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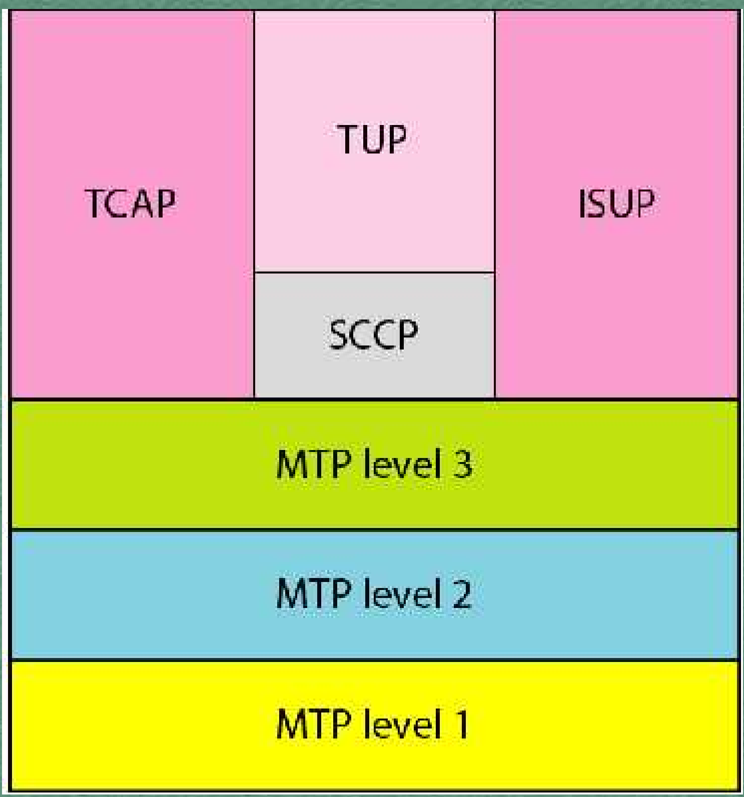
TCS5

SIGNALLING IN TELECOMMUNICATION



Indian Railways Institute of
Signal Engineering and Telecommunications

SECUNDERABAD - 500 017

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INDIAN RAILWAYS INSTITUTE OF SIGNAL ENGINEERING & TELECOMMUNICATIONS, SECUNDERABAD - 500 017

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SIGNALLING IN TELECOMMUNICATION

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CHAPTER-1

SIGNALLING IN TELECOMMUNICATIONS

1.0 INTRODUCTION

A telecommunication network establishes and releases temporary connections, in accordance with the instructions received from subscriber lines and inter-exchange trunks. Therefore, it is necessary to interchange information between an exchange and its external environment, i.e., between subscriber lines and exchange, and between different exchanges. Though these signals may differ widely in their implementation, they are collectively known as telephone signals.

A signaling system uses a language which enables two switching equipments to converse for the purpose of setting up calls. Like any other language, it possesses a ‘vocabulary’ of varying size and varying precision, i.e., a list of signals which may also vary in size, and a ‘syntax’ in the form of, more or less, a complex set of rules, governing the assembly of these signals.

This chapter discusses the growth of signaling and various type of signalling codes used in Indian telecommunication.

1.1 Types of Signalling Information

1.1.1 The signaling information can be categorized under four main heads.

i. Call Request and Release Information
ii. Selection (Address) Information
iii. End-of-Selection Information
iv. Supervisory Information

1.1.2 Call Request and Release Information

i. Call request information, i.e., calling subscriber off-hook, or seizure signal on an incoming trunk, indicates a new call. On its receipt, the exchange connects an appropriate equipment for receiving address information (called number).

ii. Release information, i.e., subscriber on-hook or release signal on a trunk, indicates that the call is over. The exchange releases all the equipment, held busy for the call, and clears up any other information used for setting and holding the call.

1.1.3 Selection (Address) Information

When the exchange is ready to receive the address information, it sends back a request which is known as proceed-to-send (PTS) signal in trunk signaling and dial-tone in subscriber signalling.

Address information, essentially, comprises of full or part of the called subscriber’s number and possibly with additional service data.

1.1.4 End-of-Selection Information

This information indicates the status of the called line, or the reason for non completion of the call attempt, essentially indicating called line free of busy.

1.1.5 Supervisory Information

It specifies the on/off-hook condition of a called subscriber, after the connection has been set up.

i. Called subscriber off-hook
   Called subscriber has answered and charging may commence.
ii. Called subscriber on-hook
   Called subscriber has hung up to terminate the call, and the call is disconnected after a time-delay, if the calling subscriber does not hang up.

The on/off-hook conditions of the calling subscriber are covered by call request and release information.

1.2 Call connection

The interchange of signalling information can be illustrated with the held of a typical call connection sequence. The circled number in Fig. 1 correspond to the steps listed below

i. A request for originating a call is initiated when the caller lifts the hand set.
ii. The exchange sends dial-tone to the caller to indicate to caller to start dialing.
iii. The called number is transmitted to the exchange, when the caller dials the number.

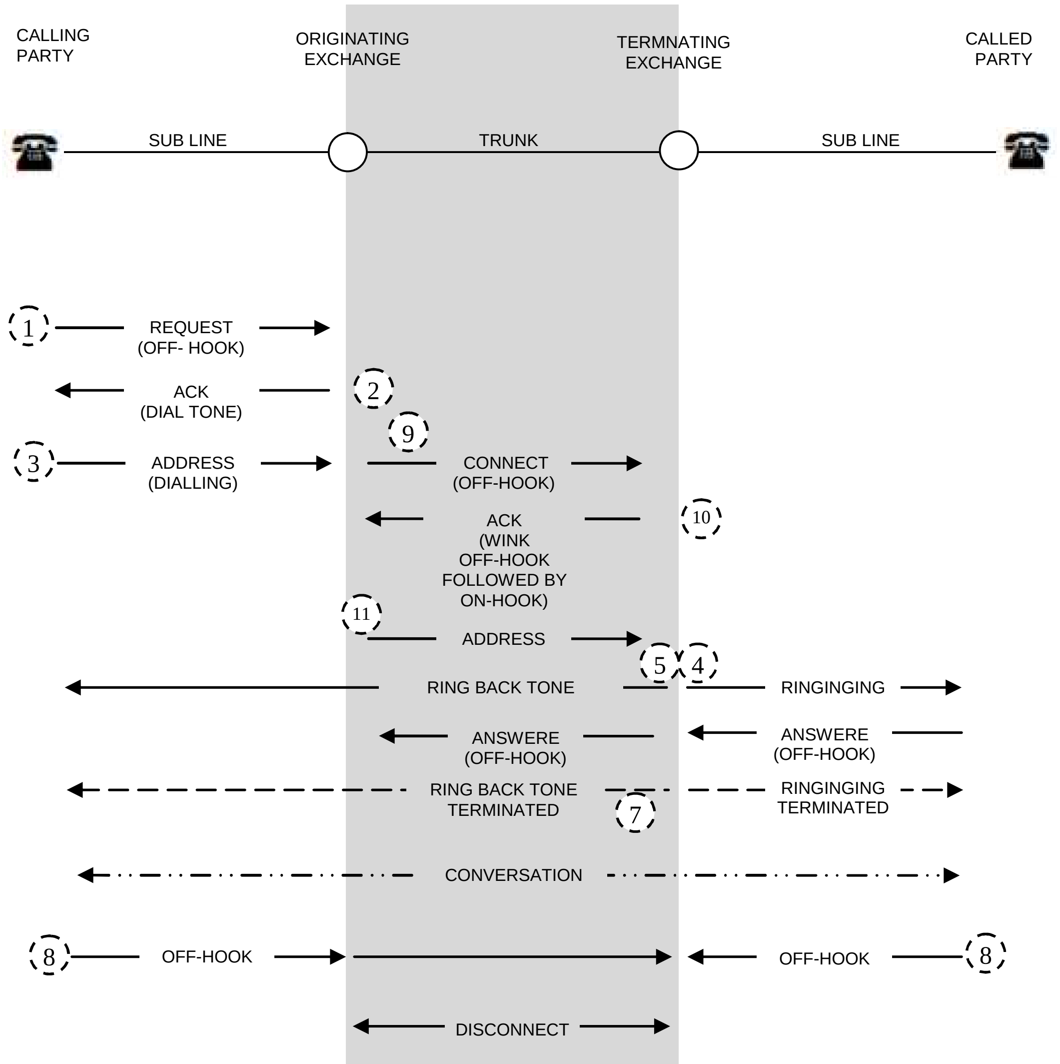


Fig. 1.1 Signalling on a typical completed call

iv. If the called number is free, the exchange sends ringing current to him.
v. Feed-back is provided to the caller by the exchange by sending
vi. Ring-back tone, if the called subscriber is free (shown in Fig.1)
vii. Busy tone, if the called subscriber is busy (not shown in figure)
viii. Recorded message, if provision exists, for non-completion of call due to some other constraint (not shown in figure).
ix. The called subscriber indicates acceptance, of the incoming call by lifting the handset.
x. The exchange, recognizing the acceptance, terminates the ringing current and the ring back tone, and establishes a connection between the calling and called subscribers.
xi. The connection is released when either subscriber replaces the handset.
When the called subscriber is in a different exchange, the following inter-exchange trunk signal functions are also involved, before the call can be set up.
xii. The originating exchange seizes an idle inter-exchange trunk, sends an off-hook signal on the trunk, and requests a digit register at the terminating exchange.
xiii. The terminating exchange sends a 'wink' (an off-hook followed by an on-hook signal), to indicate a register ready or start-dial status to the originating exchange, to request transmission of digits.
xiv. The originating exchange sends the digits. The steps iv to viii are, then, performed to set up the call.

1.3 Signalling

Telephony started with the invention of magneto telephones which used a magneto to generate the ringing current, the only signal, sent over a dedicated line between two subscribers.

The need for more signals was felt with the advent of manual switching. Two additional signals were, therefore, introduced to indicate call request and call release. The range of signals increased further with the invention of electro-mechanical automatic exchanges and is still growing further at a very fast pace, after the advent of SPC electronic exchanges.

1.4 SUBSCRIBER LINE SIGNALLING

1.4.1 Calling Subscriber Line Signalling

In automatic exchanges, the power is fed over the subscriber's loop by the centralized battery at the exchange. Nominally, it is – 48V. The power is fed irrespective of the state of the subscriber, viz., idle, busy or talking.

1.4.2 Call Request

When the subscriber is idle, the line impedance is high. The line impedance falls, as soon as, the subscriber lifts the hand-set, resulting in increase of line current. This is detected as a new increase of line current. This is detected as a new call signal and the exchange, after connecting an appropriate equipment to receive the address information, sends back dial-tone signal to the subscriber.

1.4.3 Address Signal

After the receipt of the dial-tone signal, the subscriber proceeds to send the address digits. The digits may be transmitted either by Decadic dialing or Multifrequency Pushbutton dialing.

i. Decadic Dialling

The address digits may be transmitted as a sequence of interruption of the DC loop by a rotary dial or a decadic pushbutton keypad.

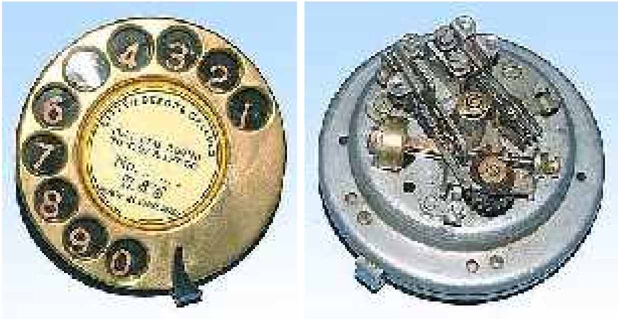


Fig.1.2 Decading dialler

The number of interruption (breaks), indicate the digit, except 0, for which there are 10 interruptions. The rate of such interruptions is 10 per second and the make/break ratio is 1:2. There has to be a inter-digital pause of few hundred milliseconds to enable the exchange to distinguish between consecutive digits (refer fig 1.3). The methods is, therefore, relatively slow and signals cannot be transmitted during the speech phase.

