Channel associated signaling – Common channel signaling - SS7 protocol - traffic – grade of service – Modeling switching system –Blocking models and delay system.

## 3.1 SIGNALING SYSTEMS

A signaling system in a data communication networks exchanges signaling information effectively between subscribers. The signaling systems are essential building blocks in providing automatic telephone services.

### 3.1.1 SIGNALLING TECHNIQUES

Signaling systems link the variety of switching systems, transmission systems and subscriber equipments in a telecommunication network to enable the network to function as a whole.

Three forms of signaling are involved in telecommunication network:

1. Subscriber loop signaling
2. Intraexchange or register signaling
3. Interexchange or inter register signaling

**Subscriber loop signaling**

Subscriber loop signaling depends upon the type of telephone instrument used. Multi frequency signaling has brought about new services like data in-voice answer, which fall in the class of user to user signaling facilities.

**Intraexchange or register signaling**

Intraexchange signaling is internal to the switching system and is heavily dependent upon the type and design of a switching system. It varies from one model to another even with the same manufacturer.

**Interexchange or inter register signaling**

Interexchange signaling takes place between exchanges with common control subsystems and is called inter register signaling. In this main purpose is the exchange of address digits which pass from exchange to exchange on a link by link basis.

**Line signaling**

Network wide signaling also involves end to end signaling between originating exchange and terminating exchange and that form is called line signaling.

**Classification of signaling system**

Signaling techniques falls under two broad classes:

(1) Inchannel signaling

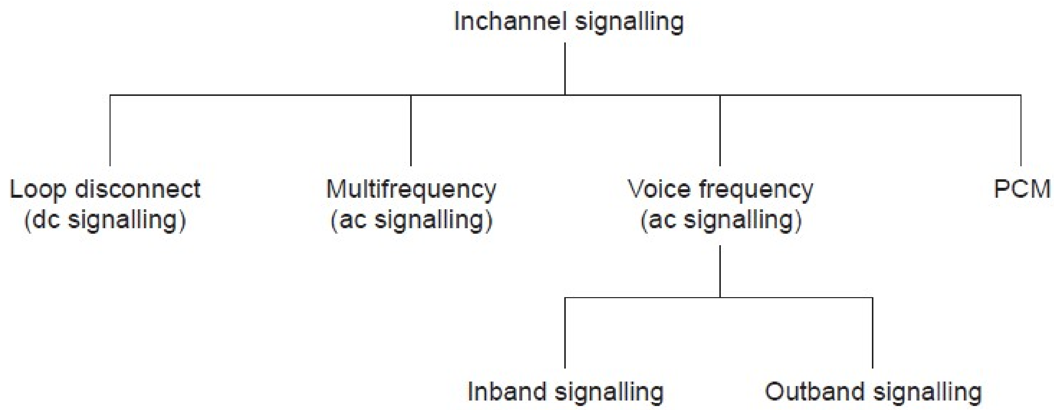
(2) Common channel signal

In in-channel signaling, signals are transferred using the same channel which carries voice or data to pass control signals related to that call or connection.

In common channel signaling, Signals are transferred using the separate common channel for passing control signals. It does not use the speech or the data path for signaling.

**Inchannel signaling**

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




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**. 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

**In band signalling.** If the control signal frequencies are within the speech band (300– 3400 Hz), the signalling is called inband signalling. As it uses the same frequency band as the voice (300–3400 Hz), it must be protected against imitations of speech.

**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.

**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 the another and causes error in that signalling system. This limitation occurs when several transmissions links one connected end-to-end.

**Outband signalling**

This signalling has frequencies above the voice band but below the upper limit of 4 kHz. The recommended frequency for outband signalling is 3825 Hz, but 3700 Hz and 3850.

**Advantages:**

1. The requirement of line splits is not necessary to avoid signal limitation.
2. Signals and speech can be transmitted simultaneously without disturbing the conversation.

**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.

## 3.2 COMMON CHANNEL SIGNALLING

In CCS, signaling is completely separate from switching and speech. In this case signaling is done over a channel that is different from the one which carries the voice or data. Here separate analog voice channel is used for signaling.

