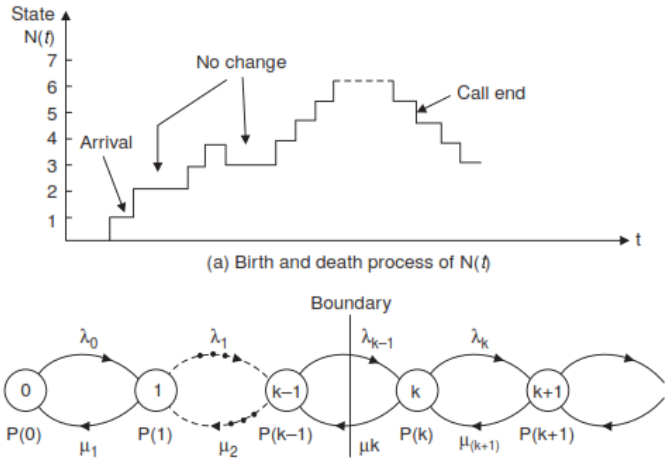
3. The probability of transition from state k to state $k - 1$ in the time interval Δt is $\mu k\ \Delta t$, where μ is called the death rate in state k.

4. The probability of no change of state in the time interval Δt is equal to $1 - (\lambda k)\ \Delta t$.

5. The probability in Δt, from state k to a state other than $k + 1$ or $k - 1$ is zero. Based on the above properties, birth and death process of N(t) and state transition rate diagram are shown in Fig. At statistical equilibrium (*i.e.* stationary), let P_{jk} is the conditional probability, that is the probability of state increases from j to k. Similarly P_{kj} is the probability of state decrease from k to j.

The probabilities P(0), P(1), ... P(N) are called the **state probabilities** and the conditional probabilities P_{jk}, P_{kj} are called **transition probabilities.** The transition probabilities satisfy the following condition:

$$P_{jk}(t) \geq 0, \sum_{k=0}^{\infty} P_{jk}(t) = 1$$



(a) Birth and death process of N(t)

(b) State transition diagram

Numbered circles denote the state of the process at t

Markov theorem states that for any Markov process characterized by the transition probability Pjk, the limit

$$\lim_{t \to \infty} P_{jk} = P(k)$$

exist and does not depend on j and the probability P(k).

According to Markov's,

$$\frac{d}{dt} P(k) = -(\lambda_k + \mu_k) P(k) + \lambda_{k-1} P(k-1) + \mu_{k+1} P(k+1)$$
$$k = 0, 1, 2, ..., \text{ with } \lambda_{-1} = \mu_0 = P_{-1} = 0$$

This set of differential-difference equations represents the dynamic behaviour of the birth and death process. As $t \rightarrow \infty$,

$$-(\lambda_k + \mu_k) P(k) + \lambda_{k-1} P(k-1) + \mu_{k+1} P(k+1) = 0$$

with $\lambda - 1 = \mu_0 = P_{-1} = 0$.

This set of equations together with the normalization condition uniquely determines the required

$$\sum_{k=0}^{\infty} P(k) = 1$$

and state probabilities P(k) as $$P(k) = \frac{\lambda_{k-1}}{\mu_k} P(k-1)$$

The probability P(0) can be determined by the equation

$$P(k) = \frac{\lambda_0 \lambda_1 \ldots\ldots \lambda_{k-1}}{\mu_1 \mu_2 \ldots\ldots \mu_k} P(0), k = 1, 2, 3, \ldots\ldots \qquad P(0) = \left[1 + \sum_{k=1}^{\infty} \frac{\lambda_0 \lambda_1 \ldots\ldots \lambda_{k-1}}{\mu_1 \mu_2 \ldots\ldots \mu_k}\right]^{-1}$$

$$P(k) = \frac{\lambda_{k-1} P(k-1)}{\mu_k}$$

LOSS SYSTEMS

The service of incoming calls depends on the number of lines. If number of lines equal to the number of subscribers, there is no question of traffic analysis. But it is not only uneconomical but not possible also. So, if the incoming calls finds all available lines busy, the call is said to be **blocked.** The blocked calls can be handled in two ways. The type of system by which a blocked call is simply refused and is lost is called **loss system.** Most notably, traditional analog telephone systems simply block calls from entering the system, if no line available. Modern telephone networks can statistically multiplex calls or even packetize for lower blocking at the cost of delay. In the case of data networks, if dedicated buffer and lines are not available, they block calls from entering the system. In the second type of system, a blocked call remains in the system and waits for a free line. This type of system is known as **delay system.** These two types differs in network, way of obtaining solution for the problem and GOS.

For loss system, the GOS is probability of blocking. For delay system, GOS is the probability of waiting.

Erlang determined the GOS of loss systems having N trunks, with offered traffic A, with the following assumptions. (*a*) Pure chance traffic (*b*) Statistical equilibrium (*c*) Full availability and (*d*) Calls which encounter congestion are lost. The first two are explained in previous section. A system with a collection of lines is said to be a fully-accessible system, if all the lines are equally accessible to all in arriving calls. For example, the trunk lines for interoffice calls are fully accessible lines. The lost call assumption implies that any attempted call which encounters congestion is immediately cleared from the system. In such a case, the user may try again and it may cause more traffic during busy hour. The Erlang loss system may be defined by the following specifications.

1. The arrival process of calls is assumed to be Poisson with a rate of λ calls per hour.

2. The holding times are assumed to be mutually independent and identically distributed random variables following an exponential distribution with $1/\mu$ seconds.

3. Calls are served in the order of arrival.

There are three models of loss systems. They are:

1. Lost calls cleared (LCC)

2. Lost calls returned (LCR)

3. Lost calls held (LCH)

Lost Calls Cleared (LCC) System

The LCC model assumes that, the subscriber who does not avail the service, hangs up the call, and tries later. The next attempt is assumed as a new call. Hence, the call is said to be cleared. This also referred as blocked calls lost assumption. The first person to account fully and accurately for the effect of cleared calls in the calculation of blocking probabilities was A.K.Erlang in 1917. Consider the Erlang loss system with N fully accessible lines and exponential holding times. The Erlang loss system can be modelled by birth and death process with birth and death rate as follows.

$$\lambda_k = \begin{cases} \lambda, k = 0, 1, \ldots\ldots N-1 \\ 0 \quad k \geq N \end{cases}$$

$$\mu_k = \begin{cases} k\mu, k = 0, 1, \ldots\ldots N \\ 0, \quad k > N \end{cases}$$

$$P(k) = \frac{\lambda_0 \lambda_1 \ldots\ldots \lambda_{k-1}}{\mu_1 \mu_2 \ldots\ldots \mu_k} P(0), k = 1, 2, 3$$

Substituting in the above equation, we get

$$P(k) = \frac{1}{k!}\left(\frac{\lambda}{\mu}\right)^k P(0), k = 1, 2, 3, \ldots\ldots N$$

From equation the offered traffic is

$$A = \frac{\lambda}{\mu}$$

Substituting we get

$$P(k) = \frac{1}{k!}(A)^k P(0), k = 1, 2, 3, \ldots\ldots N$$

The probability P(0) is determined by the normalization condition

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

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

Substituting we get

$$P(k) = \frac{A^k / k!}{\sum_{k=0}^{N} \frac{A^k}{k!}}$$

The probability distribution is called the truncated **Poisson distribution** or **Erlang's loss distribution.** In particular when k = N, the probability of loss is given by

$$P(N) = B(N, A) = \frac{A^N}{N! \sum_{k=0}^{N} \left(\frac{A^k}{k!}\right)}$$

where A = λ/μ .

This result is variously referred to as **Erlang's formula of the first kind, the Erlang's-B formula or Erlang's loss formula.**

Equation specifies the probability of blocking for a system with random arrivals from an infinite source and arbitrary holding time distributions. The Erlang B formula gives the time congestion of the system and relates the probability of blocking to the offered traffic and the number of trunk lines. Values from B(N, A) obtained from equation have been plotted against the offered traffic 'A' Erlang's for different values of the number of N lines in Fig. In design problems, it is necessary to find the number of trunk lines needed for a given offered traffic and a specified grade of service.

For example problems please refer text books.

A' = A [1 – B (N, A)]

Thus, the carried load is the position of the offered load that is not lost from the system. The carried load per line is known as the trunk occupancy.

$$\rho = \frac{A'}{N} = \frac{A(1-B)}{N}$$

The trunk occupancy ρ is a measure of the degree of utilization of a group of lines and is sometimes called the utilization factor.

Example 8.7. *A group of 7 trunks is offered 4E of traffic, find (a) the grade of service (b) the probability that only one trunk is busy (c) the probability that only one trunk is free and (d) the probability that at least one trunk is free.*

Sol. Given data : N = 7, A = 4E

From equation 8.54,

(*a*)
$$B(7, 4) = \frac{4^7}{7!\left[1+\frac{4}{1}+\frac{4^2}{2!}+\frac{4^3}{3!}+\frac{4^4}{4!}+\frac{4^5}{5!}+\frac{4^6}{6!}+\frac{4^7}{7!}\right]}$$

$$= \frac{16384}{5040\,[1+4+8+21.3+10.6+8.5+5.7+3.25]}$$

$$B = 0.052 = \text{GOS}.$$

(*b*) The probability of only one trunk is busy

$$P(k) = \frac{A^k / k!}{\sum_{k=0}^{N} (A^k / k!)}$$

For $k = 1$
$$P(1) = \frac{4/1!}{62.35} = 0.064$$

(*c*) The probability that only one trunk is free

$$P(6) = \frac{4^6/6!}{62.35} = \frac{5.68}{62.35} = 0.0912$$

(*d*) The probability that at least one trunk is free

$$P(k < 7) = 1 - P(7) = 1 - B = 1 - 0.052 = 0.948.$$

Lost Calls Returned (LCR) System

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 arising 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 time's λ 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' is given as

$$\lambda' = \lambda + B\lambda + B^2\lambda + B^3\lambda + \ldots\ldots$$
$$\lambda' = \frac{\lambda}{1-B}$$

where B is the blocking probability from a lost calls cleared (LCC) analysis.