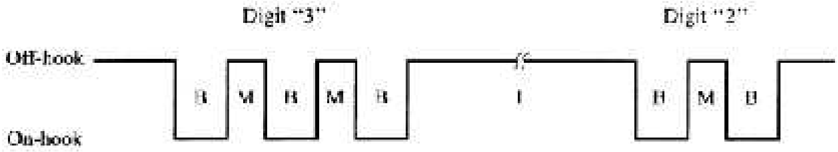


Fig. 1.3 Dial pulse

ii. Multi-frequency Push button Dialling

This method overcomes the constraints of the decadic dialing. It uses two sets of four voice frequencies. Pressing a button (key), generates a signal comprising of two frequencies, one from each group. Hence, it is also called Dual-Tone-Multi-Frequency (DTMF) dialing. The signal is transmitted as long as the key is kept pressed. This provides 16 different combinations. As there are only 10 digits, at present the highermost frequency, viz., 1633 Hz, is not used and only 7 frequencies are used, as shown in Fig. 1.4.

Fig. 1.4 DTMF Key pad and frequencies

By this method, the dialing time is reduced and almost 10 digits can be transmitted per second. As frequencies used lie in the speech band, information may be transmitted during the speech phase also, and hence, DTMF telephones can be used as access terminals to a variety of systems, such as, computers with voice output. The tones have been so selected as to minimize harmonic interference and probability of simulation by human voice.

1.4.4 End-of-Selection Signal

The address receiver is disconnected after the receipt of complete address. After the connection is established or if the attempt has failed, the exchange sends any one of the following signals.

i. Ring-back tone to the calling subscriber and ringing current to the called subscriber, if the called line is free.
ii. Busy –tone to the calling subscriber, if the called line is busy or otherwise inaccessible.
iii. Recorded announcement to the calling subscriber, if the provision exists, to indicate reasons for call failure, other than called line busy.

Ring-back tone and ringing current are always transmitted from the called subscriber's local exchange and busy-tone and recorded announcements, if any, by the equipment as close to the calling subscriber as possible, to avoid unnecessary busying of equipment and trunks.

1.4.5 Answer Back Signal

As soon as the called subscriber lifts the handset, after ringing, a battery-reversal signal is transmitted on the line of the calling subscriber. This may be used to operate special equipment attached to the calling subscriber, e.g., short-circuiting the transmitter of a CCB, till a proper coin is inserted in the coin-slot.

1.4.6 Release Signal

When the calling subscriber releases, i.e., goes on-hook., the line impedance goes high. The exchange recognizing this signal, releases all equipment involved in the call. This signal is normally of more than 500 milliseconds duration.

1.4.7 Permanent-Line (PG) Signal

Permanent-line (PG) signal is sent to the calling subscriber, if he fails to release the call even after the called subscriber has gone on-hook, and the call is released after a time delay. The PG signal may also be sent, in case the subscriber takes too long to dial. It is normally busy-tone.

1.5 Called Subscriber Line Signals

i. Ring Signal

On receipt of a call to the subscriber, whose line is free, the terminating exchange sends the ringing current to the called telephone. This is, typically, 25 or 50 Hz with suitable interruptions. Ring-back tone is also fed back to the called subscriber by the terminating exchange.

ii. Answer Signal

When the called subscriber lifts the hand-set on receipt of ring, the line impedance goes low. This is detected by the exchange which cuts-off the ringing current and ring-back tone.

iii. Release Signal

If after the speech phase, the called subscriber goes on-hook before the calling subscriber, the state of line impedance going high from a low value, is detected. The exchange sends a permanent line signal to the calling subscriber and releases the call after a time delay, if the calling subscriber fails to clear in the meantime.

1.6 Register Recall Signal

With the use of DTMF telephones, it is possible to enhance the services, e.g., by dialing another number while holding on to the call in progress, to set up a call to a third subscriber. The signal to recall the dialing phase during the talking phase, is called Register Recall signal. It consists of, interruption of the calling subscriber's loop for a duration less than the release signal. It may be of 200 to 320 milliseconds duration.

1.7 Inter-Exchange Signalling

1.7.1 Inter-exchange signaling can be transmitted done over each individual inter-exchange trunk. The signals may be transmitted using the same frequency band as for speech signals (in-band signaling), or using the frequencies outside this band (out-of-band signaling). The signaling may be

i. Pulsed

The signal is transmitted in pulses. Change from idle condition to one of active states for a particular duration characterizes the signal, e.g., address information.

ii. Continuous

The signal consists of a transition from one condition to another. A steady-state condition does not characterises any signal.

iii. Compelled

It is similar to the pulsed mode but the transmission is not of a fixed duration but continues till acknowledgement of the receiving unit is received back at the sending unit. It is a highly reliable mode of signal transmission and enables transmission of complex signals.

1.7.2 Line Signals

The simplest, cheapest, and most reliable system of signaling on trunks, was DC signaling, also known as metallic loop signaling, exactly the same as used between the subscriber and exchange. i.e.,

i. Circuit seizure/release, corresponding to off/on-hook signal of the subscriber.

ii. Address information in the form of decadic pulses

1.7.3 In-Band and Out-of-Band Signals

Exchanges separated by long distances cannot use any form of DC line signalling. Suitable interfaces have to be interposed between them for conversion of the signals into certain frequencies, to enable them to be carried over long distance. A single frequency (SF) may be used to carry the on/off-hook information. The dialing pulses can also be transmitted by pulsing of the states. The number of signals is small and they can be transmitted in-band or out-of-band. The states involved are shown in Table 1.1

Table 1. 1 Single Frequency Signalling States

State	Outgoing	Incoming
Tone-off	Seizure	Idle/busy
Tone-on	Idle	Ringing

For In-band signaling, the tone frequency is chosen to be 2600 Hz or 2400 Hz. As the frequency lies within the speech band, simulation of tone-on condition indicating 'end-of-call' signal by the speech, has to be guarded against, for pre-mature disconnection.

Out-of-band signaling overcomes the problem of 'tone-on condition imitation' by the speech, by selecting a tone frequency of 3825 Hz which is beyond the speech band. However, this adds up to the hard-ware costs.

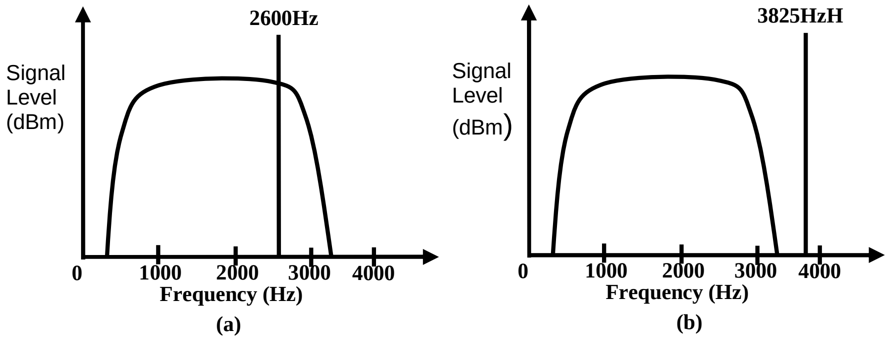


Fig.1.5 SF (Single Frequency) Signalling (a) In-band (b) Out-of-band

1.7.4 E & M Signals

E & M lead signaling may be used for signaling on per-trunk basis. An additional pair or circuit, reserved for signaling is employed. One wire is dedicated to the forward signals (M-wire for transmit) which corresponds to receive wire of the destination exchange, and the other wire dedicated to the backward signals (E-wire for recEive) which corresponds to transmit wire of the destination exchange. The signaling states are shown in Table 1.2.

Table 1.2. E & M Signalling States

State	From Switching System (M-lead)	To Switching System (E-lead)
On hook	Earth	Open
Off hook	-48V	Earth

This type of signaling is normally used conjunction with an interface to change the E & M Signals into frequency signal to be carried along with the speech.

1.7.5 Register Signals

It was, however, felt that the trunk service could not be managed properly without the trunk register which basically is an address digit receiver. With such development, the inter-exchange signaling was sub-divided into two categories.

i. Line signalling, in which the signals operate throughout the duration of call, and
ii. Register signalling, during the relative short phase of setting up the call essentially for transmitting the address information.

In other words, register signals are interchanged between registers during a phase between receipt of trunk seizure signal and the exchange switching to the speech phase. These signals are proceed-to-send (PTS) signal, address signals, and signals indicating the result of the call attempt.

The register signals may be transmitted in-band or out-of-band. However, in the latter case, the signaling is relatively slow and only limited range of signals may be used. For example, a single out-of-band frequency may be selected and information sent as pulses.

In-band transmission can be used easily as there can be no possible interference with the speech signals. To reduce transmission time and to increase reliability, a number of frequencies are used in groups. Normally, 2 out of 6 frequencies are used. To make the system more reliable, Compelled Sequence is used. Hence, this system is normally called Compelled Sequence Multi-Frequency (CSMF) signaling, as shown in Fig.3. In CCITT terminology , it is termed as R2 system.

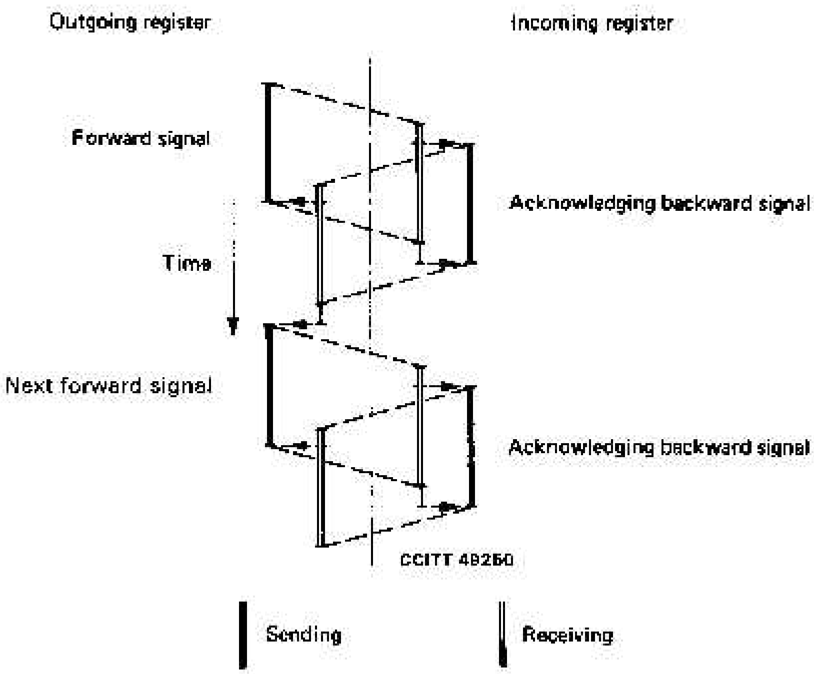


Fig.1.6 Compelled sequence Multi frequency signaling (CSMF) process

As the frequencies need be transmitted only for a short duration to convey the entire information, the post dialing delay is reduced.

1.7.6 When more than two exchanges are involved in setting up the connection, the signaling may be done in either of the two modes,

i. End-to-end signalling

The signaling is always between the ends of the connection, as the call progresses. Considering a three exchange, A-B-C, connection, initially the signalling is between A-B, then between A-C after the B-C connection is established.

ii. Link-by-link signaling

The signaling is always confined to individual links. Hence, initially the signaling is between A-B, then between B-C after the B-C connection is established.