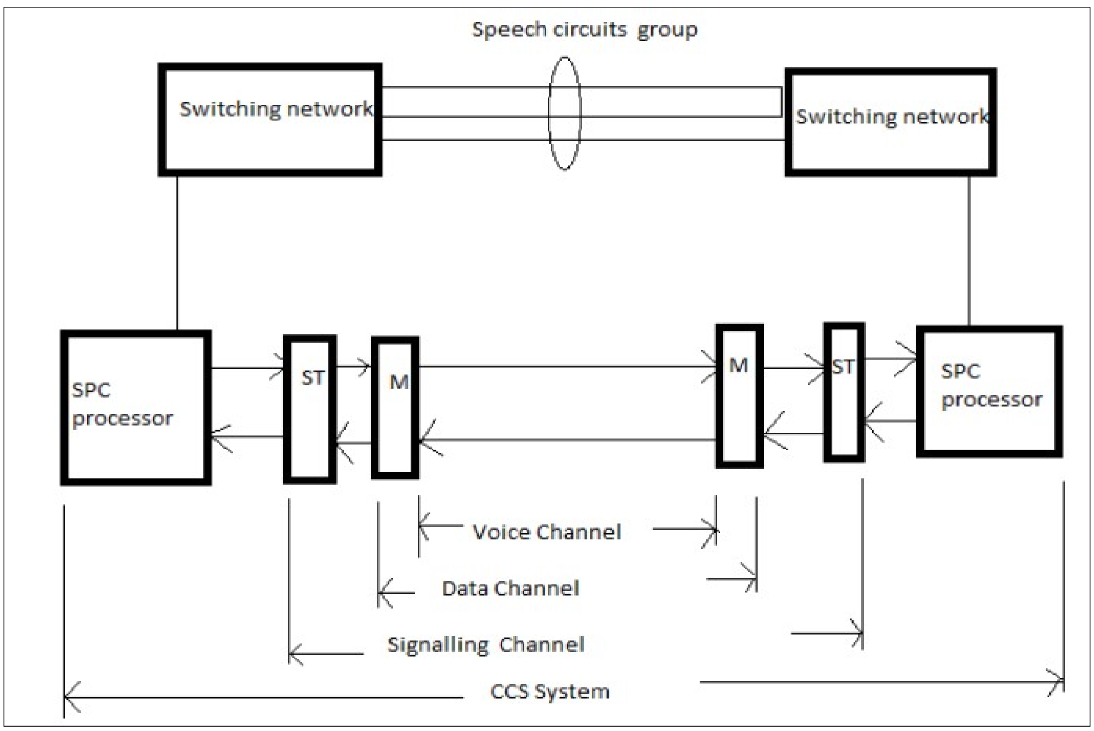



**Figure. 3.1: Basic scheme of CCS**

In the above figure.3.1, two signalling channels, one for each direction, are used in a dedicated manner to carry signalling information. Since the channels are dedicated for signalling they are capable of carrying signalling information for a group of circuits.

The CCS network is basically a **store and forward (S&F) network** where signaling information travels on a link-by-link basic along the route. When the signalling information is received at a node, it is stored, processed and forwarded to the next node in the route.

In CCS, signalling information is transferred as message of varying length usually defined as one or more fixed length is called **signaling units (SUs**).A message of one signal unit length is called **single unit message (SUM)** and one with multiple signal units as **multiunit message (MUM)**.

| | | | |
|---|---|---|---|
| Header | Signallling information | Circuit label | Error check |

(a) Signal unit message

| | | | |
|---|---|---|---|
| Header | Signallling information | Circuit label | Error check |
| Sub Header | Length | Other signalling information | Error check |
| Sub Header | Length | Address digits | Error check |

(b) Three unit message

**Figure 3.2: Typical CCS signalling message formats**

## CCS Implementation

Common channel signalling may be implemented in two ways. They are

(*a*) Channel associated mode

(*b*) Channel Non-Associated mode

## Channel associated mode

In associated CCS signalling mode, there is a direct link between two exchanges. In this mode, the signalling path passes through the same set of switches as does the speech path. Network topologies of the signalling network and the speech network are the same.

**Channel Non-Associated mode**

In non-associated CCS signalling, there is separate control of the networks from the switching machines themselves. In multiexchange network, signal message passing through several intermediate nodes is referred as **non-associated signalling**.

**Comparison of Inchannel Signalling and Common Channel signalling**

| **Inchannel signaling** | **Common channel signaling** |
|---|---|
| Automatic propagation of signalling information enables the simultaneous process and release of associated facilities. | Signalling information must be delayed 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 equipment 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. |

**3.3 SIGNALLING SYSTEM 7 (SS7)**

Signaling system 7 (SS7) has been designed to be an open ended CCS standard that can be used over a variety of digital circuit switched networks.