The effect of returning traffic is insignificant when operating at low blocking probabilities. At high blocking probabilities, it is necessary to incorporate the effects of the returning traffic into analysis.

Lost Calls Held (LCH) System

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 is 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.

DELAY SYSTEMS

The delay system places the call or message arrivals in a queue if it finds all N servers (or lines) occupied. This system delays non-serviceable requests until the necessary facilities become available. These systems are variously referred to as delay system, waiting-call systems and queueing systems. The delay systems are analysed using queueing theory which is sometimes known as waiting line theory. This delay system have wide applications outside the telecommunications. Some of the more common applications are data processing, supermarket checkout counters, aircraft landings, inventory control and various forms of services. Consider that there are k calls (in service and waiting) in the system and N lines to serve the calls. If $k = N$, k lines are occupied and no calls are waiting. If $k > N$, all N lines are occupied and $k - N$ calls waiting. Hence a delay operation allows for greater utilization of servers than does a loss system. Even though arrivals to the system are random, the servers see a somewhat regular arrival pattern. A queueing model for the Erlang delay system is shown in Fig.

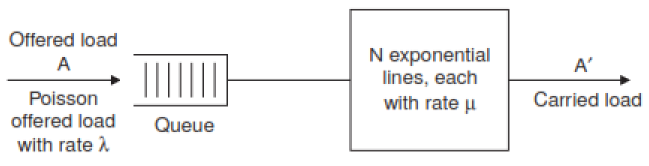


Fig. Queueing model

The basic purpose of the investigation of delay system is to determine the probability distribution of waiting times. From this, the average waiting time W as random variable can be easily determined. The waiting times are dependent on the following factors:

1. Number of sources

2. Number of servers

3. Intensity and probabilistic nature of the offered traffic

4. Distribution of service times

5. Service discipline of the queue.

In a delay system, there may be a finite number of sources in a physical sense but an infinite number of sources in an operational sense because each source may have an arbitrary number of requests outstanding. If the offered traffic intensity is less than the servers, no statistical limit exists on the arrival of calls in a short period of time. In practice, only finite queue can be realised. There are two service time distributions. They are constant service times and exponential service times. With constant service times, the service time is deterministic and with exponential, it is random. The service discipline of the que involves two important factors.

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.

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

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

Under equilibrium conditions, the state probability distribution P(k) can be obtained by substituting these birth rates into the following equation

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

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,

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

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

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

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

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

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

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

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

We know

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

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

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

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

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

where B = Blocking probability for a LCC system

N = Number of servers

A = Offered load (Erlangs)

Equation above are referred as **Erlang's second formula, Erlang's delay formula or Erlang's C formula.**

For single server systems (N = 1), the probability of delay reduces to ρ, which is simply the output utilization or traffic carried by the server. Thus the probability of delay for a single server system is also.

TELEPHONE NETWORK ORGANIZATION

A telecommunication network contains a large number of links joining different locations, which are known as the nodes of the network. These nodes may be end instruments (subscriber nodes), switching centres (switching nodes), networks providing just a link between nodes (transmission nodes) or service nodes (which provides service on demand such a voice mail boxes, stock market price announcement, sports results etc.). To provide efficient communication, a telephone network should include various transmission system (for example, terestial, microwave, optical satellite communications), switching system (to identity and connect calling and called subscriber) and to exchange information between subscriber and switching systems or between interexchange, a good signalling system required. The calling and called subscriber should be connected almost instantly. So, as an identification, a numbering system is introduced and it varies region to region and country to country. Telephone networks require certain form of procedure to route a particular call to the destination for effective and cost effective communication. So, the telephone network should be implemented with a good routing plan. An establishment of an exchange includes heavy expenses on switching equipment, establishing trunks and links, buildings, infrastructure, human resources to handle the exchange etc. These capital lost and the day-to-day expenses must be met by the exchanges through its subscribers. So, the billing and charging the subscriber calls or data transfer is a vital part of the network. Also, introducing a new exchange, extension of the existing exchanges, upgradation of the facilities and speed up the switching, changing the sales strategy based on the competition, addition of new services, management of maintenance, providing employment to the skilled peoples etc., are based on the government policies or the telephone company's business strategies. Thus, the functions of telecommunication networks is limit less and network management is an important part of any telecommunication network organization.

NETWORK MANAGEMENT

The basic goal of the network management is to maintain efficient operations during equipment failures and traffic overloads. Also controlling the flow of call requests during network overload is a vital function of network management. For the effective network management the study of various services provided by the network, offered load of the network, classification of the network based on services offered, interconnection of different types of networks and network planning is important. Based on the data available for the above factors, the network management has become updated.

Network Planning

The planning of telecommunication system comprising a network of switching centres includes various plans. From initiating the network to the extension of the network based on the increased load, network planning plays a vital role.

Network services: The capabilities often collectively referred to "as intelligence" within the network are listed below. Depending upon the applications the network is to handle the interconnectivity with other networks. The following functions and its essential parts are included in the network. Various services are:

1. **Switching.** The process of interconnecting incoming calls or data to the appropriate outgoing channel called destination is referred switching.

2. **Routing.** The ability of the network to select a path to connect calling and called subscriber for telephone conversations or providing path for data transfer between source and destination is referred as routing. The network generally choses a path and sometimes user may specify it.

3. **Flow control.** It is the ability of a network to reject traffic. Managing the rate at which traffic enters a network is referred to as flow control. A network without effective flow control procedures becomes very inefficient.

4. **Security.** There are two ways of providing security of the network. First, to increase the security of operation in presence of faults. To provide adequate security, the complete network may be duplicated or triplicated. Second, preventing unauthorized access to the network and the data it carries. This may be achieved by pass words, data encryption and providing limiting factors in accessing the network.

5. **Signalling.** A signalling system link the variety of switching system, transmission system and subscriber equipment in a telecommunication network to enable the network to function as a whole.

6. **Traffic management.** The ability of the network to keep track of traffic levels is referred as traffic management. Traffic management is useful both in short term and long term bases. On a short term basis, it can be used to support dynamic routing and flow control. Over a long term it can be used in network design to identify parts of the network where capacity may be productively increased or decreased.

7. **Accountability.** This includes charging, billing, accounting and inventory control. This is the ability of the network to track the users of the network.

8. **Administration.** It is related to the ability of the network to identify the load of a network and providing corresponding upgradation of parts, extension of networks facility. It also identities the sales strategy, investment planning etc.

9. **Inter-networking.** It is the ability of the network to perform the functions needed to communicate with and across other networks. This includes providing routes for traffic crossing through, into and out of the network, and allocating resources such as buffers and link capacity to traffic originating other networks.

Network levels: All the above functions and some other service related functions are usually classified or grouped into different levels. This grouping ease the network and the concerned network engineers to carry out the functions efficiently. The grouping differs from network to network or divisions to divisions or between telephone companies. The ability of the network to provide these functions has a profound effect on the type of network. In particular, the nodes of the network become more complex and more expensive as their functionality increases. Fig. shows the different levels of the network management.

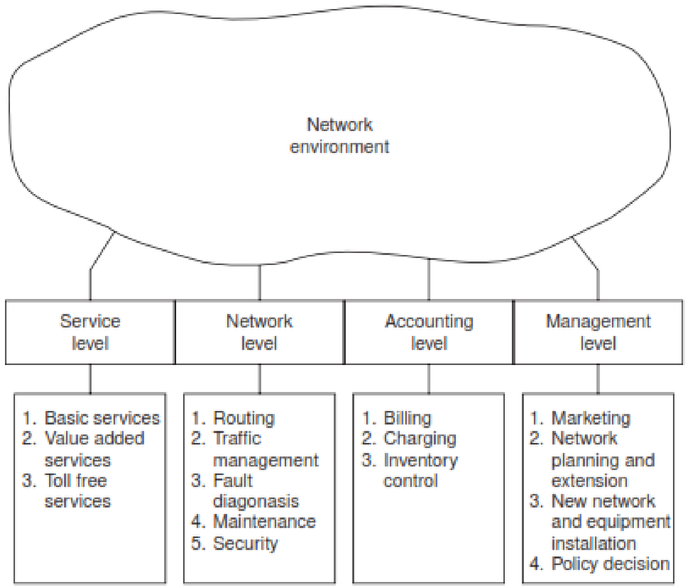


Fig. Different levels of network management.

Various networking plans: A national telecommunication network is large and complex. Therefore certain plans are needed to govern the design of network. The plans are independent and are affected by the predicted (or planned) growth rate of the telecommunication system. More specific network planning are :

1. Routing plans

2. Numbering plans

3. Charging plan

4. Transmission plan

5. Signalling plan

6. Grade of service

7. Network control and network administration.

The choice of a plan for a telecommunication system generally involves comparison of the economics of various possible plans. It also involves comparison of the economy of various possible plans and involves a certain amount of human judgement.

Types of Networks

There are in general three networks which can be used for any services (voice or data transfer).

They are:

(*a*) **Public switched network.** It allows access to the end office, connects through the long-distance network, and delivers to the end point. There are many hierarchies depends on the wish of network provider. But, the goal of the network hierarchies is to complete the call in the least amount of time and the shortest route possible.

(*b*) **Private networks.** Many companies, depending on their size and need, create or build their own networks. If their networks are underutilised, they may give their network for hire or lease. These networks employ mixture of technologies.

(*c*) **Hybrid networks.** To provide a service, if an organisation uses both private and public networks, the network is referred as hybrid network. Normally, the high-end usage services are connected via private facilities, the lower volume locations use the switched network.

This usually works out better financially for the organisation because the costs can be fully justified on a location by location basis. As the private line networks are designed to suit integration of voice, data, video graphics and facsimile transmission, pressure is more on it.