Generally, supervisory (or line) and subscriber signaling is necessarily on link-by-link basis. Address component may be signaled either by end-to-end or link-by-link, depending upon the network configuration.

1.7.7 R2 Signalling

CCITT standardized the R2 signalling system to be used on national and international routes. However, the Indian environment requires lesser number of signals and hence, a slightly modified version is being used.

There is a provision for having 15 combinations using two-frequencies out of six frequencies, viz., 1380, 1500, 1620, 1740, 1860 and 1980 Hz, for forward signals and another 15 combination using two frequencies out of six frequencies, viz., 1140, 1020, 900, 780, 660 and 540 Hz, for backward signals. In India, the higher most frequency in the forward group, i.e., 1980 Hz, and the lowermost frequency in the backward group, i.e., 540 Hz, are not used. Thus, there are 10 possible combinations in both the directions. The weight codes for the combinations used, are indicated in Table 1.3 and the significance of each signal is indicated in Table 4 and 5.

Table 1.3. Signal Frequency Index and Weight Code

Signal Frequency (Hz)					
Forward	1380	1500	1620	1740	1860
Backward	1140	1020	900	780	660
Index	f 0	f 1	f 2	f 3	f 4
Weight Code	0	1	2	4	7

Table 1.4. Forward Signals

Signal	Weight Code	Group I	Group II
1	0+1	Digit 1	Ordinary subscriber
2	0+2	Digit 2	Subscriber with Priority
3	1+2	Digit 3	Test/Mtce. Equipment
4	0+4	Digit 4	STD Coin Box
5	1+4	Digit 5	Operator
6	2+4	Digit 6	Spare
7	0+7	Digit 7	Spare
8	1+7	Digit 8	Spare
9	2+7	Digit 9	Spare
10	4+7	Digit 0	Spare

Table 1.5. Backward Signal

Signal No	Weight Code	Group A	Group B
1	0+1	Send next digit	Spare
2	0+2	Restart	Changed number
3	1+2	Address Complete, changeover to reception of Group B Signals	Called line busy
4	0+4	Calling line identification for malicious calls	Congestion
5	1+4	Send calling subscriber's category	Number unobtainable
6	2+4	Set up speech connection	Called line free, with metering
7	0+7	Send last-but-3 digit	Spare
8	1+7	Send last-but-2 digit	Spare
9	2+7	Send last-but-1 digit	Spare
10	4+7	Spare	Spare

Note: Signals A2, and A6 to A9 are used in Tandem working only.

As can be seen from the tables

i. Forward signals are used for sending the address information of the called subscriber, and category and address information of the calling subscriber.
ii. Backward signals are used for demanding address information and caller's category, and for sending condition and category of called line.

R2 signalling is fully compelled and the backward signal is transmitted as an acknowledgement to the forward signal. This speeds up the interchange of information, reducing the call set up time. However, the satellite circuits are an exception and semi-compelled scheme may only be used due to long propagation time.

Register signals may be transmitted on end-to-end basis. It is a self checking system. Each signal is acknowledged appropriately at the other end after the receiver checks the presence of only 2 and only 2 out of 5 proper frequencies.

1.7.8 An example of CSMF signaling between two exchanges may be illustrated by considering a typical case. The various signals interchanged after seizure of the circuit using DC signaling, are

i. Originating exchange sends first digit.
ii. Receipt of the digit is acknowledged by the terminating exchange by sending A5 (demanding the caller's category)
iii. A5 is acknowledged by sending any of II-1 to II-5 by the originating exchange.
iv. Terminating exchange acknowledges this by A1, demanding for next digit.
v. Originating exchange acknowledges A1 by sending any of II-1 to II-10, Sending the digit.

vi. The digits are sent in succession by interchange of steps v and vi.

vii. On receipt of last digit, the terminating exchange carries out group and line selection and then sends A3, indicating switching over to group B signals.

viii. This is acknowledged by the originating exchange by sending the caller's category again.

ix. The terminating exchange acknowledges by sending the caller's line condition by sending any of B2 to B6.

x. In response to B6, the originating exchange switches through the speech path and registers are released. Alternatively, in response to B2 to B5, the registers are released and appropriate tone is fed to the calling subscriber by the originating exchange.

1.8 Digital Signalling

All the systems discussed so far, basically, are on per-line or per-trunk basis, as the signals are carried on the same line or trunk. With the emergence of PCM systems, it was possible to segregate the signaling channel from the speech channel.

Inter-exchange signaling can be transmitted over a channel directly associated with the speech channel, Channel-Associated Signalling (CAS), or over a dedicated link common to a number of channels, Common Channel Signalling (CCS). The information transmitted for setting up and release of calls is same in both the cases. Channel Associated Signalling requires the exchanges to have access to each trunk via the equipment which may be decentralized, whereas in Common Channel. Signalling, the exchange is connected to only a limited number of signaling links through a special terminal.

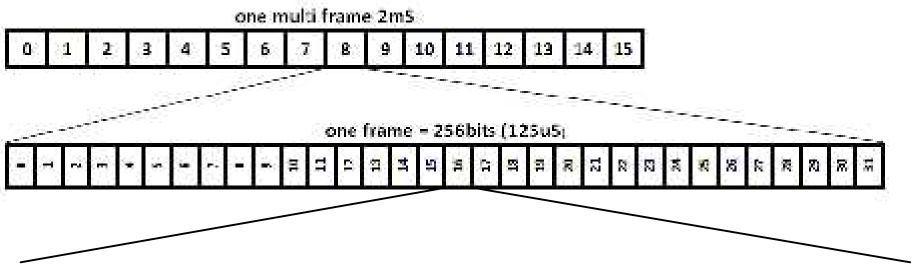
1.9 Channel-Associated Signalling

In the PCM systems the signaling information is conveyed on a separate channel which is rigidly associated with the speech channel. Hence, this method is known as Channel-Associated Signalling (CAS). Though the speech sampling rate is 8 KHz, the signals do not change as rapidly as speech and hence, a lower sampling rate of 500 Hz, for digitization of signals can suffice. Based on this concept, TS16 of each frame of 125 microseconds, is used to carry signals of 2 speech channels, each using 4 bits.

Hence, for a 30 channel PCM system, 15 frames are required to carry all the signals. To constitute a 2 millisecond multiframe of 16 frames, F0 to F15. TS16 of the frame F0 is used for multiframe synchronization. TS16 of F1 contains signal for speech channels 1 and 16 being carried in TS1 and TS17, respectively, TS16 of F2 contains signals of speech channels 2 and 17 being carried in TS2 and TS18, respectively, and so on (Ref fig 1.7). Both, line signals and address information can be conveyed by this method.

Although four bits per channel are available for signaling, only two bits are used. As the transmission is separate in the forward and backward direction, the bits in the forward link are called af and bf, and those in the backward link are called ab and bb. Values for these bits are assigned as shown in Table 1.6.

As the dialing pulses are also conveyed by these conditions, the line state recognition time is, therefore, above a threshold value. The bit bf is normally kept at 0, and the value 1 indicates as.



CHANNEL 16								
Channel 16 bit number	A	B	C	D	A	B	C	D
Frame 1	Relative to channel 1				Relative to channel 17			
Frame 2	Relative to channel 2				Relative to channel 18			

Frame 15	Relative to channel 15				Relative to channel 31			

Fig 1.7 30 channel PCM system

Table 1.6. Bit Values in Digital Signalling

State	Bit Value			
	Forward		Backward	
	af	bf	ab	bb
Idle	1	0	1	0
Seizure	0	0	1	0
Seizure Acknowledged	0	0	1	1
Answered	0	0	0	1
Clear Forward	1	0	0/1	0
Clear Back	0	0	1	1
Blocked	1	0	1	1

However, the utilization of such a dedicated channel for signaling for each speech channel is highly inefficient, as it remains idle during the speech phase. Hence, another form of signaling, known as common-channel signaling, evolved.

1.10 EFFECTS OF NUMBERING ON SIGNALING

Numbering, the assignment and use of telephone numbers, affects signaling as well as switching. There is “uniform” numbering and “nonuniform” numbering. How does each affect signaling? Uniform numbering can simplify a signaling system. Most uniform systems in the nontoll or local-area case are based on seven digits, although some are based on six. The last four digits identify the subscriber. The first three digits (or the first two in the case of a six-digit system) identify the exchange. Thus the local exchange or transit exchanges know when all digits are received. There are two advantages to this sort of scheme:

1. The switch can proceed with the call once all digits are received because it "knows" when the last digit (either the sixth or seventh) has been received.
2. "Knowing" the number of digits to expect provides inherent error control and makes "time out" simpler.4

For non-uniform numbering, particularly on direct distance dialing in the international service, switches require considerably more intelligence built in. It is the initial digit or digits that will tell how many digits are to follow, at least in theory. However, in local or national systems with non uniform numbering, the originating register has no way of knowing whether it has received the last digit, with the exception of receiving the maximum total used in the national system. With non-uniform numbering, an incompletely dialed call can cause a useless call setup across a network up to the terminating exchange, and the call setup is released only after time out has run its course. It is evident that with non-uniform numbering systems, national (and international) networks are better suited to signaling systems operating end to end with good features of backward information, such as the R-2 system.

1.11 ISDN Q-Interface Signaling Protocol Q.931

Private branch exchanges (PBXs) are crucial communication components in any organization. As an organization grows and expands to multiple locations, it needs to establish multiple PBXs to keep the locations connected. The primary challenge in operating these multiple PBXs is to get them to work as a single entity so that the user experience is consistent and reliable irrespective of location. For example, when a single voicemail system is installed at headquarters, it needs to be accessible to all remote employees as if they are accessing it locally. To achieve such operation, the PBXs must be connected (Figure 1.8)

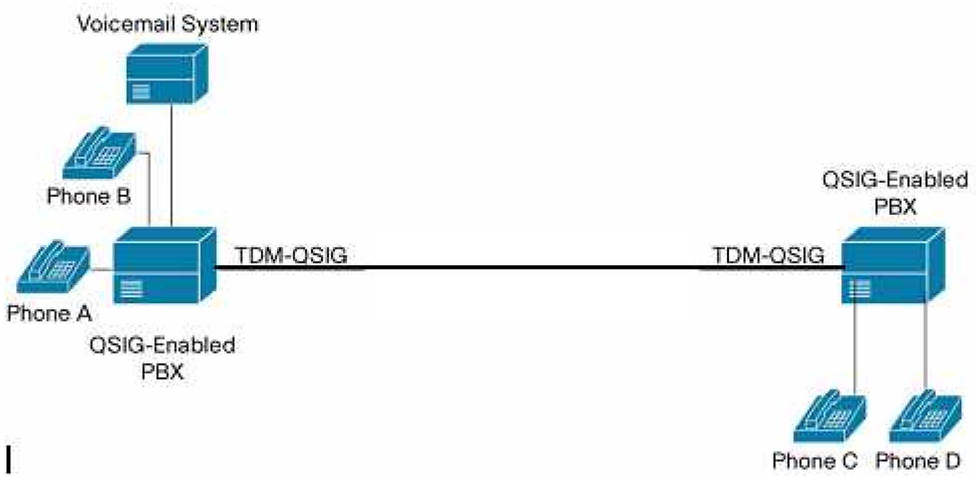


Fig. 1.8 ISDN Q-Interface Signaling

QSIG is a protocol used to connect PBXs within an enterprise with supplementary services. This protocol is designed to be independent of its own transport mechanism as well as to be independent of any means used to transport speech or data in calls established using QSIG. An example of a typical QSIG deployment is a configuration of primary-rate leased lines so that signaling takes one of the 24 (T1) or 30 (E1) 64-kbps channels while the rest act as 64-kbps bearers for speech or data. The QSIG protocol acts as a variant of ISDN D-channel voice signaling and is based on the ISDN Q.921 and Q.931 standards, setting a worldwide standard for PBX interconnection.