**SS7 Network Architecture**

The SS7 architectural principles are similar to the OSI architecture proposed for data Networks. The SS7 network is an interconnected set of network elements that is used to exchange messages in support of telecommunication functions. SS7 architecture is shown in figure 3.7

**Message transfer part.** The MTP provides a reliable service for routing messages through the SS7 network. The MTP is divided into three levels as described below.

- **MTP level 1(Signalling data link).** The lowest level MTP 1 is equivalent to the OSI physical layer. MTP level 1 defines the physical, electrical and functional characteristics of the digital signalling link.
- **MTP level 2(Signalling Link).** MTP level 2 provides link layer functionality. The main purpose of this layer is to turn a potentially unreliable physical link into a reliable data

link. When an error occurs on a signaling link the message is retransmitted. It is equivalent to OSI data link layer.

- **MTP level 3(Signaling network).** This level is equivalent in function to the OSI network layer. It extends the functionality provided by MTP level to provide network layer functionality.

**Network Service Part**

It includes MTP and Level 4 functions.MTP functions are already discussed. Level 4 functions are described below.

1. **ISDN User Part (ISUP).** ISUP is used for both ISDN and Non-ISDN calls. The ISUP defines the protocol used to setup, manage, and release trunk circuits that carry voice and data between terminating line exchanges. It manages the trunk artwork on which they rely.
2. **Telephone Use Part (TUP).** The TUP is involved in response to actions by a subscriber at a telephone. The control signalling associated with TUP deal with establishment, maintenance, and termination of telephone calls.
3. **Signalling Connection Control Part (SCCP).** SCCP provides connectionless and connection-oriented network services . The SCCP allows subsystems such as service call processing, calling-card processing, advanced intelligent network (AIN) and custom local area signalling services (CLASS) to be addressed explicitly.
4. **Transaction Capabilities Application Part (TCAP).** TCAP supports the exchange of non-circuit related data between applications across the SS7 network using the SCCP connectionless service.
5. **Operations, Maintenance and Administration part (OMAP).** OMAP module deals with messages relating to the network management, operations and maintenance. This is the area for future definitions.

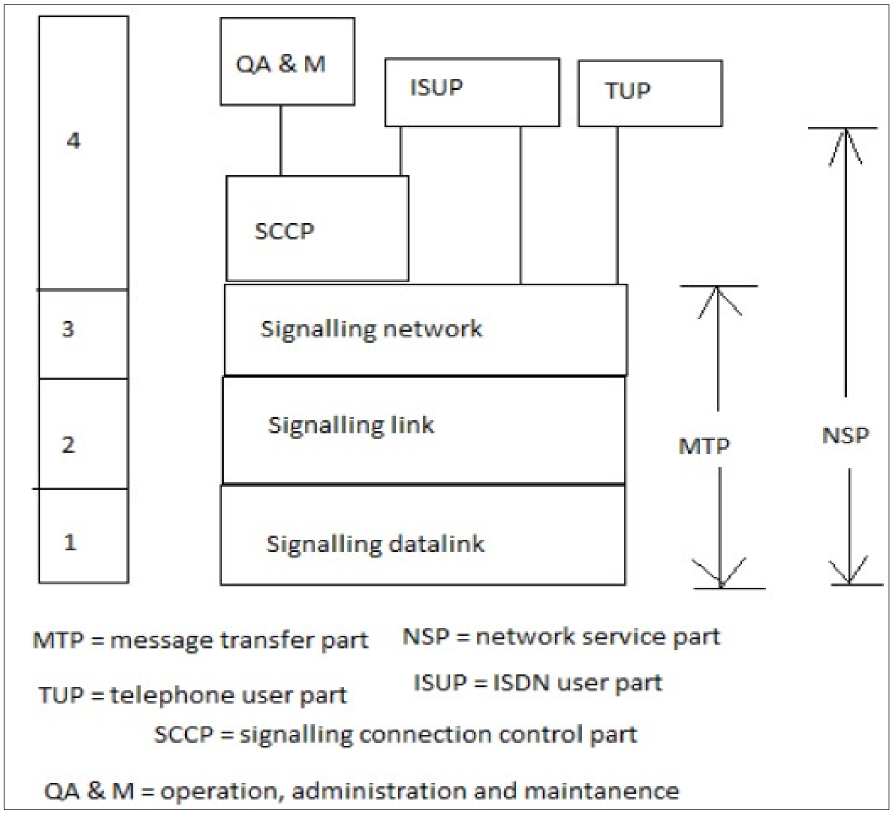
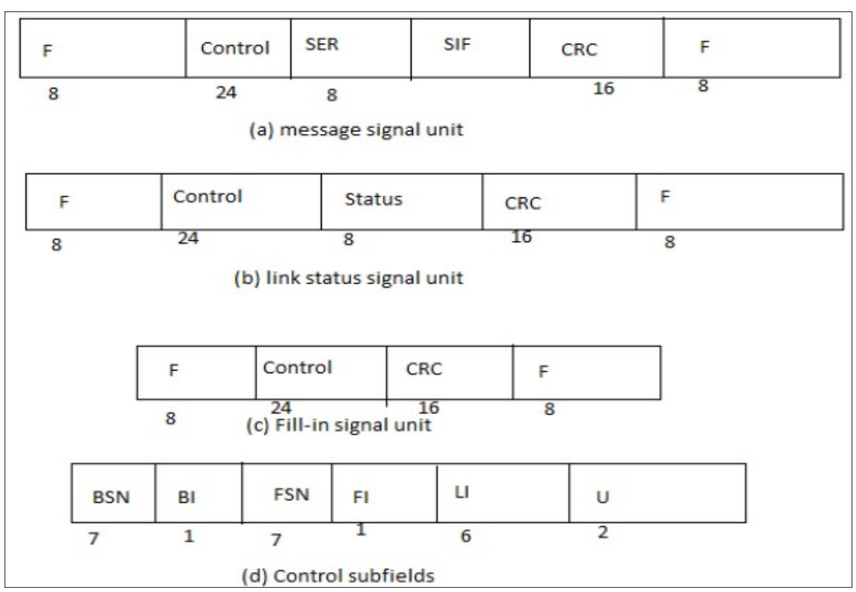