The customers are generally in need of variety of services. Even though certain networks are used to perform many services, they are more effective for a particular one or two services.

Hence, based on the services, the networks are classified as

1. The Public Switched telephone Network (PSTN)—for telephony.

2. The public switched telegraph network—for telex

3. Data networks—for voice and data

4. Cellular radio network—for mobile communication

5. Special service networks—to meet specialised demands.

As the above networks may be used to perform mixture of services, many authors categorised the networks into only three classes. They are generally considered as major telecommunication networks.

1. PSTN or POTS

2. Data networks

3. ISDN.

ROUTING PLAN

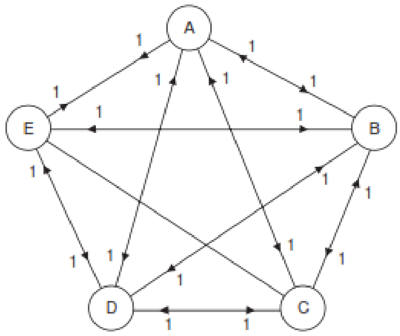
Routing planning refers to the procedures that determine which path in a network are assigned to particular connections. The switching centres may use fixed routes to each destination. Adaptive routing may be employed in which each exchange may use different routes for the same destination, depending upon traffic conditions. For effective routing of a call, some form of interconnection of switching exchanges are required. In the following paragraph various forms of networking and its related routing are discussed.

9.3.1. Basic Topologies

Three basic topologies are adopted for interconnecting exchanges. Exchanges are interconnected by group of trunk lines referred as trunk groups. Two trunk groups are required between any two exchanges. Mesh, star and mixed or hierarchical are the three basic topologies.

Determination at the total number of trunk circuits in any network is necessarily a function of the amount of traffic between each pair of stations or exchanges.

Mesh-connected network. This is also called fully connected topology. The advantage of mesh network is that each station has a dedicated connection to other stations. Therefore, this topology offers the highest reliability and security. If one link in the mesh topology breaks, the network remains active. A major or disadvantage of this topology is that it uses too many connections and therefore requires great deal of wiring, especially when the number of station increases. The mesh topology requires $N (N – 1)/2$ connections. For 100 stations, 4950 links required. Fig. depicts a full connected mesh structure.

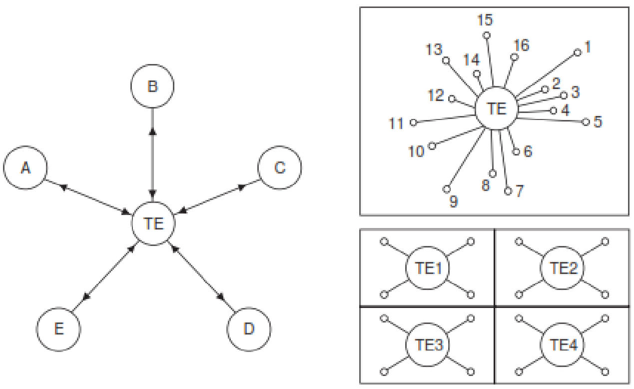


Mesh connected network.

For high traffic networks, the cost of network increases. As a compromise, generally, the circuits are divided into three groups. Two groups are used for one way trunks and third group is for both way trunks. An exchange first finds its outgoing trunk before trying one in common group.

This arrangement is practicable if number of exchanges in the network are limited and placed nearby (*i.e.* the lines are short). As the telephone network users are increasing at rapid rate, the concept of mesh network is uneconomical and the technological growth of transmission of signal by multiplexing made this technique to become completely obsolete.

Star topology. It is an alternative to the mesh arrangement. The network configuration shown in Fig. (*a*) is called star network. In star network, the number of lines is equal to the number of stations. As shown, a star connection utilises an intermediate exchange called a tandom exchange. Through the tandom exchange (TE) all other exchanges communicate.



(a) Star network (b) Area with single TE and multi TE

If the number of stations served by a TE increases, they are divided into smaller network, each served by its TE. Fig. (*b*) shows the star network with splitted setup. This configuration reduces the line cost but increases the exchange costs. With the star arrangement the outer centres require fewer trunk terminations, but the trunk centre has greatly increased number of terminations. So, the star arrangement reduces the design requirements on all but one of the switching centres. It needs larger, and more powerful trunk centre. As only one larger centre is required, star arrangement preferable. Note that the exchange area indicates that all the calls in that area are considered to be local calls.

Hierarchical networks. Many star networks may be inter connected by using an additional tandom exchange, leading to two level star network. An orderly construction of multilevel star networks leads to hierarchical networks. Fig. shows the two types of hierarchical structures of AT & T and ITU–T. Hierarchical networks are capable of handling heavy traffic with minimal number of trunk groups. The hierarchical network requires more switching nodes, but achieves significant savings in the number of trunks. Determination of the total number of trunk circuits in entire network is necessarily a function of the amount of traffic between each pair of switching nodes. The efficiency of circuit utilization is the basic motivation for hierarchical switching structures. In Fig. it is shown that, if there is a high traffic intensity between any pair of exchanges, direct trunk groups may be established between any pair of exchanges (dotted lines or trunks of AT & T hierarchical network). These direct routes are known as high usage routes or trunks. In a strictly hierarchical network, traffic from subscriber A to B and vice versa flows through the highest level of hierarchy. A traffic route via the highest level of hierarchy is known as the final route. Whenever high usage route exists, route is primarily routed through them. The overflow traffic is routed through hierarchical network. Traffic is always routed through the lowest available level of the network. In addition to the high usage trunks, the tandom switches which is employed at the lowest level (not part of toll network) is augmented. The term tandom refers specifically to intermediate switching within the exchange area. The exchange area is an area within which all calls are considered to be local calls. The basic function of a tandom office is to interconnect those central offices (or class 5 or local exchanges) within an exchange area having insufficient interoffice traffic volumes to justify direct trunks. Tandom exchanges also provide alternate routes for exchange area call get blocked on direct routes between end offices.

NUMBERING PLAN

The numbering plan is used to identify the subscribers connected in a telecommunication network. The main objective of numbering plan by any nation is to standardise the number length wherever practical according to CCITT recommendations. Other objectives includes (*a*) to meet the challenges of the changing telecom environment (*b*) to meet subscriber needs for a meaningful and user friendly scheme (*c*) to reserve numbering capacity to meet the undefined future needs.

9.4.1. ITU Recommendations in Numbering

Some important recommendations of ITU are described below:

Recommendation E.164: It provides the number structure and functionality for three categories of numbers used for international public telecommunication. The three categories of numbers are:

1. **National telephone services.** An international public telecommunication number (for geographic areas) is also referred to as the national significant number (NSN). NSN consists of the country code (CC), national destination code (NDC) and the subscriber number (SN).

2. **Global telephone services.** An international public telecommunication number for global telephone service consists of a three digit country code and global subscriber number.

The country code is always in the 8XX or 9XX range.

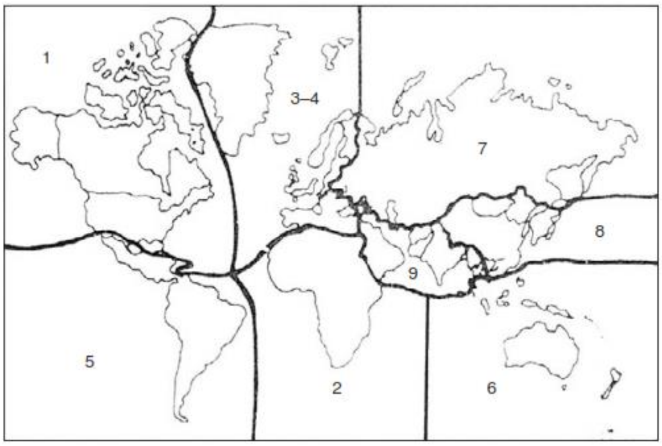
3. **International networks.** An international public telecommunication number for international networks consists of three digit country code, a network identification code and a subscriber number. The country code is always in the 8XX range. The identification code is one to four digits.

Recommendation E.123. This defines a standard way to write telephone numbers, email addresses and web addresses. It recommends how to use hyphen (-), space ( ), or period (•). ( ) are used to indicate digits that are sometimes not dialled, / is used to indicate alternate numbers and • is used in web addresses.

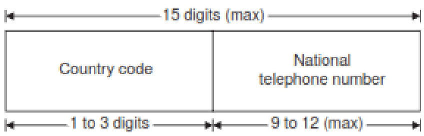
Recommendation E.162. This recommendation describes that the originating country must analyse a maximum of seven digits of the E.164 international number. When a number is being analysed, it will be done according to this recommendations. Also, the international numbering plan or world numbering plan has been defined in recommendations E.160; E.161 and E.162.

International Numbering Plan

This plan has to be implemented irrespective of a country's national numbering plane and implemented in accordance to the recommendations of ITU. With some standard international framework, subscribers from different countries can call each other. This plan makes it possible to access all countries with the same country code anywhere in the world. For the international numbering plan, the world has been divided into nine geographical area. The general rule is that within each global region each country code starts with the same digit. Fig. 9.5 shows the geographical map of world numbering zones.

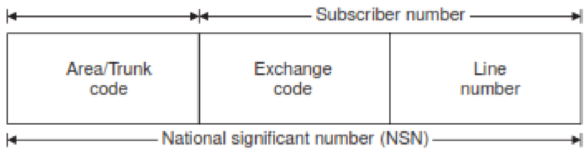


An international telephone number starts with one to three digit country code followed by 9 to 12 subscriber number. The dialling procedure is that the international prefix '00' should be dialled first followed by the telephone number.