1.11.1 QSIG Overview

QSIG is an internationally standardized signaling protocol for use in corporate and enterprise voice and integrated services networks. Typically employed between PBXs, the QSIG protocol is used to establish and release calls (basic services) and to manage supplementary services between PBXs.

In a basic QSIG call, a user on one PBX can place a call to a user on another (remote) PBX. The called party receives the calling party's name or number when the call rings, and the calling party receives the called party's name and number when the called party's phone rings on the remote PBX. Additionally, the QSIG protocol helps provide supplementary and additional network features as long as the corresponding set of QSIG features is supported at both ends of the call. Here are some standard QSIG supplementary services available on various PBXs:

- Multiple Subscriber Number
- Call Waiting
- Calling-Line Identification Presentation (CLIP)
- Calling-Line Identification Restriction (CLIR)
- Connected-Line Identification Presentation (COLP)
- Connected-Line Identification Restriction (COLR)
- Malicious Call Identification
- Call Hold
- Advice of Charge
- Three-Way Conference
- Call Diversion
- CFU Supplementary Service
- Path Replacement (ANF-PR)
- Call Transfer by Join (SS-CT)
- Call Completion to Busy Subscriber (CCBS)
- Explicit Call Transfer

1.12 Intelligent Network

The Intelligent Network, typically stated as its acronym IN, is a network architecture intended both for fixed as well as mobile telecom networks. It allows operators to differentiate themselves by providing value-added services in addition to the standard telecom services such as PSTN, ISDN and GSM services on mobile phones.

In IN, the intelligence is provided by network nodes owned by telecom operators, as opposed to solutions based on intelligence in the telephone equipment, or in Internet servers provided by any part. IN is based on the Signaling System #7 (SS7) protocol between telephone network switching centers and other network nodes owned by network operators.

Examples of such services are: Televoting, Call screening, Telephone number portability, Toll free calls/Freephone, Prepaid calling, Account card calling, Virtual private networks (eg : Family group calling), Centrex service (Virtual PBX)

The IN concepts, architecture and protocols were originally developed as standards by the ITU-T which is the standardization committee of the International Telecommunication Union. Prior to this a number of telecommunications providers had proprietary IN solutions. The primary aim of the IN was to enhance the core telephony services offered by traditional telecommunications networks, which usually amounted to making and receiving voice calls, sometimes with call divert. This core would then provide a basis upon which operators could build services in addition to those already present on a standard telephone exchange.

A complete description of the IN emerged in a set of ITU-T standards named Q.1210 to Q.1219, or Capability Set One (CS-1) as they became known. The standards defined a complete architecture including the architectural view, state machines, physical implementation and protocols. They were universally embraced by telecom suppliers and operators, although many variants were derived for use in different parts of the world.

Objective:

1. A signaling system uses a language which enables two switching equipments to converse for the purpose of <u>setting up calls</u>.

2. QSIG is an internationally standardized signaling protocol for use in corporate and enterprise <u>voice and integrated services</u> networks.

3. When the exchange is ready to receive the address information, it sends back a <u>dial-tone</u> in subscriber signalling.

4. As soon as the called subscriber lifts the handset, after ringing, a <u>battery-reversal</u> signal is transmitted on the line of the calling subscriber.

5. <u>Permanent-line (PG)</u> signal is sent to the calling subscriber, if he fails to release the call even after the called subscriber has gone on-hook, and the call is released after a time delay.

Subjective:

1. What are the different types of signaling information is carried in Telecommunication networks?

2. What are the different methods used for interexchange signaling?

3. Write short notes on
   a. E&M signaling
   b. In-band and out -of –band signaling

4. What is channel associated signaling? How it is implemented in PDH frame format? Explain.

5. What is Q signaling? What are its features?

CHAPTER-2

SIGNALLING SYSTEM-7 (SS7)

2.0 Evolution

The SS7 protocols were developed by AT&T since 1975 and defined as standard by ITU-T during 1981 in ITU-T's Q.7XX-series recommendations. SS7 was designed to replace Signalling System #5 (SS5) and Signalling System #6 (SS6) and R2, all of which are ITU standards defined by ITU-T prior to SS7 and were once in widespread international use. SS7 has substantially replaced SS6, SS5, and R2, with the exception that R2 variants are still used in numerous nations. SS5 and earlier used in-band signalling, where the call-setup information was sent by playing special tones into the telephone lines (known as bearer channels in the parlance of the telecom industry). This led to a number of security problems when users discovered on certain telephone switching equipment that they could play these tones into the telephone handset and control the network even without the "special keys" on an operators handset. So-called phreaks experimented with fooling the telephone exchanges by sending their own user-generated signalling tones from small electronic boxes known as blue boxes. Modern designs of telephone equipment that implement in-band signalling protocols explicitly keep the end-user's audio path—the so-called speech path—separate from the signalling phase to eliminate the possibility that the MF tones used for signalling are introduced by the end-user, which defeats the blue-box phreaking technique.

SS7 moved to a system in which the signalling information was out-of-band, carried in a separate signalling channel. This avoided the security problems earlier systems had, as the end user had no connection to these channels. SS6 and SS7 are referred to as so-called Common Channel Interoffice Signalling Systems (CCIS) or Common Channel Signaling (CCS) due to their hard separation of signalling and bearer channels.

2.1 Introduction

Signalling system 7 is architecture for performing out of band signalling in support of the call establishment, billing, routing, and information exchange functions of the publics switched telephone network (PSTN). It identifies function to be performed by a signalling system network and protocol to enable their performance.

2.2 The Role of SS7

The purpose of this chapter is to introduce Signalling System No. 7 (SS7) and give the reader an indication of how it affects the lives of nearly two billion people globally. The chapter begins by providing a brief introduction to the major services that SS7/C7 provides and explains how the protocol has been and will continue to be a key enabler of new telecommunication services. It concludes with an explanation of why SS7/C7 is a cornerstone of convergence.

SS7/C7 is the protocol suite that is employed globally, across telecommunications networks, to provide signalling; it is also a private, "behind the scenes," packet-switched network, as well as a service platform. Being a signalling protocol, it provides the mechanisms to allow the telecommunication network elements to exchange control information.

AT&T developed SS7/C7 in 1975, and the International Telegraph and Telephone Consultative Committee (CCITT) adopted it in 1981 as a worldwide standard. Over the past quarter of a century, SS7 has undergone a number of revisions and has been continually enhanced to support services that are taken for granted on a daily basis.

SS7/C7 is the key enabler of the public switched telephone network (PSTN), the integrated services digital network (ISDN), intelligent networks (INs), and public land mobile networks (PLMNs).

Each time you place and release a telephone call that extends beyond the local exchange, SS7/C7 signalling takes place to set up and reserve the dedicated network resources (trunk) for the call. At the end of the call, SS7/C7 takes action to return the resources to the network for future allocation.

Calls placed between subscribers who are connected to the same switch do not require the use of SS7/C7. These are known as interoffice, intraexchange, or line-to-line calls.

Each time a cellular phone is powered up, SS7/C7-based transactions identify, authenticate, and register the subscriber. Before a cellular call can be made, further transactions check that the cellular phone is not stolen (network dependent option) and qualify permission to place the call (for example, the subscriber may be barred from International usage). In addition, the SS7/C7 network tracks the cellular subscriber to allow call delivery, as well as to allow a call that is already in progress to remain connected, even when the subscriber is mobile.

Although the average person typically uses SS7/C7 several times a day, it is largely unheard of by the general public because it is a "behind the scenes" private network—in stark contrast to IP. Another reason for its great transparency is its extreme reliability and resilience. For example, SS7/C7 equipment must make carrier grade quality standards—that is, 99.999 percent availability. The three prime ways it achieves an industry renowned robustness is by having a protocol that ensures reliable message delivery, self-healing capabilities, and an over-engineered physical network.

Typically, the links that comprise the network operate with a 20–40 percent loading and have full redundancy of network elements. SS7/C7 might well be the most robust and reliable network in existence.

SS7/C7 is possibly the most important element from a quality of service (QoS) perspective, as perceived by the subscriber.

QoS is quickly becoming a key in differentiating between service providers. Customers are changing service providers at an increasing pace for QoS reasons, such as poor coverage, delays, dropped calls, incorrect billing, and other service-related impairments and faults. SS7/C7 impairments nearly always impact a subscriber’s QoS directly. A complete loss of signalling means a complete network outage, be it a cellular or fixed-line network. Even a wrongly provisioned screening rule at a SS7/C7 node in a cellular network can prohibit subscribers from roaming internationally or sending text messages. A loss of one signalling link could potentially bring down thousands of calls. For this reason, the SS7/C7 network has been designed to be extremely robust and resilient.

2.3 Impact of SS7 Network Failure

The critical nature of the SS7 network and the potential impact of failures was demonstrated in January 1990 when a failure in the SS7 software of an AT&T switching node rippled through over 100 switching nodes. The failure caused a nine-hour outage, affecting an estimated 60,000 people and costing in excess of 60 million dollars in lost revenue as estimated by AT&T.

2.4 Performance of SS7

i. To handle the various interfaces and provide the promised services, congestion control and delay control are essential.
ii. SS7 protocols provide congestion control schemes (like link monitoring and diverting traffic/ activating link / de-activating link etc.).

Delays at SPs & STPs are to be within limits prescribed by ITU-T under Q.706, Q.716, and Q.766.

2.5 The special Services are obtained from the Signalling System No. 7

In addition to setting up and releasing calls, SS7 is the workhorse behind a number of telecommunication services, including:

- Telephone-marketing numbers such as toll-free and free phone
- Televoting (mass calling)
- Single Directory Number
- Enhanced 911 (E911)—used in the United States
- Supplementary services
- Custom local area signalling services (CLASS)
- Calling name (CNAM)
- Line information database (LIDB)
- Local number portability (LNP)
- Cellular network mobility management and roaming
- Short Message Service (SMS)
- Enhanced Messaging Service (EMS)— Ring tone, logo, and cellular game Delivery
- Local exchange carrier (LEC) provisioned private virtual networks (PVNs)
- Do-not-call enforcement

2.6 The following sections describe these telecommunications services.

2.6.1 Telephone-Marketing Numbers

The most commonly used telephone-marketing numbers are toll-free calling numbers (800 calling), known as free phone (0800) in the United Kingdom. Because the call is free for the caller, these numbers can be used to win more business by increasing customer response.

Telephone-marketing numbers also provide premium rate lines in which the subscriber is charged at a premium in exchange for desired content.

Examples of such services include adult services and accurate road reports. These services were first introduced by Bell Systems to provide toll-free access calling party to the services and database offered by private party.

i. 800 Services are offered under two plans: -
   800 NXX plan: - First 6 digits of an 800 call are used to select the IXC (Inter-exchange carrier).
ii. 800 Database plan: - Call is looked-up in database to decide appropriate carrier and routing.

Another popular telephone-marketing number is local call, with which a call is charged as a local call even though the distance might be national. In recent years in the United Kingdom, marketing numbers that scarcely alter the call cost have been a popular means of masking geographical location. These numbers allow for a separation between the actual number and the advertised number.

2.6.2 Tele voting

Televoting is a mass calling service that provides an easy method of surveying the public on any imaginable subject. The host (for example, a deejay at a radio station) presents specific questions and the caller uses a telephone keypad to select a choice; the caller's action adds to the vote for that particular choice. The conversation phase is usually limited to a simple, automated "thank you for…" phrase. Televoting can also be used in many other areas, such as responding to fundraising pleas and telephone-based competitions. A single night of televoting might result in 15 million calls. Televoting services represent some of the most demanding—as well as lucrative—call scenarios in today's telephone networks. Revenue generation in this area is likely to grow as customers shift more toward an "interactive" experience, on par with convergence.