**Figure 3.5: Architecture of SS7**




F = Flag CRC = cyclic redundancy code SIF = signalling information SER = service information field
BI = backward indicator BSN = backward sequence number sequence number LI = length indicator
FI = forward indicator U = unused

**Figure 3.6: Formats of signalling units**

The common fields used in all the signalling units

- **Flag.** The flag indicates the beginning of a new signal unit and implies the end of the previous signal unit (if any). The binary value of the flag is 0111 1110. The same sequence may occur in messages and wrongly interpreted as flags. By the technique known as bit stuffing and unstuffing, false flag can be prevented.
- **Backward Sequence Number (BSN).** The BSN is used to acknowledge the receipt of signal units by the remote signalling point. The BSN contains the sequence number of the signal unit being acknowledged.
- **Backward Indicator Bit (BIB).** A negative acknowledgement is indicated by inverting the BIB bit, which remains unchanged for all subsequent positive acknowledgement.
- **Forward Sequence Number (FSN) and Forward Indicator Bit (FIB).** The FSN contains the sequence number of the signal unit. The FSN identifies the SU uniquely using modulo 128 count.
- **Cyclic Redundancy Check (CRC).** The CRC value is used to detect and correct data transmission errors. The error check field is immediately before the closing flag. It contains 16 bits generated as a CRC code.
- **Length Indicator (LI).** The length indicator indicates the number of octets (8 bit bytes) between itself and the CRC check sum. It serves both as a check on the integrity of the SU and as a means of discrimination between different types of SUS at level 2. Accroding to the protocol only 6 of the 8 bits in the length indicator field are actually used to store this length
- **Signalling Information Field (SIF).** SIF may consist of upto 272 octets and contains the information to be transmitted. The SIF in an MSU contains the routing label and signaling information. LSSUs and FISUs contain neither a routing label nor SIO as they are sent between two directly connected signalling points.

## 3.4 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 telecommunication system has to service the voice traffic and data traffic. Traffic is analyzed by using the following parameters

- **Calling rate.** This is the average number of requests for connection that are made per unit time.
- **Holding time.** The average holding time or service time '$h$' is the average duration of occupancy of a traffic path by a call

- **Busy hour.** Continuous 60 minutes interval for which the traffic volume or the number of call attempts is greatest.
- **Peak busy hour.** It is the busy hour each day varies from day to day, over a number of days.
- **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.
- **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.

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.
- **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

## 3.5 GRADE OF SERVICE AND BLOCKING PROBABILITY

### 3.5.1 Grade of Service

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.**

GOS = BlockedBusyHourCalls / OfferedBusyHourCalls

$GOS = (A - A_0)/A$

Where $A_o$ = carried traffic

A = offered traffic

$A-A_o$ = loss traffic

The smaller the value of grade of service, the better is the service. 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

(*b*) Waiting system.

**Lost 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.

**Waiting system**. In waiting system, a GOS objective could be either the percentage of calls which are delayed or the percentages which are delayed more than a certain length of time.