National Numbering Plane

Each country decides for itself what kind of numbering plan it can have. A numbering plan may be open, semi open or closed. Each country decides what rules to follow when issuing telephone numbers. Such a numbering plan is called national numbering plan. An open numbering plane or non-uniform numbering scheme allows variations in the number of digits to be used to identify the subscriber. This plan is used in countries equipped extensively with non-director strowger switching system. This scheme is almost extinct. A closed numbering plan or uniform numbering plan refers to a numbering plan which only allows telephone numbers of a predetermined length. Special services (toll free, premium rate, etc.) are usually excluded from this rule. A semi-open plan permits number lengths to differ by almost one or two digits. Today, this scheme is the most common and is used in many countries including India, Sweden, Switzerland and U.K. The dialling procedure for national numbering plan are also comes in two categories. A closed numbering plan 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 the trunk prefix. In open dialling plan local calls can be placed without the trunk prefix and area code.



Numbering Plan in India

DOT India has released its national numbering plan dated April 2003. It was last reviewed during 1993. This existing numbering plan was formulated at a time when there was no competition in the basic telecom services were not available in the country. Further, the existing numbering plan was meant to address monopolistic environment in national and international long distance dialling. The new numbering plan has been formulated for a projected forecast of 50% tele density by the year 2030 and thus making numbering space available for 75 crore telephone connections in the country comprising of 30 crore basic and 45 crore cellular mobile connections. The new national numbering plan will be able to meet the challenges of multi operator, multi service environment and will be flexible enough to allow for scalability for next 30 years without any change in basic structure. This plan is aimed at PSTN services, cellular mobile services and paging services.

National Numbering Scheme. There are ten levels of numbering schemes starting from 0 to 9. In each level, there are many sublevels as a classification of services. All the levels are defined briefly and some sublevels are explained in the following paragraph.

Level 0. Prefix codes. There are various sublevels. Some are described. Sublevel ‘000’ as a prefix shall be used for home country direct service (Bilateral) and international toll free service (Bilateral). The format is 000 + country code + operator code ‘000800’ is used for bilateral international toll free service. Sublevel ‘00’ as a prefix shall be used for international dialling. The format as per E.164 recommendation is 00 + country code + NSN.

Sublevel ‘0’ as a prefix shall be used for national long distance calls. The format is 0 + SDCA code + subscriber number the sublevel ‘09’ is used for cellular mobile services, satellite based services and Intelligent Network (IN) Services.

Level 1. Special services. This level is used for accessing special services like emergency services, supplementary services, inquiry and operator assisted services. The format contains 3 to *N* digit depending on service.

Level 2 to 6. PSTN subscriber number. This format contains the telephone exchange code and subscriber number.

Level 7 and 8. These two levels not being allocated and the same are reserved for new services.

Level 9. Services. The range of numbers in level ‘9’ except ‘90’, ‘95’ and ‘96’ (‘96’ is used in paging services) are reserved for cellular mobile services. Starting from 90 to 99, there are 9 sublevels. ‘90’ is used as spare not allocated for any service.

SIGNALLING TECHNIQUES

A subscriber can be able to talk with or send data to someone in any part of the world almost instantly and an exchange is able to set path and clear it after the conversation instantly by an effective signalling system. A signalling system link the variety of switching system, transmission systems and subscriber equipment in a telecommunication network to enable the network to function as a whole. The signalling are classified according to the internal signalling of an exchange, signalling between exchanges and signalling between an exchange and subscriber. Thus a signalling system must be obviously be compatible with the switching systems which itself partitioned into subsystems in a network. Traditional exchanges sent signals over the same circuit in the network. The introduction of SPC in exchanges enhances the services and introduced new services to the subscriber. These services require more signals to be transmitted and hence needs a separate data channel. The former method of signalling is referred as channel associated signalling and the latter is common channel signalling.

SIGNALLING CLASSIFICATION

Communication networks generally connect equipment such as telephones and fax machines via several line sections, switches and transmission media for exchange of speech, text and data. To achieve this, control information has to be transferred between exchanges for call control. Call control is the process of establishing and releasing a call. This is referred as signalling. In general, signalling is defines as follows. ‘‘Signalling is the process of generating and exchanging information among components of a telecommunication system to establish, monitor or release connections and to control related network and system operations’’.

(*i*) **Supervisory signals or line signals.** These are the signals necessary to initiate a call setup and to supervise it, once it has been established. It is also referred as subscriber loop signalling. Line signals can be transmitted by the use of a single control channel in each direction line.

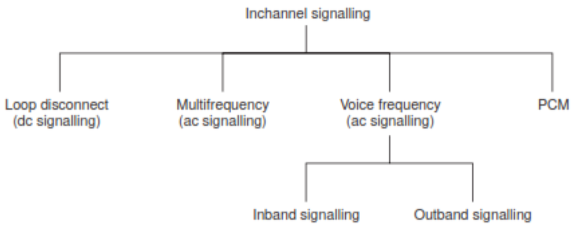
(*ii*) **Routing signals or register signals.** Information transfer related to call setup is usually referred to as register signals. The basic information is the dialled code which indicates to the subsequent switching centres the required routing. In addition to the basic information, signals such as route information, terminal information, register control signals, acknowledgement signals, status of called terminal etc. are also involved.

(*iii*) **Management signals or inter register signalling.** These signals are used to convey information or control between exchanges. This signalling also referred as inter exchange signalling. This signalling involves remote switching of private circuits, routing plans, modification of routing plans, and traffic over load, priority of the call, class of service etc. The signalling may be performed by link-by-link basis, which passes signals exchange to exchange or end-to-end signalling which is between originating and terminating exchange referred as line signalling.

Classification

Traditional signalling uses the same channel to carry voice/data and control signals to carry out the path setup for speech or data transfer. This signalling is referred as in channel signalling or per trunk signalling (PTS). An alternative to in channel signalling is called as common channel signalling (CCS). CCS, uses a separate common channel for passing control signals. It couples the signals for a large number of calls together and send them on a separate signalling channel. Hence basically, signalling technique is classified into 1. In channel signalling/per trunk signalling (PTS) 2. Common channel signalling (CCS)

The inchannel signalling is classified further into four categories as shown in Fig.



The common channel signalling (CCS) is classified according to the transmission of signals between exchangers. They are:

1. associated signalling

2. Non-associated signalling

3. Quasi associated signalling

INCHANNEL SIGNALLING

For data transfer and path setup for conversation, various forms if signalling were developed in automatic switching system. Various classifications are mentioned in last section. In this section, the inchannel signalling which uses same path for control signals and data/speech transfer are discussed.

Loop Disconnect - dc Signalling

The earliest and still the most common telephone set is the rotary dial telephone. The mechanism to transmit the identity of the called subscriber to exchange is pulse dialling. The basic idea is to interrupt the d.c. path of the subscriber's loop for a specified number of short periods to indicate the number dialled. This is called loop-disconnect (or rotary) signalling. The clear signal is produced when the subscriber replaces the handset, by breaking the d.c. path. Any break of more than 100 m sec is assumed as a clear signal. This signalling is relatively cheap but slow. For long distance lines due to the characteristics of lines, performance variance of equipment, the pulse shape may be degraded. This cause more errors in dc signalling.

Multi frequency (mf) – ac Signalling

The slow dialling and hence slow call processing of the rotary dial telephone reduces the productivity of the switching system. The touch tone phone created by AT & T's Western Electric Division sends out frequency based tones instead of electrical impulses. This is called a dial tone multi frequency (DTMF). When a number is pressed, two separate tones are generated and sent to the local exchange simultaneously. As each number has distinct set of dial tones, the switching system recognizes the number dialled. By rotary dial, the pulses are sent at a rate of 10 per second (*i.e.* 10 pulses per sec), whereas tone dialling generated at 23 pulses per sec (PPS). Thus tone dialling are must faster and hence quicker caller processing and increased productivity. The extra keys available in touch tone key pad enables additional/new features. The detection of the digit is carried out by using frequency filters. The frequencies are within the voice band and care must be taken to reduce the risk of speech imitations. The path setup for a rotary call is around 43 seconds, whereas with tone dialling, the call can be setup in approximately 6 to 10 seconds. An *mf* key phone has a little inter-digit pause because the numbers can be keyed very quickly.

Voice Frequency (vf) – ac Signalling

The base band of the telephone channel is 0-4000 Hz. Normally, the speech band occupies the bandwidth of 300-3400 Hz. If the signalling frequencies are chosen within the range of base band of telephone channel, then signalling is referred as voice frequency signalling. The choice of frequency for the control signal depends on various parameters. Based on the selection of frequency the voice frequency signalling are classified into two classes. They are

1. In band signalling

2. Out band signalling

This signalling is also referred as frequency division multiplex (fdm) system.

In band signalling. If the control signal frequencies are within the speech band (300–3400 Hz), the signalling is called inband signalling. Typical in the range of 2280 or 2600 Hz. As it uses the same frequency band as the voice (300–3400 Hz), it must be protected against imitations of speech. For example, if a tone is around 2600 Hz and lasts more than 50 m sec, the switching equipment assume it as a line disconnect signal. Thus, the control signal frequencies must be selected carefully to avoid this limitation. The equipment must be able to distinguish speech and signal. Various parameters are available to distinguish signal from speech. Some of them are

(*i*) Selection of frequency for signal.

(*ii*) Signal recognition time

(*iii*) Voice characteristics.

Advantages of Inband signalling:

1. Inband signalling can be used on any transmission medium.

2. The control signals can be sent to every part where a speech signal can reach.

3. Owing to the flexibility of operation, it is the most widely used signalling system for long distance telephone networks.