2.6.3 Single Directory Number

Another service that uses SS7/C7 and has been deployed in recent years is the single directory number, which allows a company with multiple offices or store locations to have a single directory number. After analyzing the calling party's number, the switch directs the call to a local branch or store.

2.6.4 Touch star services

These are also known as CLASS of services and are controlled by a switch –

i. Call return
ii. Call forwarding
iii. Repeat dialling
iv. Call block
v. Call tracing
vi. Caller ID

2.6.5 Alternate billing services

These services enable calling party to bill a call to a personal number (third party number or calling card or collect card) from any number.

2.6.6 Enhanced 911

E911, which is being deployed across some states in the United States, utilizes SS7 to transmit the number of the calling party, look up the corresponding address of the subscriber in a database, and transmit the information to the emergency dispatch operator to enable a faster response to emergencies. E911 might also provide other significant location information, such as the location of the nearest fire hydrant, and potentially the caller's key medical details. The Federal Communications Commission (FCC) also has a cellular 911 program in progress; in addition to providing the caller's telephone number, this program sends the geographical location of the antenna to which the caller is connected. Enhancement proposals are already underway to obtain more precise location information.

2.6.7 Supplementary Services

Supplementary services provide the subscribers with more than plain old telephony service (POTS), without requiring them to change their telephone handsets or access technology. Well-known supplementary services include three-way calling, calling number display (CND), call waiting, and call forwarding. Note that the exact names of these services might differ, depending on the country and the operator.

Recently, supplementary services have been helpful in increasing operators' revenues since revenues against call minutes have been on the decline. Usually the subscriber must pay a fixed monthly or quarterly fee for a supplementary service.

2.6.8 Custom Local Area Signalling Services (CLASS)

Custom local area signalling services (CLASS) are an extension of supplementary services that employ the use of SS7 signalling between exchanges within a local geographical area. Information provided over SS7 links, such as the calling party number or the state of a subscriber line, enable more advanced services to be offered by service providers. A few examples of CLASS services include:

i. Call block— Stops pre-specified calling party numbers from calling.
ii. Distinctive ringing— Provides a distinct ringing signal when an incoming call originates from a number on a predefined list. This feature is particularly beneficial to households with teenagers.
iii. Priority ringing— Provides a distinct ring when a call originates from a pre-specified numbers. If the called subscriber is busy and has call waiting, the subscriber receives a special tone indicating that a number on the priority list is calling.
iv. Call completion to busy subscriber (CCBS)— If a subscriber who has CCBS calls a party who is engaged in another call, the subscriber can activate CCBS with a single key or sequence. When activated, CCBS causes the calling party's phone to ring when the called party becomes available; when the calling party answers, the called party's phone automatically rings again. This feature saves the calling party from continuously attempting to place a call to a party is still unavailable.

Note that the exact names of these services might differ, depending on the country and the operator. In addition, the term "CLASS" is not used outside of North America.

2.6.9 Calling Name (CNAM)

Calling name (CNAM) is an increasingly popular database-driven service that is only available in the United States at this time. With this service, the called party receives the name of the person calling in addition to their number. The called party must have a compatible display box or telephone handset to use this service. The CNAM information is typically stored in regional telecommunications databases. SS7/C7 queries the database for the name based on the number and delivers the information to the called party's local switch.

2.6.10 Line Information Database (LIDB)

Line information database (LIDB) is a multipurpose database that stores valuable information about individual subscribers to provide feature-based services (it is only available in the United States at this time). Such information might include the subscriber's profile, name and address, and billing validation data. The name and address information can be used to power CNAM, for example. The billing validation data is used to support alternate billing services such as calling card, collect, and third number billing. Alternate billing services allow subscribers to bill calls to an account that is not necessarily associated with the originating line. For example, it can be used to validate a subscriber's calling card number that is stored in the LIDB, designating this as the means of payment. SS7/C7 is responsible for the real-time database query/response that is necessary to validate the calling card before progressing to the call set-up phase.

2.6.11 Local Number Portability (LNP)

Local number portability (LNP) provides the option for subscribers to retain their telephone number when changing their telephone service. There are three phases of number portability:

- i. Service Provider Portability
- ii. Service Portability
- iii. Location Portability

The FCC mandated this feature for fixed-line carriers in the United States as part of the Telecommunications Act of 1996; later that same year, the act was also clarified to cover cellular carriers.

LNP is primarily aimed at stimulating competition among providers by removing the personal inconvenience of changing phone numbers when changing service providers. For example, many businesses and individuals spend relatively large sums of money to print their phone numbers on business cards, letterheads, and other correspondence items. Without LNP, people would have to reprint and redistribute these materials more often. This contributes to the inconvenience and detracts from the profitability of changing the telephone number, thereby making changing providers far more prohibitive.

Since telephone networks route calls based on service provider and geographic numbering plan information, SS7 must figure out where the ported number's new terminating switch is by performing additional signalling before setting the call up. This step should add only a second to the call overhead set-up; however, it is a technically challenging network change because it complicates the process by which SS7 establishes a call behind the scenes.

Cellular networks use SS7 for the same reasons they use fixed line networks, but they place much higher signalling demands on the network because of subscriber mobility. All cellular networks, from 2G (GSM, ANSI-41, and even PDC, which is used in Japan) to 3G (UMTS and cdma2000), use SS7 for call delivery, supplementary services, roaming, mobility management, prepaid, and subscriber authentication.

2.6.12 Short Message Service (SMS)

Short Message Service (SMS) forms part of the GSM specifications and allows two-way transmission of alphanumeric text between GSM subscribers. Although it is just now catching on in North America, SMS has been an unexpected and huge revenue source for operators around the world. Originally, SMS messages could be no longer than 160 alphanumeric characters. Many handsets now offer concatenated SMS, which allows users to send and receive messages up to 459 characters (this uses EMS described below). Cellular operators usually use SMS to alert the subscribers that they have voice mail, or to educate them on how to use network services when they have roamed onto another network. Third party companies offer the additional delivery services of sending SMS-to-fax, fax-to-SMS, SMS-to-e-mail, e-mail-to-SMS, SMS-to-web, web-to-SMS, and SMS notifications of the arrival of new e-mail.

Some European (Spain, Ireland, and Germany, for example) and Asian countries (the Philippines, for example) are rolling out fixed-line SMS, which allows users to send SMS through their fixed phone line to cell phones and vice versa, as well as to other fixed-line SMS-enabled phones, fax machines, e-mail, and specialized web pages. Thus far, each European rollout has also offered SMS-to-voice mail. If a caller sends a text message to a subscriber without fixed-line SMS facility, the SMS is speech-synthesized to the subscriber's and their voice mailbox. Fixed-line SMS requires compatible phones, which are becoming readily available.

SMS is carried on the SS7/C7 network, and it makes use of SS7/C7 for the required signalling procedures.

2.6.13 Enhanced Messaging Service (EMS)

Enhanced Messaging Service (EMS) adds new functionality to the SMS service in the form of pictures, animations, sound, and formatted text. EMS uses existing SMS infrastructure and consists largely of header changes made to a standard SMS message. Since EMS is simply an enhanced SMS service, it uses the SS7/C7 network in the same way; the SS7/C7 network carries it, and it uses SS7/C7 for the required signalling procedures.

EMS allows users to obtain new ring tones, screensavers, pictures, and animations for their cell phones either by swapping with friends or purchasing them online.

Operators have recently begun using EMS for downloading games (from classics like Asteroids, to newer games like Prince of Persia), which can be purchased from operator web sites.

2.6.14 Private Virtual Networks

Although the private virtual networks concept is not new, SS7/C7 makes it possible for a Local exchange carrier (LEC) to offer the service. The customer receives PVNs, which are exactly like leased (private) lines except that the network does not allocate dedicated physical resources. Instead, SS7/C7 signalling (and a connected database) monitors the "private customer" line. The customer has all the features of a leased-line service as well as additional features, such as the ability to request extra services ad hoc and to tailor the service to choose the cheapest inter-exchange carrier (IC), depending on the time of day, day or week, or distance between the two parties.

2.6.15 Do-Not-Call Enforcement

In the United States, federal and state laws have already mandated do-not-call lists [108] in over half the states, and all states are expected to follow suit. These laws restrict organizations (typically telemarketers) from cold-calling individuals. To comply with these laws, SS7 can be used to query state and federal do-not-call lists (which are stored on a database) each time a telemarketer makes an outbound call. If the number is on a do-not-call list, the call is automatically blocked and an appropriate announcement is played to the marketer.

2.7 Common Channel Signalling:

In the Channel Associated Signalling (In band signalling), each speech channel carries its own signalling, both supervisory (line) signalling and address (Register) signalling.

With the introduction of Stored Program Controlled exchanges, making use of Central Processors, which enable call set up to be carried out quite fast, the need for an equally fast, reliable and efficient signalling system was felt, which has resulted in the parallel development of the common channel signalling system (out of band), also based on a central processor for signalling.

The switching control and the common channel signalling control can be handled by the same central processor, if adequate capacity is available or by use of distributed processing (i.e.) separate processors for control and signalling.

If signalling functions are handled by a separate processor, it is advantageous to separate the signalling from the speech channel and establish a separate data link between the signalling processors of different exchanges to handle the various signalling functions in setting up, holding and release of a call.

Common Channel Signalling, therefore, separates signalling from its associated speech path by placing the signalling for a group or groups of speech channels on a separate path dedicated to signalling only. Signalling on the telephone network is basically binary in nature (i.e.) it has only two possible states. Viz., On hook or Off hook, idle or busy etc. The signalling information on common channel signalling is transmitted in the Binary Format.

Each signalling link consists of a data link with terminal equipment at each end, and typically performing functions necessary for the orderly and correct transfer of signalling information, such as Error Control, Identifying the signalling information pertaining to different channels, etc. The CCITT No. 6 system evolved earlier makes use of 2.4 Kbps data link, optimized for use in Analogue networks can deal with 2048 trunks. The line and register signalling information pertaining to a particular trunk is transmitted in one or more of the 28 bit packets (20 bits for information and 8 bits for error checking). The first packet relating to a new signal for a particular trunk contains the indication of the number of packets and an identifier for the particular trunk.

With conventional signalling, the speech path and signalling path occupy the same channel. Since signalling is effected prior to setting up of the call, continuity of the speech path is automatically ensured. In common channel signalling since the speech path and signalling path are different, the speech path continuity will have to be checked up through other means before the call is put through.

The CCITT No. 7 signalling system is developed for use in a Digital environment using a signalling link of 64 Kbps. It is also suitable for use at lower signalling rates and in an Analogue environment.

The basic characteristics and advantages of the common channelling systems are as follows: -

i. Physical and functional separation of the signalling network from the speech network.
ii. Higher speed of signalling, compared to conventional system, implies decrease in post dialling delay and in holding time of switching equipment and circuit.
iii. Simultaneous signalling both ways.
iv. Capacity only limited by the speed of the data link and the processing capacity of the processors.
v. Simply stated, CCS evolved for ISDN & IN.
vi. For a large hierarchical NW, CAS is not okay especially with ISDN features.
vii. CAS or In-channel signalling (as in PCM) inadequate for ISDN.
viii. CCS needed to facilitate internal control, network intelligence essential for ISDN.
ix. SS7 is the culmination of transition from CAS to CCS.
x. Subscriber (user) to NW signalling has also evolved to facilitate mapping with SS7 for smooth signalling transaction end to end access control for BRI/ PRI.