**Example :** During a busy hour, 1400 calls were offered to a group of trunks and 14calls were lost. The average call duration has 3 minutes. Find (a) Traffic offered (b) Traffic carried (c) GOS and (d) The total duration of period of congestion.

**Given data:** n = 1400

h = 3

T = 60, lost calls = 14

**Sol.**

(a) Traffic offered A = (1400 X 3) / 60 = 70E

(b) Traffic carried A0 = (1386X3) / 60 = 69.3E

(c) GOS = (A – A0) / A

where A – A0 = 70 – 69.3 = 0.7 E (lost traffic)

GOS = 0.7 / 69.3 = 0.01

(d) Total duration = 0.01 × 3600 = 36 seconds.

### 3.6 MODELLING SWITCHING SYSTEM:

In a telecommunication network the call generation by the subscribers and the behavior of the network or the switching system are random process. A random process or a stochastic process is one in which one or more quantities vary with time such that the instantaneous variables predictable with certain probability.

We have four different types of stochastic process namely

i. Continuous time continuous state

ii. Continuous time discrete state

iii. Discrete time continuous state

iv. Discrete time discrete state

- A discrete state stochastic process is called **chain.**
- Random processes whose statistical parameters do not change with time are known as **stationary process**
- The random processes which have identical time and ensemble averages are known as **ergodic process**
- In random process if the mean and variance alone are stationary and other higher order moments may vary with time are known as **wide sense stationary process**

A telecommunication network carries traffic generated by a large number of individual subscribers connected to the network. Subscribers generate calls in a random manner. The call generated by the subscribers and therefore the behaviour of the network or the switching systems in it can be described as a random process. A random or stochastic processes one in which one or more quantities vary with time in such a way that the instantaneous values of the quantities are not determinable precisely but are predictable with certain probability. These quantities are called random variables.

A stochastic process $X(t)$ consists of an experiment with a probability measure $P[\cdot]$ defined on a sample space $S$ and a function that assigns a time function $x(t, s)$ to each outcome $s$ in the sample space of the experiment. A sample function $x(t, s)$ is the time function associated with outcome $s$ of an experiment.

Types of Stochastic Processes

• **Discrete Value and Continuous Value Processes**: $X(t)$ is a discrete value process if the set of all possible values of $X(t)$ at all times $t$ is a countable set $S_X$; otherwise, $X(t)$ is a continuous value process.

• **Discrete Time and Continuous Time Process:** The stochastic process $X(t)$ is a discrete time process if $X(t)$ is defined only for a set of time instants, $tn = nT$, where $T$ is a constant and $n$ is an integer; otherwise $X(t)$ is a continuous time process.

• **Random variables from random processes**: consider a sample function $x(t, s)$, each $x(t1, s)$ is a sample value of a random variable. We use $X(t1)$ for this random variable. The notation $X(t)$ can refer to either the random process or the random variable that corresponds to the value of the random process at time $t$.

**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 the process is known as birth-death (B–D) process.

Types of Stochastic Processes

**Markov process**

A Markov process is a stochastic process with the following properties:

(a.) The number of possible outcomes or states is finite.

(b.) The outcome at any stage depends only on the outcome of the previous stage.

(c.) The probabilities are constant over time.

If $X_0$ is a vector which represents the initial state of a system, then there is a matrix M such that the state of the system after iteration is given by the vector.

## 3.7 Blocking models

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 find all available lines busy, the call is said to be **blocked.** The blocked calls can be handled in two ways.

1) Loss System
2) Delay System

**Loss System**

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.

## 3.8 Delay 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.**

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

There are three models of loss systems. They are

1. Lost calls cleared (LCC)
2. Lost calls returned (LCR)
3. Lost calls held (LCH)

All the three models are described in this section.

**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.

**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. Including the retried calls, the offered traffic now comprises two components *via* new traffic and retries 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 avoids 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.

**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.

## 3.9 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 check out 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 $k \leq N$, $k$ 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.

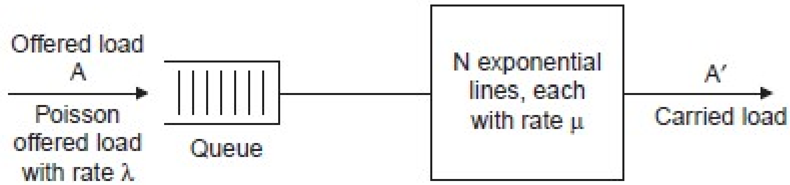



**Figure 3.7 Queuing model**

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.