4. It is operations are simpler.

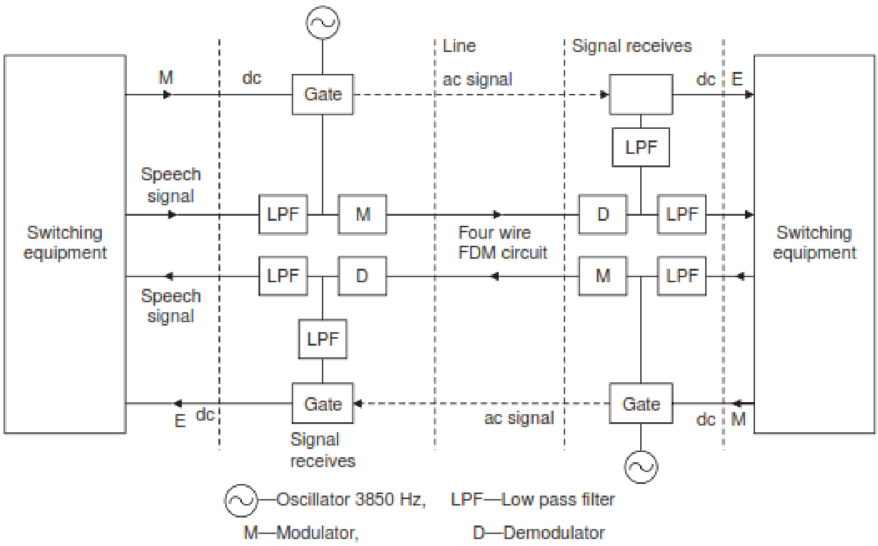
Disadvantages of Inband signalling:

1. More possibility of speech signals imitating control signals. This problem can be reduced using suitable guard circuit.

2. The inband signal may 'spill-over' from one link to another and causes error in that signalling system. This limitation occurs when several transmission links one connected end-to-end. The spill over problem can be eliminated by operating a line split to disconnect link whenever a signal is detected. The line split is designed generally to operate within 35 ms.

Outband signalling

This signalling has frequencies 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 outband signalling is shown in Fig.



Advantages:

1. The requirement of line splits are not necessary to avoid signal 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.

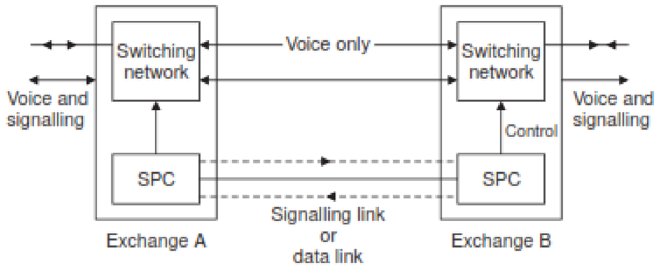
3. More dependent on the transmission system.

PCM Signalling

In PCM systems, signalling and speech are sampled, coded and transmitted within the frame of PCM channels. Thus, with PCM, a convenient way of transmission is possible. The signalling information and speech information carried in the same time slot is referred as in slot signalling. The signalling information carried in a separate time slot is referred as out slot signalling. The timeslot of the inslot signalling is fixed at eight bits. As one bit is used for signalling, the speech bit rate is reduced to 56 kbps from 64 kbps and the bandwidth is reduced. Telephone channels are combined by time division multiplexing to form an assembly of 24 or 30 channels. This is known as primary multiplex group. Two frame structures are widely used in practice. They are DS1 24 channel system and European 30 channel system. DS1 24 channel system is popular in North America and Japan. Originally, DS1 is called as T1 system. T1 (Telecommunication standards entity number 1) is a standards committee designated as ANSI T1-nnn-date. T1 uses some very specific conventions to transmit information between both ends. One of these is a framing sequences that formats the samples of voice or data transmission. Framing undergone several evolutions.

COMMON CHANNEL SIGNALLING

Introduction of SPC digital switching systems with high speed processors in the telecom network has necessitated modernisation of signalling. Also, in order to meet the transfer of varieties of information for call management and network management and to satisfy the subscribers' requirement on various features, the uninterrupted, high speed signalling has become inevitable. The rapid development of digital systems paved way for the new signalling system called common channel signalling. Instead of using the same link for signalling information and message as in In channel signalling, the common channel signalling (CCS) uses a dedicated line for the signalling information between the stored program control elements of switching systems. Fig. shows the basic schematic of CCS.



The data link sends messages that identify specific trunks and events. Two signalling channels, one for each direction are used in a dedicated manner to carry signalling information. Hence, they are capable of carrying information for a group of circuits. At the bit rate of 2.6 kbps, CCS can carry signals for 1500–2000 speech circuits. CCS network is basically a store and forward network. In CCS network, the signalling information travels in a link-by-link basis along the route. The information arrived at a node is sorted, processed and forwarded to the next node in the route. The CCS technique is also called the transparent mode for signalling.

Advantages and Disadvantages of CCS

Advantages. The major advantages of CCS are listed below:

1. For each associated trunk group, only one set of signalling facility is required. The channel used for CCS need not be associated with any particular group. Fig. shows a CCS network that is disassociated from the message network structure.

2. The introduction of SPC switching machines and CCS provides efficient routing procedure.

3. CCS allows for signalling at any time in the entire duration of a call, not only at the beginning.

4. CCS removes most of the signalling costs associated with inter office trunks.

5. Information can be exchanged between processor at high speed.

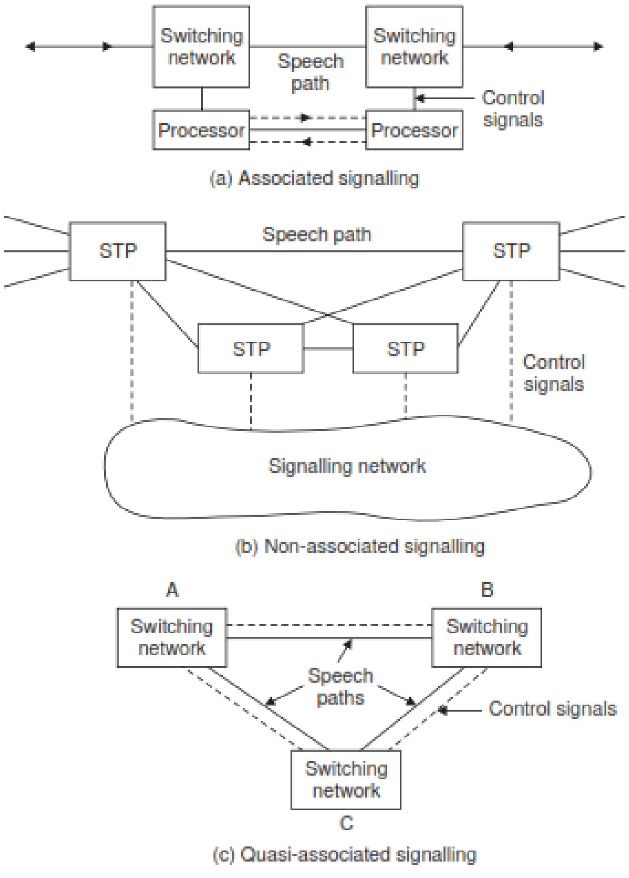
6. The CCS provides acceptable quality for network related signalling tones such as DTMF, MF and SF achieved.

7. As there is no need of line signalling equipment on every function, considerable cost savings can be achieved.

8. The D1 channel bank use 1 bit per time slot for signalling and 7 bits for voice which provides signalling rate 8 kbps. D2 channel provides higher data rate which use 1 bit in every sixth frame for signalling. This is referred as ‘‘robbed bit signalling.’’ When CCS is utilized, the associated T carrier system no longer need to carry signalling information on a per channel basis and a full 8 bits of voice can be transmitted in every time slot of every frame.

9. As separate channels are used for voice and control. There is no chance of mutual interference and the error rate is very low.

10. CCS enables more services to the subscribers. A signalling link operating at 64 kbit/s normally provides signalling up to 1000 or 1500 speech circuits.



Disadvantages of CCS:

Some disadvantages of CCS are listed below:

1. The CCS network is basically a store and forward network. So, in a established circuit, the signalling information are stored, processed and then forwarded to next node. This causes additional overhead and disconnect to the continuity.

2. If one node fails to transmit properly, the facilities downstream from the disconnect will not be released. Thus, a high degree of reliability is required for the common channel.

3. Proper interfacing facility is necessary, as most of the present day telephone networks are equipment with inchannel signalling systems.

4. As the signalling information is not actually sent over speech paths in CCS, the integrity of speech path is not assured. As a remedy, routing testing of idle paths and the continuity test of an established path become necessary in CCS.

5. Different trunks in a group may terminate at different switch, say local exchange, other foreign exchange circuits etc. With CCS, all trunks are first terminated to the local central office and then forward to the different destination.

In channel and common channel signalling comparison is tabulated in Table.

Inchannel signalling	**Common channel signalling**
Automatic propagation of signalling information enables the simultaneous process and release of associated facilities.	Signalling information must be relayed from one node to the next in a store and forward fashion.
Integrity of speech and signalling are maintained as they are on the same path.	Special equipments should be provided for the integrity as they are travelling on separate path.
Separate signalling equipment is required for each trunk and hence is expensive.	Only one set of signalling equipment is required for a whole group of trunk circuits and therefore CCS is economical.
Transfer of information such as address digits is from common control network originating office — Voice channel — receiving office — common control network.	Transfer of information is directly between control elements (processors of SPC systems).
Trunks are held up during signalling.	Trunks are not required for signalling.

Module-IV: OVERVIEW OF

ISDN

Integrated Services Digital Network is a telephone system network. It is a wide area network becoming widely available. Prior to the ISDN, the phone system was viewed as a way to transport voice, with some special services available for data. The key feature of the ISDN is that it integrates speech and data on the same lines, adding features that were not available in the classic telephone system.

ISDN is a circuit -switched telephone network system, that also provides access to packet switched networks, designed to allow digital transmission of voice and data over ordinary telephone copper wires, resulting in better voice quality than· an analog phone. It offers circuit-switched connections (for either voice or data), and packet-switched connections (for data), in increments of 64 Kbit/s.

Another major market application is Internet access, where ISDN typically provides a maximum of 128 Kbit/s in both upstream and downstream directions (which can be considered to be broadband speed, since it exceeds the narrowband speeds of standard analog 56k telephone lines). ISDN B-channels can be bonded to achieve a greater data rate; typically 3 or 4 BRIs (6 to 8 64 Kbit/s channels) are bonded.