2.8 SS7 Features

i. Optimized for use in Digital Telecom NWs in conjunction with stored program control exchanges, utilizing E0/E1/T1 connectivity.
ii. Supports information transfer required for call control, remote control, management, and maintenance.
iii. Provides reliable transfer of information in correct sequence without loss or duplication.
iv. Supports speeds below 64 kbps & hence can work on analogue channels.
v. Suitable for point to point terrestrial links/satellite links
vi. Suitable for Cellular Networks (GSM).

2.9 Three Essential Concepts

i. Signalling packet
ii. Overlaid Network
iii. Assignment of flexible function

2.9.2 Concept of 'Signalling Packets'

Common Channel for signalling carries messages in the form of packets known as 'Signalling packets'. Signalling packets have messages for call management (Set-up, maintenance, termination), network management (Link, route management).

2.9.3 Concept of 'Overlaid NW'

i. Network being controlled is 'Circuit switched'
ii. 'Packet Switched NW' overlaid on 'Circuit Switched NW' (Ref Fig 2.1)

SCP –signal control point

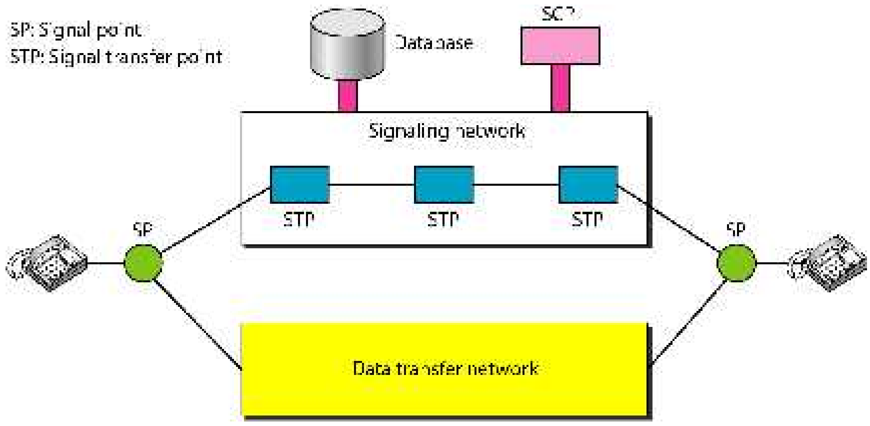


Fig. 2.1 Overlaid signaling network

Objective:

1. In conventional signalling, the speech path and signalling path occupy the same channel.

2. In common channelling systems, signalling network is physically and functionally separated from the speech network.

3. Short Message Service (SMS) forms part of the GSM specifications and allows two-way transmission of alphanumeric text between GSM subscribers.

4. Local number portability (LNP) provides the option for subscribers to retain their telephone number when changing their telephone service.

Subjective:

1. What are the special Services that are obtained from the Signalling System No. 7?

2. What are the basic characteristics and advantages of the common channelling systems?

3. What are the salient features of SS7?

CHAPTER-3

SS7 ARCHITECTURE

3.0 SS7 Architecture

A telecommunications network consists of a number of switches and application processors interconnected by transmission circuits. The SS7 network exists within the telecommunications network and controls it. SS7 achieves this control by creating and transferring call processing, network management, and maintenance to the network's various components.

3.1 Signaling points

An SS7 network has three distinct components: Service Switching Points, Signal Transfer Points, and Service Control Points. These components may be generically referred to as "nodes" or "signaling points" and are connected to each other via "data links". These nodes are graphically represented as

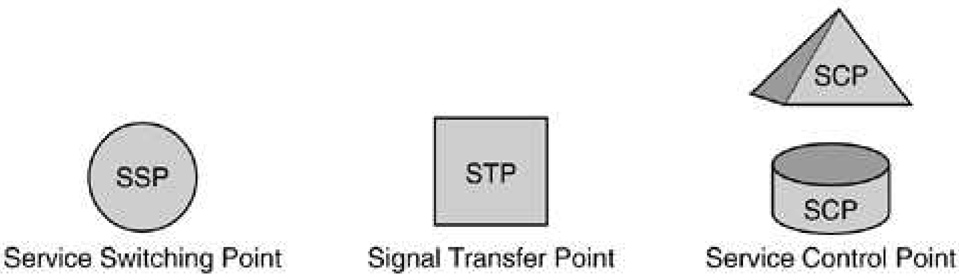


Fig 3.1 Signalling points

3.1.1 SP or SSP (Signal switching Point)

This is the end point of control messages. It handles SS7 control messages but it cannot process messages that are not addressed to it.

3.1.2 STP (Signal Transfer Point)

The incoming signalling information is processed and selected the desired route of the Signalling packets for next desired STP or SSP. They also performed specialized Routing function

3.1.3 SCP (Signal control point)

SCP are databases that provides information massages for advance call Processing capabilities. Signalling information and route information is stored here and ST can collect signalling information from SCP

3.1.4 SL (Signal Link)

The paths connect SPs & STPs

3.2 SS7 LINK TYPE

3.2.1 Access Links (A Links)

Access links (A links), shown in Fig. 3.2, provide access to the network. They connect "outer" SPs (SSPs or SCPs) to the STP backbone. A links connect SSPs and SCPs to their serving STP or STP mated pair.

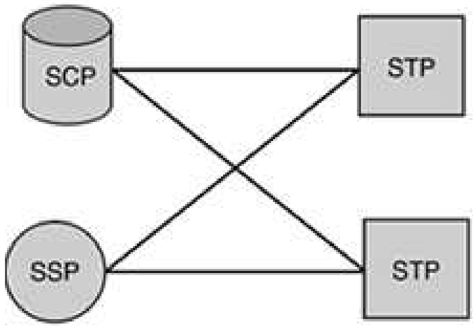


Fig. 3.2 Access Links

3.2.2 Cross Links (C Links)

Cross links (C links), shown in Fig. 3.3, are used to connect two STPs to form a mated pair—that is, a pair linked such that if one fails, the other takes the load of both. C links are used to carry MTP user traffic only when no other route is available to reach an intended destination. Under normal conditions, they are used only to carry network management messages.

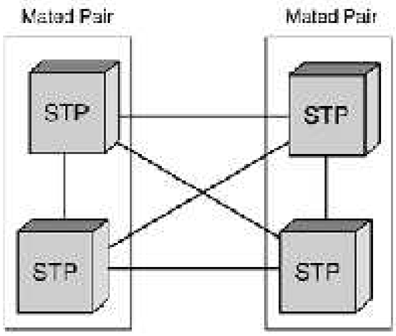


Fig. 3.3 Cross Links

3.2.3 Bridge Links (B Links)

Bridge links (B links) are used to connect mated pairs of STPs to each other across different regions within a network at the same hierarchical level. These links help form the backbone of the SS7 network. B links are normally deployed in link quad configuration between mated pairs for redundancy. Fig. 3.4, shows two sets of mated pairs of B links.

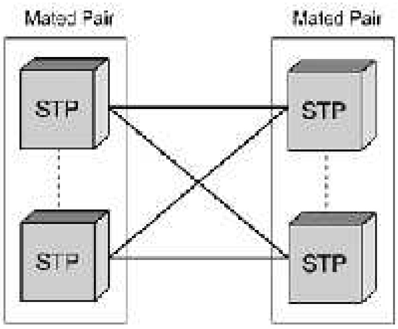


Fig. 3.4 Bridge links

3.2.4 Diagonal Links (D Links)

Diagonal links (D links), shown in Fig3.5, are the same as B links in that they connect mated STP pairs. The difference is that they connect mated STP pairs that belong to different hierarchical levels or to different networks altogether. For example, they may connect an interexchange carrier (IXC) STP pair to a local exchange carrier (LEC) STP pair or a cellular regional STP pair to a cellular metro STP pair.

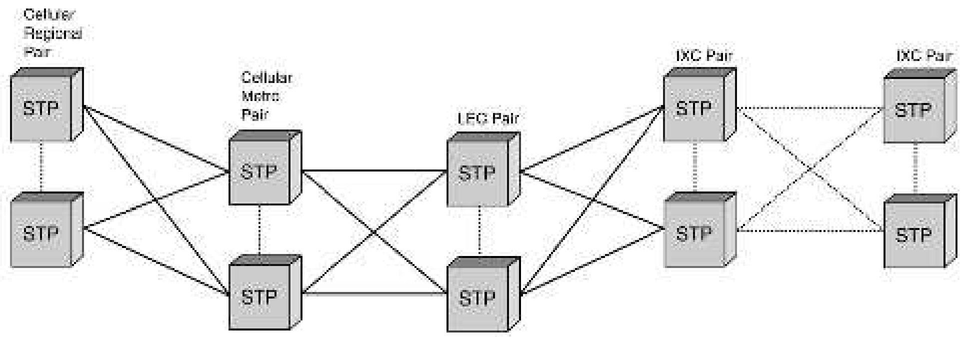


Fig. 3.5 Diagonal Links

3.2.5 Extended Links (E Links)

Extended links (E links), shown in Fig 3.6, connect SSPs and SCPs to an STP pair, as with A links, except that the pair they connect to is not the normal home pair. Instead, E links connect to a nonhome STP pair. They are also called alternate access (AA) links. E links are used to provide additional reliability or, in some cases, to offload signaling traffic from the home STP pair in high-traffic corridors. For example, an SSP serving national government agencies or emergency services might use E links to provide additional alternate routing because of the criticality of service.

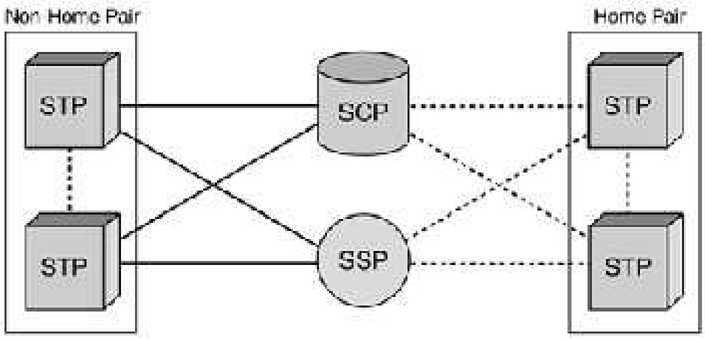


Fig 3.6 Extended Links

3.2.6 Fully-Associated Links (F Links)

Fully-associated links (F links), shown in Fig. 3.7, are used to connect network SSPs and/or SCPs directly to each other without using STPs. The most common application of this type of link is in metropolitan areas. F links can establish direct connectivity between all switches in the area for trunk signaling and Custom Local Area Signaling Service (CLASS), or to their corresponding SCPs.

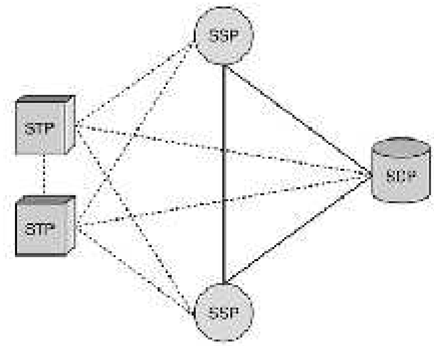


Fig. 3.7 Fully-Associated Links

Fig. 3.8 shows an SS7 network segment. In reality, there would be several factors more SSPs than STPs.