ISDN should not be mistaken for its use with a specific protocol, such as Q.931 whereby ISDN is employed as the network, data-link and physical layers in the context of the OSI model. In a broad sense ISDN can be considered a suite of digital services existing on layers 1, 2 and 3 of the OSI model. ISDN is designed to provide access to voice and data services simultaneously.

However, common use has reduced ISDN to be limited to Q.931 and related protocols, which are a set of protocols for establishing and breaking circuit switched connections, and for advanced call features for the user. They were introduced in 1986. In a videoconference, ISDN provides simultaneous voice, video, and text transmission between individual desktop videoconferencing systems and group (room) videoconferencing systems.

The first generation of ISDN is called as a narrowband ISDN and it is based on the use of 64 kbps channel as the basic unit of switching and has a circuit switching orientation. The main device in the narrowband ISDN is the frame relay. The second generation of ISDN is referred to as the broadband ISDN (B-ISDN).

It supports very high data rates (typically hundreds of Mbps). It has a packet switching orientation. The main important technical contribution of B-ISDN is the asynchronous transfer mode (ATM), which is also called as cell relay.

ISDN Interfaces

There are several kinds of access interfaces to the ISDN:

Basic Rate Interface (BRI)

Basic Rate Interface service consists of two data-bearing channels ('B' channels) and one signalling channel ('D' channel) to initiate connections. The B channels operate at 64 Kbps maximum; however, (in the U.S. it can be limited to 56 Kbps.

The D channel operates at a maximum of 16 Kbps. The two channels can operate independently. For example, one channel can be used to send a fax to a remote location, while the other channel is used as a TCP/IP connection to a different location. ISDN service on the iSeries supports basic rate interface (BRl).

The basic rate interface (BRl) specifies a digital pipe consisting of two B channels and 16 Kbps D channel. Two B channels of 64 Kbps each, plus one D channel of 16 Kbps, equal 144 Kbps. In addition, the BRl service itself requires 48 Kbps of operating overhead. BRl therefore requires a digital pipe of 192 Kbps. Conceptually, the BRl service is like a large pipe that contains three smaller pipes, two for the B channels and one for the D channel.

The remainder of the space inside the large pipe carries the overhead bits required for its operation.

Primary Rate Interface (PRI)

Primary Rate Interface service consists of a D channel and either 23 (depending on the country you are in). PRI is not supported on the iSeries. Or 30 B channels

The usual Primary Rate Interface (PRI) specifies a digital pipe with 23 *B* channels and one 64 Kbps D channel. Twenty-three B channels of 64 Kbps each, plus one D channel of 64 Kbps equals 1.536 Mbps. In addition, the PRI service itself uses 8 Kbps of overhead.

PRI therefore requires a digital pipe of 1.544 Mbps. Conceptually; the PRI service is like a *large* pipe containing 24 smaller pipes, 23 for the B channels and 1 for the D channel. The rest of the pipe carries the overhead bits required for its operation.

Broadband-ISDN (B-ISDN)

Narrowband ISDN has been designed to operate over the current communications infrastructure, which is heavily dependent on the copper cable. B-ISDN however, relies mainly on the evolution of fibre optics. According to CCITT B-ISDN is best described as 'a service requiring transmission channels capable of supporting rates greater than the primary rate.

Principle of ISDN

The ISDN works based on the standards defined by ITU-T (formerly CCITT). (The Telecommunication Standardization Sector (ITU-T) coordinates standards for telecommunications on behalf of the International Telecommunication Union (ITU) and is based in Geneva, Switzerland. The standardization work of ITU dates back to 1865, with the birth of the International Telegraph Union.

It became a United Nations specialized agency in 1947, and the International Telegraph and Telephone Consultative Committee (CCITT), (from the French name "Comite Consultatif International Telephonique et Telegraphique") was created in 1956. It was renamed ITU-T in 1993.

Principle of ISDN according to ITU –T

The ISDN is supported by a wide range of voice and non-voice applications of the same network. It provides a range of services· using a limited set of connections and multipurpose user-network interface arrangements.

ISDN supports a variety of applications that include both switched and non-switched connections. The switched connections. Include both circuit and packet switched connections.

As far as possible, new services introduced into an ISDN should be arranged to be compatible with the 64 Kbps switched digital connections.

A layered protocol structure should be used for the specification of access to an ISDN.

This is the same as the OSI reference model. The standards which have already been developed for OSI applications such as X.25 can be used for ISDN.

ISDNs may be implemented in a variety of configurations.

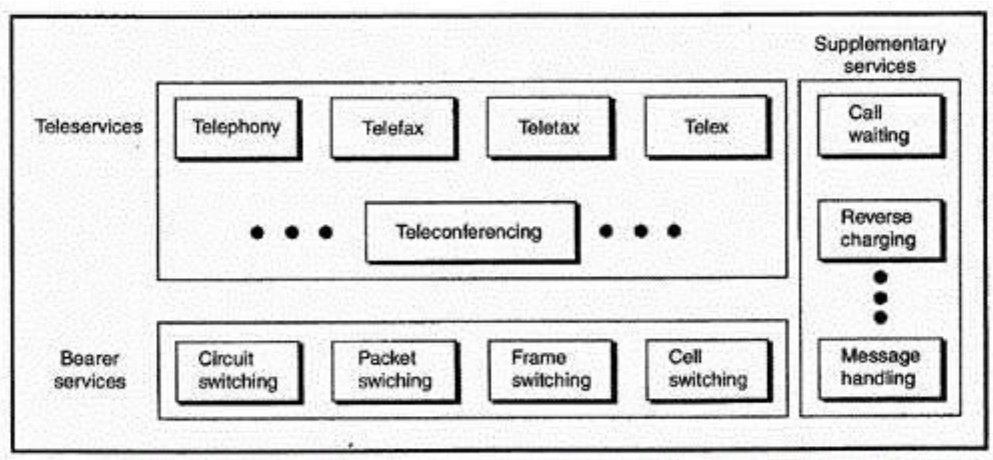
ISDN Services

The purpose of the ISDN is to provide fully integrated digital services to users. These services fall into categories- better services, teleservices and supplementary services.

1. Bearer Services: Bearer services provide the means to transfer information (voice, data and video) between users without the network manipulating the content of that information. The network does not need to process the information and therefore does not change the content.

Bearer services belong to the *first* three layers of the OSI model and are well defined in the ISDN standard. They can be provided using circuit-switched, packet-switched, frame-switched, or cell-switched networks.

2. Teleservices: In teleservices, the network may change or process the contents of the data. These services correspond to layers 4-7 of the OSI model. Teleservices relay on the facilities of the bearer services and are designed to accommodate complex user needs, without the user having to be aware of the details of the process. Teleservices include telephony, teletex, telefax, videotex, telex and teleconferencing. Although the ISDN defines these services by name, they have not yet become standards.



3. Supplementary Service: Supplementary services are those services that provide additional functionality to the bearer services and teleservices. Examples of these services are reverse charging, call waiting, and message handling, all familiar from today's telephone company services.

Principles of ISDN

The various principles of ISDN as per ITU-T recommendation are:

I. To support voice and non-voice applications

The main feature of the ISDN concept is the support of a wide range of voice (for *e.g.* Telephone calls) & non-voice (for *e.g.* digital data exchange) applications in the same network.

2. To support switched and non-switched applications

ISDN supports both circuit switching and packet switching. In addition ISDN supports non-switched services in the form of dedicated lines.

3. Reliance on 64-kbps connections

ISDN provides circuit switched and packet switched connections at 64 kbps. This is the fundamental building block of ISDN. This rate was chosen because at the time, it was standard rate for digitized voice.

4. Intelligence in the network

An ISDN is expected to provide sophisticated services beyond the simple setup of circuit switched calls. These services include maintenance and network management functions.

5. Layered protocol architecture

A layered protocol structure should be used for the specification of the access to an ISDN. Such a structure can be mapped into OSI model.

6. Variety of configurations

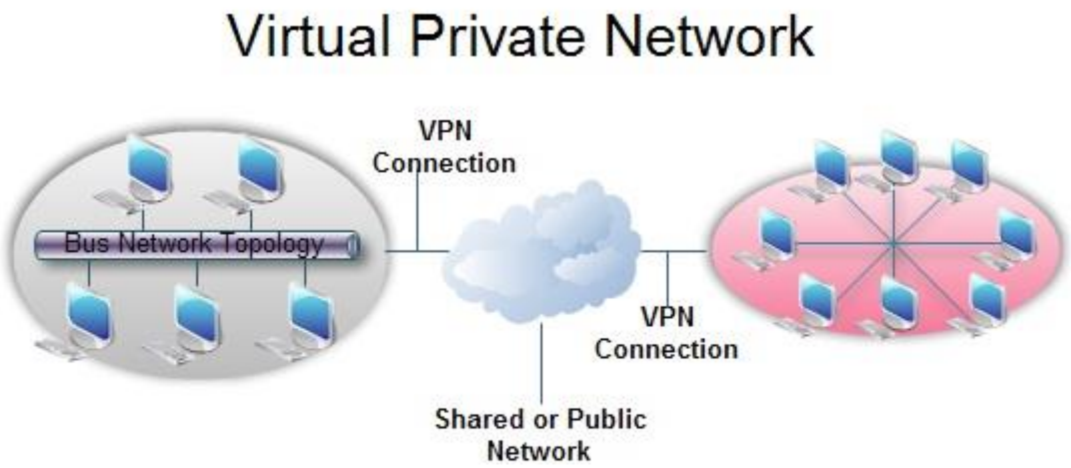
Several configurations are possible for implementing ISDN. This allows for differences in national policy, in state of technology and in the needs and existing equipment of the customer base.

VPN (*Virtual Private Network*) *Definition*: VPN meaning that it is a private point-to-point connection between two machines or networks over a shared or public network such as the internet. VPN (Virtual Private Network) technology, can be use in organization to extend its safe encrypted connection over less secure internet to connect remote users, branch offices, and partner private, internal network. VPN turn the Internet into a simulated private WAN.

A VPN client uses TCP/IP protocol, that is called tunnelling protocols, to make a virtual call to VPN server.

What is VPN

VPN allows users working at home or office to connect in a secure fashion to a remote corporate server using the routing infrastructure provided by a public inter-network (such as the Internet). From the user's perspective, the VPN is a point-to-point connection between the user's computer and a corporate server. The nature of the intermediate inter-network is irrelevant to the user because it appears as if the data is being sent over a dedicated private link.



Main Network Protocols

There are three network protocols are used within VPN tunnels. That are:

IPSec

IPsec (*Internet Protocol Security*) is a framework for uses cryptographic security services developed by the IETF to protect secure exchange communications over Internet Protocol (IP).It supported encryption modes are transport and tunnel.

PPTP

PPTP (*Point-to-Point Tunneling Protocol*) is a network protocol that extending the organization private networks over the public Internet via "tunnels.

L2TP

L2TP (*Layer Two Tunneling Protocol*) is an extension of the Point-to-Point Tunneling Protocol (PPTP) used by Internet service providers (ISPs) to operate Virtual Private Networks (VPNs).

Privacy, Security and Encryption

Data sent across the public Internet is generally not protected from curious eyes, but you can make your Internet communications secure and extend your private network with a virtual private network (VPN) connection. VPN uses a technique known as tunneling to transfer data securely on the Internet to a remote access.

The Internet connection over the VPN is encrypted and secure. New authentication and encryption protocols are enforced by the remote access server. Sensitive data is hidden from the public, but it is securely accessible to appropriate users through a VPN.

How to Setup a VPN

There are following two ways to create a VPN connection:

By dialling an Internet service provider (ISP)

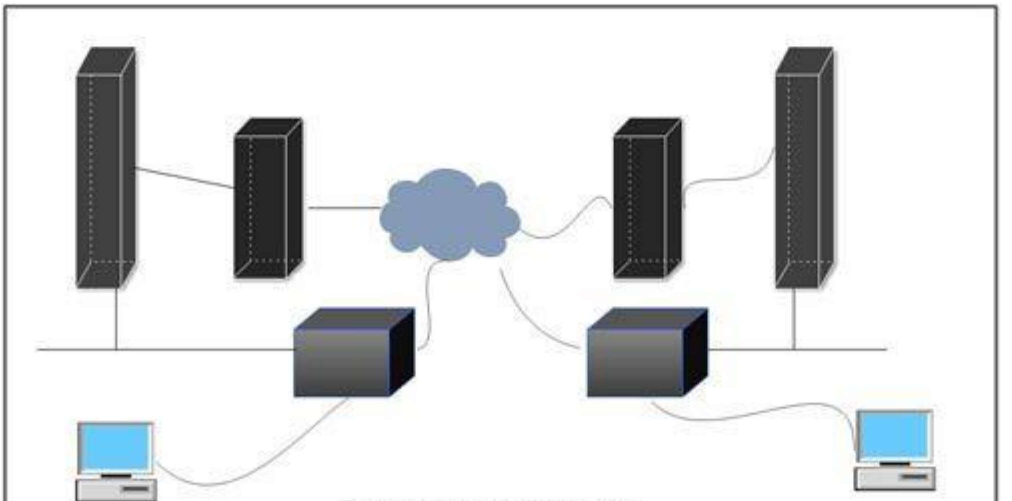
If you dial-in to an ISP, your ISP then makes another call to the private network's remote access server to establish the PPTP or L2TP tunnel after authentication, you can access the private network.

By connecting directly to the Internet

If you are already connected to an Internet, on a local area network, a cable modem, or a digital subscriber line (DSL), you can make a tunnel through the Internet and connects directly to the remote access server. After authentication, you can access the corporate network.

Equipment of VPN

Equipment having the VPN function includes routers and firewalls. Basically, communication is made via VPN equipment. Information is encrypted by the transmission VPN equipment before transmission and decoded by the receiving VPN equipment after receipt of information. The key for encrypt the data is set in VPN equipment in advance. The VPN equipment at receiving side decodes encrypted data before sending it to the receiving computer.



Equipment having the VPN function

The advantages of encryption by way of cryptography may be looked into other services, such as

1. Assuring integrity check: This ensures that undesirable person has not tampered data delivered to the destination during transmission.
2. Providing authentication: Authentication authorizes the sender identity.

Features of a Typical VPN solution

When the remote offices connect each other to share vital resources and secret information, the VPN solution must ensure the privacy and integrity of the data as it traverses the Internet. Therefore, a VPN solution must provide at least all of the following:

Keep data confidential (encryption)

• Data carried on the public network must be rendered unreadable to unauthorized clients on the network.

Ensure the identities of two parties communicating (authentication)

• The solution must verify the user's identity and restrict VPN access to authorized users only. It must also provide audit and accounting records to show who accessed what information and when.

• Safeguard the identities of communicating parties (tunnelling)

• Guard against packets being sent over and over (replay prevention)

• Ensure data is accurate and in its original form (non-repudiation)

Address Management. The solution must assign a client's address on the private net and ensure that private addresses are kept private.

Key Management. The solution must generate and refresh encryption keys for the client and the server.

Multiprotocol Support. The solution must handle common protocols used in the public network. These include IP, Internet Packet Exchange (IPX), and so on.

An Internet VPN solution based on the Point-to-Point Tunneling Protocol (PPTP) or Layer 2 Tunneling Protocol (L2TP) meets all of these basic requirements and takes advantage of the broad availability of the Internet. Other solutions, including the new IP Security Protocol (IPSec), meet only some of these requirements, but remain useful for specific situations.

Benefits of VPN

The main benefit of a VPN is the potential for significant cost savings compared to traditional leased lines or dial up networking. These savings come with a certain (in amount of risk, however, particularly when using the public Internet as the delivery mechanism for VPN data.

The performance of a VPN will be more unpredictable and generally slower than dedicated lines due to public Net traffic. Likewise, many more points of failure can affect a Net-based VPN than in a closed private system. Utilizing any public network for communications naturally raises new security concerns not present when using more controlled environments like point-to-point leased lines.

Voice over Internet Protocol, also called Voice over IP or just VoIP technology is having a major impact on the telecommunications industry. VoIP technology provides advantages for both the user and also the provider, allowing calls to be made more cheaply, as well as enabling data and voice to be carried over the same network efficiently. In view of the way VoIP technology is being adopted, telecommunications providers are having to adopt the new technology. Already it has caused some impact on major businesses, and there will be more to come.

Until recently voice traffic was carried using a circuit switched approach. Here a dedicated circuit was switched to provide a call for a user. Now with new data and Internet style technology used for VoIP, packet data and Internet Protocol (IP) is used to enable a much more efficient use of the available capacity.

What is VoIP?

- *VoIP definition:* VoIP (voice over IP) is the transmission of voice and multimedia content over Internet Protocol (IP) networks. VoIP is enabled by a group of technologies and methodologies used to deliver voice communications over the internet, enterprise local area networks or wide area networks.

The concept of Voice over Interent Protocol, Voice over IP, or VoIP, is quite straightforward. A VoIP system basically consists of a number of endpoints which may be VoIP phones, mobile phones, VoIP enabled browsers on computers, etc and also an IP network over which the packet data is carried.

In a VoIP system, the phone or computer acting as an endpoint consists of a few blocks. It includes a vocoder (voice encoder / decoder) which converts the audio to and from the analogue format into a digital format. It also compresses the encoded audio, and in the reverse direction it decompresses the reconstituted audio. The data generated is split into packets in the required format by the network interface card which sends them with the relevant protocol into the outside world. Signalling and call control is also applied through this card so that calls may be set up, pulled down, and other actions may be undertaken.

The IP network accepts the packets and provides the medium over which they can be forwarded, routing them to their final destination. As complete circuits are not dedicated to a given user, at times when no data needs to be sent, for example during quiet periods in speech, etc., the capacity can be used by other users. This makes a significant difference to the efficiency of a system, and allows significant savings to be made.

Traditionally the term VoIP referred to systems where IP was used to connect private branch exchanges, PBXs, but now the term is more widely used and encompasses IP telephony.

VoIP Protocols

In order to be able to communicate using a VoIP system, there are a number of protocols that may be used.

- *H323:* The signalling protocol is used to control and manage the call. It includes elements such as call set up, clear down, call forwarding and the like. The first protocol to be widely used for VoIP was H323. However this is not a particularly rigorous definition and as a result other variants have been developed.
- *Skinny:* One other signalling protocol that was used was known as "Skinny" and is a Cisco Proprietary protocol and is from Nortel and another is called Unistem. In view of this there are often interfacing problems.
- *SIP:* SIP, Session Initiation Protocol, is now being widely adopted as the main standard is a far more rigorous protocol for signalling and is the one that is most widely used now.

- *RTP:* RTP, Real Time Protocol, is a data exchange protocol and this can handle both audio and video. RTP handles the data exchange, but in addition to this a codec is required. Where voice is used a vocoder is used (a codec can be used for any form of data including audio, video, etc).
- *G711:* G711 is possibly the most widely used VoIP vocoder and it is the standard for transmitting uncompressed packets. G.729 is the standard for compressed packets. Many equipment vendors also use their own proprietary codecs. Voice quality may suffer when compression is used, but compression reduces bandwidth requirements. There are many other vocoders/ codecs that are used with varying data rates and providing different levels of voice quality.

Service quality

Quality of Service, QoS, for the data link has a major impact on VoIP perceived sound quality. The data exchange must take place in real time and any delays in the system cause significant disruption to the traffic. Delayed packets may mean that packets arrive out of order, or with varying gaps between them, resulting in garbled speech, Packets may even disappear resulting in lost information.