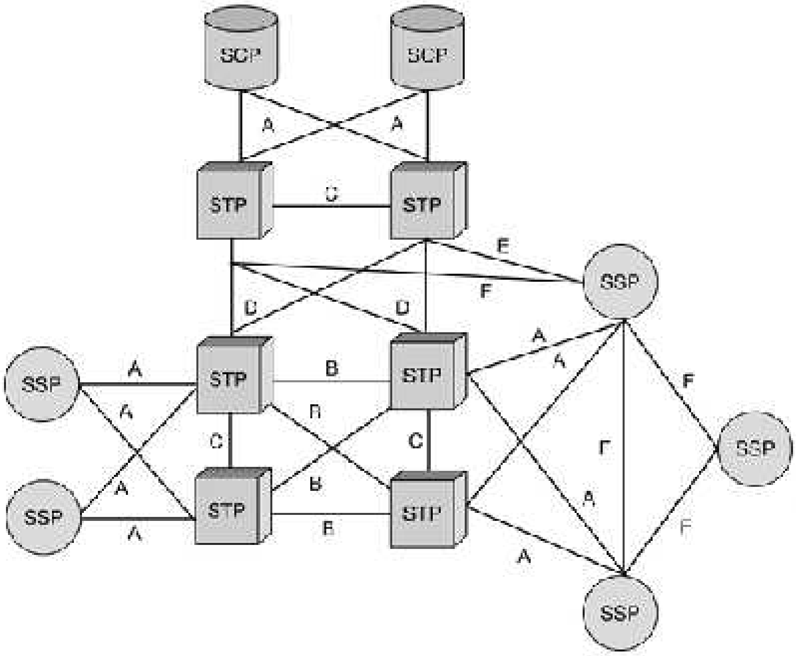


Fig. 3.8 SS7 network segment

3.3 Basic call setup

Switch A analyzes the dial digits and determines that it needs to send the call to switch B. Selects an idle trunk between A and B. Formulates an initial address message (IAM) that identifies the initiating switch A, the destination switch B, the trunk selected, the calling and called number.

STP (W) receives a message, inspects its routing table and determines that it is to be routed to switch B. It transmits the message on link BW. B serves the dialled message to called number.

B formulates an address complete message (ACH) that includes IAM reached its proper destination, identifies recipient switch A, sending switch B and the selected trunk to send ring back tone to A and send ringing current to called party.

Called party lifts telephone. Switch B formulates an answer message (ANM), indicating switch A, switch B and a free trunk to send a message from B to A

Released message (REL) is generated by switch A and received by switch B. B in turn disconnects the Link.

Release complete message (RLC) generated by switch B to send switch A and switch A idles the identified trunk that is earlier seized. See Fig. 3.9

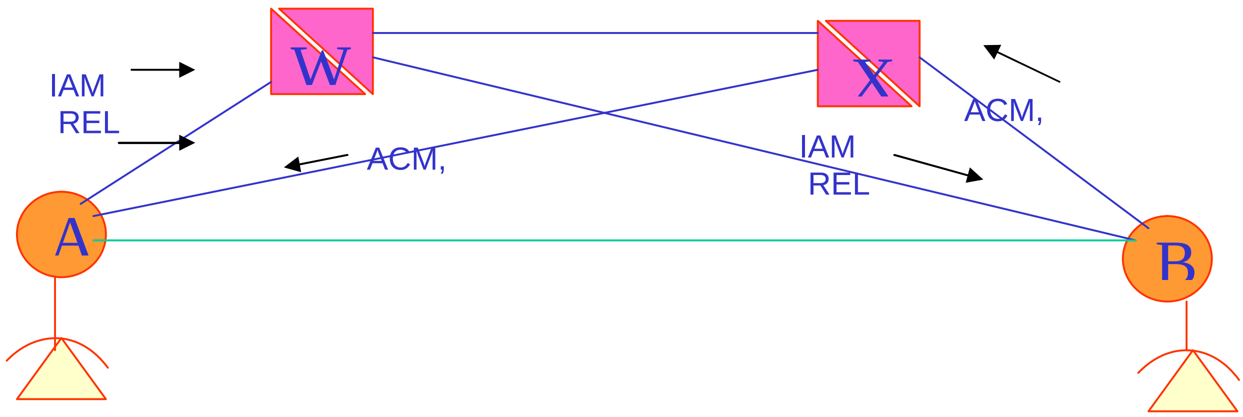


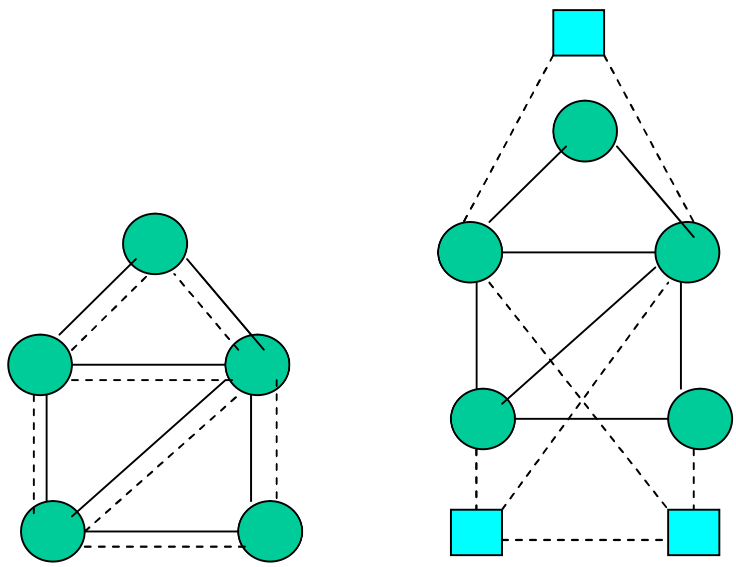
Fig. 3.9 Basic call setup

IAM – initial address message
ACM – address complete message
ANM- answer message
REL – release message
RLC- release complete message

---------- Subscriber line
---------Voice trunk
-----------Signalling link

3.4 Concept of ‘Flexible function assignment

SS7 defines the functions to be performed by the ‘overlaid packet switched NW’. It does not dictate any hardware implementation. Functions to be performed can be assigned in a flexible manner. The SS7 has the capability of functioning as associated and disassociated. In associated functioning the circuit switching nodes can do all SS7 functions as additional functions. In disassociated functioning Separate switching points that carry only ‘Signalling packets’ and do not involve in ‘circuit carrying’, can be used. See Fig. 3.10



Associated **Disassociated**

Fig. 3.10 Flexible function assignment

Objective:

1. SSP (Signal switching Point) is the end point of control messages
2. STP (Signal Transfer Point) perform specialized Routing functions.
3. SCP are databases that provides information massages for advance call Processing capabilities

Subjective:

1. What are the different signaling points in an SS7 network and explain?

CHAPTER-4

SS7 PROTOCOL SUITE

4.0 SS7 protocol suite

The number of possible protocol stack combinations is growing. It depends on whether SS7 is used for cellular-specific services or intelligent network services, whether transportation is over IP or is controlling broadband ATM networks instead of time-division multiplexing (TDM) networks, and so forth. This requires coining a new term—traditional SS7—to refer to a stack consisting of the protocols widely deployed from the 1980s to the present:

- Message Transfer Parts (MTP 1, 2, and 3)
- Signaling Connection Control Part (SCCP)
- Transaction Capabilities Application Part (TCAP)
- Telephony User Part (TUP)
- ISDN User Part (ISUP)

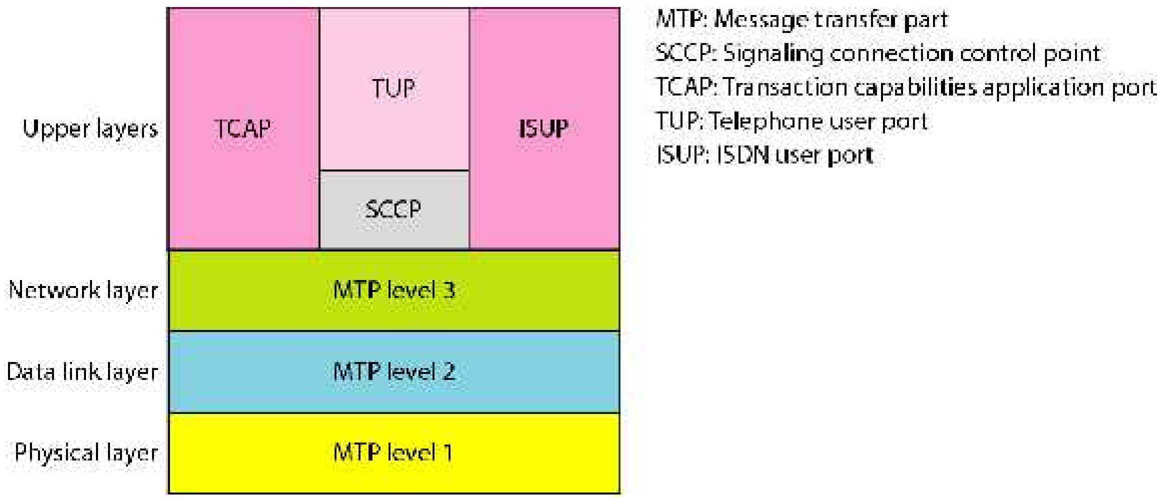


Fig 4.1 Layer model (Protocol Suite) of SS7

4.1 Massage transfer part (MTP)

We have so far established that signalling is used for setting up calls, and that there are standard sets of messages, which are sent back and forth to help facilitate this. The part responsible for taking these messages from one network element to another network element is known as the messages transfer part (MTP). The entire SS7 is built on the foundation of this MTP, which consists of three sub layers.

The lowest level, MTP layer 1 (physical connections), defines the physical and electrical characteristics. MTP layer 2(data link control) helps in error free transmission of the signalling messages between adjacent elements. MTP layer 3 (network layer) is responsible for taking the message from any element in a signalling network to any other element within the same network.

4.1.1 Functions of MTP-1 layer (physical layer)

This defines the physical and Electrical characteristic of the signalling links of SS7 network.

- It provides full-duplex data connection dedicated to SS7 traffic.
- The data rate Can be E0 (64 kbps) / E1 (2.048 Mbps) / T1 (1.544 Mbps).
- The speeds lower than 64 kbps is also supported.
- Analog channels can be used with modems.

4.1.2 Functions of layer-2 of ss7 (data link layer)

- It ensures that the two end points of signalling link can reliably exchange signalling messages.
- It incorporates such capabilities are error checking, flow control over sender to avoid buffer overflow and sequence checking to provide reliable data link.
- It contains the node-to-node address in header and tailor.
- All transmitted blocks of data are delivered with no losses or duplication.

4.2 SS7 Layer-2 (MTP Level-2) Data Blocks

Signalling information is passed over the signalling link in messages, which are called signalling units (SUs). Blocks of data handled here.

There are three types of SUs, each with its own format: the fill-in signal unit (FISU), the link status signal unit (LSSU), and the Message Signal Unit (MSU). An in-service signaling link carries a continuous SU stream in each direction.

FISUs and LSSUs are used only for MTP2 functions. MSUs also contain the same MTP2 fields, but they have two additional fields filled with information from MTP3 and Level 4 users that contain the real signaling content. This chapter describes the MTP2 fields and the functions they perform. It begins by presenting the three SU formats.

4.2.1 Fill-In Signal Units

FISUs are the most basic SU and carry only MTP2 information. They are sent when there are no LSSUs or MSUs to be sent, when the signaling link would otherwise be idle. Sending FISUs ensures 100 percent link occupancy by SUs at all times. A cyclic redundancy check (CRC) checksum is calculated for each FISU, allowing both signaling points at either end of the link to continuously check signaling link quality. This check allows faulty links to be identified quickly and taken out of service so that traffic can be shifted to alternative links, thereby helping meet the SS7/C7 network's high availability requirement. Because MTP2 is a point-to-point protocol, only the MTP2 level of adjacent signaling points exchanges FISUs.

The seven fields that comprise a FISU, shown in Fig. 4.1 are also common to LSSUs and MSUs. MTP2 adds the fields at the originating signalling point and processes and removes them at the destination signaling point (an adjacent node).