For any packet passing through an IP network it is possible to define the class of service required. It is important that packets that need to be transferred in real time are given a higher quality of service than those that can be transferred as the network permits. This is particularly important for services like VoIP that are termed delay sensitive applications.

Advantages

Voice over IP, VoIP technology provides a number of significant advantages to operators and to users. For the user one of the main advantages is the flexibility. Phones are software based, sometimes being attached to computers. As a result a considerable degree of flexibility is afforded to the user. It is possible to move the phone around and by enabling the system to recognise the individual phone it is possible to route the data to it automatically. In addition to this ideas such as mobile IP could enable the user to be located away from the home network and still receive calls.

A further advantage is that the wireless network technologies such as 802.11 can carry the calls as voice is simply another form of application. This gives further flexibility as the phone does not have to be physically wired to a network. Again Quality of Service is a major factor and this is being addressed under 802.11e

For the operator some of the advantages are different. One of the major drivers towards the use of VoIP is cost. Previously digital traffic was handled using time division techniques. This had the disadvantage that when a particular time slot allocated to a user was dormant, it could not be used. Using IP techniques much higher levels of efficiency can be attained. Although the system required to carry packet data is more complicated, the returns far outweigh the additional costs.

As with all technologies there are disadvantages. The main one with VoIP is voice quality. This results from the use of a vocoder to digitise and compress the audio. Quality is comparable with that from a mobile phone, but for the future with rapidly improving standards of vocoders there are likely to be significant improvements in this area.

In the long term VoIP is the way the market is moving, and now with increasing speed. Offering not only great improvements in flexibility, but also major cost savings, but with the requirement for large levels of investment, this is the way that the telecommunications market is moving. However to remain competitive it will be necessary to adopt the new VoIP technology.

VoIP protocols overview

Although working together, there are a number of different organizations and bodies that are mentioned when referring to VoIP protocols:

- IETF This is the Internet Engineering Task Force. It is a community of engineers that defines some of the prominent standards used on the Internet (including VoIP protocols) and seeks to spread understanding of how they work.
- ITU the International Telecommunication Union. This is an international organization within the United Nations System used by where governments and private sector companies to coordinate and standardize telecommunications networks, services and standards on a global basis.

In addition to the organizations involved, there is also a variety of different VoIP protocols and standards.

- H.248 H.248 is an ITU Recommendation that defines "Gateway Control Protocol" and it is also referred to as IETF RFC 2885 (Megaco). It defines a centralized architecture for creating multimedia applications and it extends MGCP. H.248 is the result of a joint collaboration between the ITU and the IETF and it is another VoIP protocol.
- H.323 This is ITU Recommendation that defines "packet-based multimedia communications systems." H.323 defines a distributed architecture for multimedia applications, and it is thus a VoIP protocol.
- Megaco This is also known as IETF RFC 2885 and ITU Recommendation H.248. H.248 defines a centralized architecture for creating multimedia applications.
- Media Gateway Control Protocol (MGCP) This is also known as IETF RFC 2705. It defines a centralized architecture for creating multimedia applications, and it is therefore a VoIP protocol.
- Real-Time Transport Protocol (RTP) This VoIP protocol is defined under IETF RFC 1889 and it details a transport protocol for real-time applications. RTP provides the transport mechanism to carry the audio/media portion of VoIP communication and is used for all VoIP communications.
- Session Initiation Protocol (SIP) This is also known as IETF RFC 2543 and it defines a distributed architecture for creating multimedia applications.

Centralised and distributed architectures

One of the advantages of VoIP is that it does not legislate for the architecture of the network that carries the data. Early telecommunications networks used a centralised structure where all the intelligence was contained at the switching station or exchange. With the advent of packet technology, the routing and intelligence can be distributed to where it is most convenient to locate it. This may be by having a distributed architecture, or a centralised one. While both architectures can be employed with VoIP, the type of architecture does have an impact on the optimum VoIP protocols to use. This is one of the reasons why a number of VoIP protocols are used, and will remain to be used.

P networks carrying VoIP traffic are very complicated. They carry both voice and data traffic and this results in a variety of traffic with different requirements being carried and this presents many challenges. In order to ensure that all the requirements are met and the network operates to its maximum efficiency can present many challenges. Obviously the design must be correct, but once implemented testing of the network is needed to ensure that it is able to operate correctly when installed, and then maintained correctly ensuring that its performance is maintained or optimised to provide the performance meets the needs of the network provider and the user. For VoIP, testing is an essential element of any network. However specialised VoIP testing techniques are required.

VoIP network architecture

The structure of a VoIP network comprises many entities and this means that VoIP testing is essential to ensure that the network is operating satisfactorily. A typical VoIP network will include many different entities:

- Signalling gateways
- Media gateways
- Gatekeepers
- Class 5 switches
- SS7 network
- Network management system
- Billing system

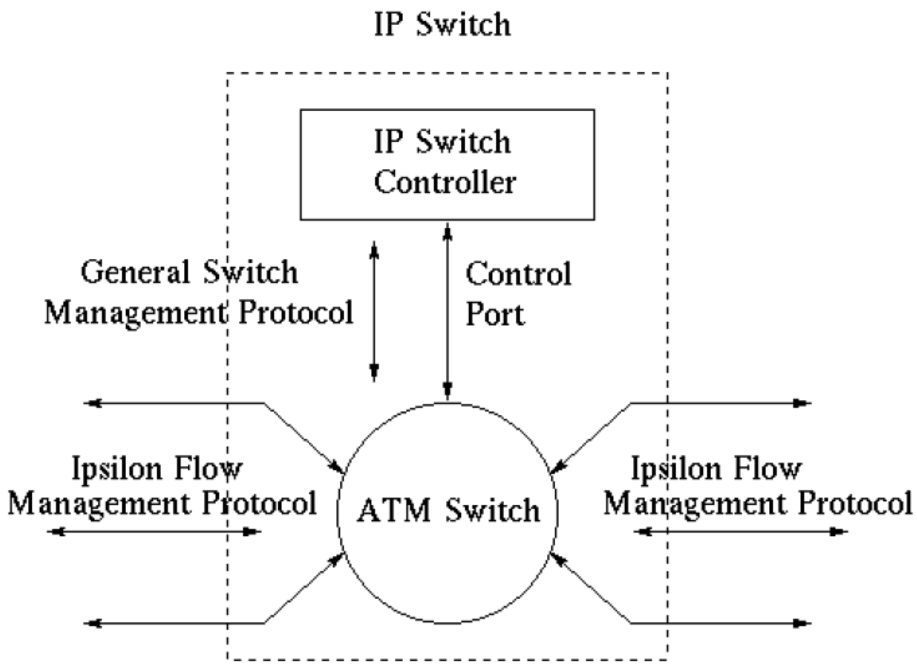
This variety of different entities within the VoIP network all communicate with each other using a variety of protocols. To perform correctly it is necessary to ensure that they communicate efficiently and that no bottlenecks are created. Analysing the performance of a VoIP network is not always easy. However it can be achieved and significant improvements in performance can be achieved if the VoIP testing scenarios are carefully chosen and planned, and the data analysed to reveal any problems.

IP Switching

This technology [Flow Labelled IP] has been developed by Ipsilon Networks Inc. The goal of Ipsilon is to make IP faster and offer the quality of service support. The "IP over ATM" approach tries to hide the underlying network topology from IP layer by treating the datalink layer as large, opaque network cloud. However, this leads to inefficiency, complexity and duplication of functionality in the resulting network [ATM under IP]. Ipsilon's approach is to discard the connection-oriented ATM software and implement the connectionless IP routing directly on the top of ATM hardware. This approach takes the advantage of robustness and scalability of connectionless IP and speed, capacity and scalability of ATM switches.

IP Switch

IP switch is basically an IP router with attached switching hardware that has the ability to cache routing decisions in switching hardware. To construct an IP switch, a standard ATM switch is taken, the hardware is left untouched, but all the control software above AAL-5 is removed. It is replaced by standard IP routing software, a flow classifier to decide whether to switch a flow or not and a driver to control the switch hardware. At system start up a default virtual channel is established between the control software of the IP switch and its neighbours, which is then used for default hop-by-hop forwarding of IP datagrams. To gain the benefits of switching, a mechanism has been defined to associate IP flows with the ATM labels.



Structure of an IP switch

Flow Classification

A flow is a sequence of packets from a source to a destination which get the same forwarding treatment at a router. An IP flow is characterized by the fields in IP/TCP/UDP header such as source address, destination address, port number, etc. Currently two types of flows have been defined by Ipsilon. Host pair flow type is defined for traffic between same source and destination IP address. A Port pair flow type is defined for traffic between same source and destination port between same source and destination IP address. When a packet is assembled and submitted to controller in IP switch, apart from forwarding it to the next hop it also does flow classification. Flow classification decision is a policy decision local to a IP switch. Depending on the classification, the switch decides to forward or switch the next packets of that flow. Usually the controller would decide to forward short term flows like database queries, DNS messages and switch long term flows like ftp or telnet data.

IP Switching Features

Following are some important features of IP Switching:

- Point-to-Point IP Switching advocates point-to-point network model for ATM rather than a logical shared medium model as proposed by some competing approaches.

- Multicast In case of multicast, a flow coming to an IP Switch will branch out to multiple destinations. Each of these branches can be individually redirected. If the whole flow is labelled, then multicast capabilities of ATM can be used.

- Quality of Service, An IP Switch can make QoS decision according to its local policy. The QoS information can be included in the flow classification process. Individual QoS requests for each flow can be supported using resource reservation protocols like RSVP. An IP Switch can be configured to utilize traffic management capabilities of ATM to guarantee the desired QoS.

- Latency The latency in setting up of a virtual channel from source to destination can be low for connection oriented protocols like TCP. Due to 3-way handshake in TCP setup procedure, a virtual channel can be established even before the first data packet is sent across. Also the tolerance in case of failures is good because the IP Switches can always go back to connectionless packet forwarding via a different route if any link fails.