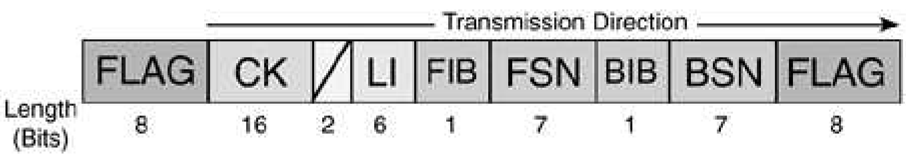


Fig. 4.1 Fields that comprise a FISU

4.2.2 Link Status Signal Units

LSSUs carry one or two octets of link status information between signaling points at either end of a link. The link status controls link alignment, indicates the link’s status, and indicates a signaling point’s status to the remote signaling point. The presence of LSSUs at any time other than during link alignment indicates a fault—such as a remote processor outage or an unacceptably high bit error rate affecting the ability to carry traffic.

The timers associated with a particular status indication govern the transmission interval. After the fault is cleared, the transmission of LSSUs ceases, and normal traffic flow can continue. As with FISUs, only MTP2 of adjacent signaling points exchanges LSSUs. LSSUs are identical to FISUs, except that they contain an additional field called the Status field (SF). Fig. 4.2 shows the eight fields that comprise an LSSU.

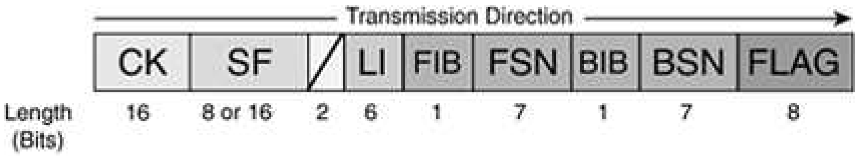


Fig. 4.2 fields that comprise an LSSU

Currently only a single-octet SF is used, even though the specifications allow for a two-octet SF. From the single octet, only the first 3 bits are defined. These bits provide the status indications shown in Table 4.1.

Table 4.1. Values in the Status Field

C	B	A	Status Indication	Acronym	Meaning
0	0	0	O: Out of Alignment	SIO	Link not aligned; attempting alignment
0	0	1	N: Normal Alignment	SIN	Link is aligned
0	1	0	E: Emergency Alignment	SIE	Link is aligned
0	1	1	OS: Out of Service	SIOS	Link out of service; alignment failure
1	0	0	PO: Processor Outage	SIPO	MTP2 cannot reach MTP3
1	0	1	B: Busy	SIB	MTP2 congestion

4.2.3 Message Signal Units

As shown in Fig. 4.3, MSUs contain the common fields of the FISU and two additional fields: the Signalling Information Field (SIF) and the Service Information Octet (SIO). MSUs carry the signalling information (or messages) between both MTP3 and Level 4 users. The messages include all call control, database query, and response messages. In addition, MSUs carry MTP3 network management messages. All messages are placed in the SIF of the MSU.

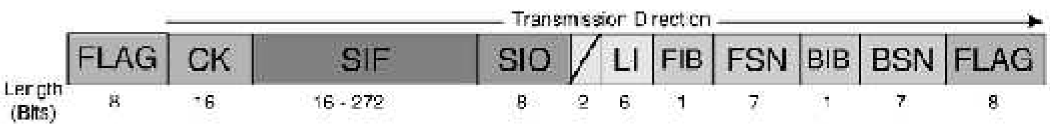


Fig. 4.3 Message Signal Units

Table 4.2. Field Descriptions

Field	Length in Bits	Description
Flag	8	A pattern of 011111110 to indicate the start and end of an SU.
BSN	7	Backward sequence number. Identifies the last correctly received SU.
BIB	1	Backward indicator bit. Toggled to indicate an error with the received SU.
FSN	7	Forward sequence number. Identifies each transmitted SU.
FIB	1	Forward indicator bit. Toggled to indicate the retransmission of an SU that was received in error by the remote SP.
LI	6	Length indicator. Indicates how many octets reside between itself and the CRC field. The LI field also implies the type of signal unit. LI = 0 for FISUs, LI = 1 or 2 for LSSUs, and LI >2 for MSUs.
SF	8 to 16	Status field. Provides status messages in the LSSU only.
CK	16	Check bits. Uses CRC-16 to detect transmission errors.
SIO	8	Service Information Octet. Specifies which MTP3 user has placed a message in the SIF.
SIF	16 to 2176	Signaling Information Field. Contains the "real" signaling content. The SIF is also related to call control, network management, or databases query/response.

4.2.4 Differences between MSU & LSSU

- MSU has 2 user-data fields: SIO & SIF
- LSSU has only one user-data field: SF
- MSU carries user data from higher layers
- LSSU indicates sender's view about the status of link

4.3 Addressing in the SS7 network

Every network must have an addressing scheme, and the SS7 network is no different. Network addresses are required so that a node can exchange signaling nodes to which it does not have a physical signaling link. In SS7, addresses are assigned using a three level hierarchy. Individual signaling points are identified as belonging to a cluster of signaling points. Within that cluster, each signalling point is assigned a member number. Similarly, a cluster is defined as being part of a network. Any node in the American SS7 network can be addressed by a three level number defined by its network, cluster, and member numbers. Each of these numbers is an 8-bit number and can assume values from 0 to 255. This three-level address is known as the point code of the signalling point. A point code uniquely identifies a signaling point within the American SS7 network and is used whenever it is necessary to address that signaling point.

A neutral party assigns network numbers on a nationwide basis. Regional Bell operating companies, major independent telephone companies and inter-exchange carriers already have network numbers assigned. Because network numbers are relatively scare resource, company's networks are expected to meet certain size requirements in order to be assigned network number. Smaller networks can be assigned one or more cluster numbers within network numbers 1, 2, 3 and 4. The smallest networks are assigned point codes within network number 5. The cluster to which they are assigned is determined by the state in which they are located. The network number 0 is not available for assignment and network number 255 is reserve for future use.

4.4 Functions of MTP layer-3 of ss7 (network layer)

The level –3 portion of message transfer part extends the functionality provided by MTP level – 2 to provide network layer functionality. It ensures that messages can be delivered between signalling points across the SS7 network regardless of whether they are directly connected. It includes such capabilities as node addressing, routing, and alternate routing and congestion control.

4.4.1 The function of layer–3 is divided into two categories.

i. Signalling Message Handling functions
   - Discrimination
   - Routing
   - Distribution

ii. Signalling NW (Network) management functions
   - Traffic management
   - Link management
   - Route management

4.4.2 SIG. MESSAGE HANDLING FUNCTIONS

i. Discrimination
   - This function is provided at STP only.
   - It is determined if a message is at destination or is to be relayed to another node.
   - Destination code in Routing label of MSU (Message signal unit) is examined to determine service option.

ii. Routing
   - Determines the Sig. Link to be used to forward message.
   - (Message received from discrimination function or local level-4 entity).
   - Routing decision is based on value of SLS (Signalling link selection) field.
   - 4 bits in SLS – So 16 possible routes are possible (Internal Virtual Circuits).
   - In general, all messages associated with a single call go on same link unless link develops problem.

iii. Distribution
   - Determines the user part to which message is to be delivered.
   - Service indicator portion of SIO helps in deciding service option.

4.4.3 Signalling NW management functions

- Performance of NW's Signalling sub-system affects all subscribers in the NW.
- NWs often support international traffic. Degradation in one nation's signalling sub-system will have repercussions beyond national boundaries.
- Recovery & restoration actions involve multiple networks (across many nations). Hence, failure & congestion recovery are to be incorporated.

Objective of Sig. NW Management Functions –

- Performance criterion: - Overcome link degradation (congestion or failure)
- Availability criterion: - 99.998%
- Permitted un-availability: - 0.002% or 10 min. per year

i. Signalling Traffic Management is used to –

- Divert signalling traffic without loss or duplication, from un-available signalling links or routes to one or more alternative signalling links or routes.
- Reduce traffic in case of congestion.

The Sig. Traffic Management Functions are performed to exchange the level three messages between the SPs. These messages are carried in SIF (Signal information field) of MSU.

Table 4.3 Signalling Traffic Management Functions

Function	Description
Changeover	Diverts traffic to one or more alternative links in the event of a link un-availability
Change-back	Re-establish traffic on a signalling link that becomes un-available
Forced re-routing	Divert traffic to an alternate route when a route becomes unavailable
Controlled re- routing	Divert traffic to a route that has been made available
Signalling point restart	When a signalling point becomes available and when signaling traffic is diverted to or through this point, update the network routing status and control
Management inhibiting	Link made unavailable for maintenance/testing
Signalling traffic flow control	Limit signalling traffic at it's source when the signalling network is not capable of transferring all signalling traffic offered by the user because of network failure or congestion

ii. Signalling link management function

Signalling Link Management is used to –

- Restore failed signalling links.
- Activate new signalling links.
- De-activate aligned signalling links.

Table 4.4 Signalling Link Management Functions

Function	Description
Sig. Link activation, restoration, de-activation	Activate new links, restore failed links and de-activate links
Link set activation	Activate a link set not having any links in service
Automatic allocation of sig-Terminals and sig. data links	Allocate terminals to links

iii. Signalling Route Management

Signalling Route Management is used to

- Distribute information about signalling status to block or un-block signlling routes.
- Control source SPs connected to links, which are congested (This is required in addition to flow control of layer-2).

Table 4.5 Signalling Route Management Functions

Procedure	Description
Transfer- controlled procedure	Performed at an STP in case of link congestion. Message sources are told to stop sending messages having a congestion priority less than the congestion level of the link
Transfer- prohibited procedure	Performed at an STP to inform adjacent SPs that they must no longer route to a particular destination via this STP
Transfer-allowed procedure	Informs adjacent SPs that routing to a given destination is normal
Transfer- restricted procedure	If possible, adjacent SPs should no longer route to a particular destination via this STP
Signalling-route- set test procedure	Used by S Ps receiving transfer-prohibited and transfer- restricted messages to recover the signaling route information that may not have been received due to some failure
Signalling-route-set congestion test procedure	Used to update the congestion status associated with a route towards a particular destination

Table 4.6 Signalling Route Status

Status	Description
Available	Signalling traffic towards a particular destination can be transferred via this STP
Restricted	Signalling traffic towards a particular destination is being transferred with some difficulty via this STP
Unavailable	Signalling traffic towards a particular destination cannot be transferred via this STP

Table 4.7 Signalling Route Set Status

Status	Description
Congested	Indicates that the buffer occupancy rate of link exceeds a given threshold
Un-congested	The buffer occupancy rate of a link is within pre-determined limits

Objective:

1. MTP layer 1 defines the physical and electrical characteristics.
2. MTP layer 2 helps in error free transmission of the signalling messages between adjacent elements.
3. MTP layer 3 is responsible for taking the message from any element in a signalling network to any other element within the same network.

Subjective:

1. What are different layers of SS7Protocol Suite? Explain briefly?

CHAPTER-5

FUNCTIONS OF HIGHER LAYERS OF SS7

5.0 Signalling Connection Control Part (SCCP)

The signalling connection control part (SCCP) provides two major functions that are lacking in the MTP. The first of these is the capability to address applications within a signalling point. The MTP can only receive and deliver massages from a node as a whole; it does not deal with software applications within a node.

While MTP network management messages and basic call setup messages are addressed to a node as a whole, other messages are used by separate applications (referred to as subsystems) within a node. Examples of subsystems 800(Toll free number), call progressing, calling card processing, advance intelligent network and custom local area signalling services (Ex – repeat dialling and call return). SCCP allows these systems to be addressed explicitly.

GTT (Global title translation): The second function provided by the SCCP is the availability to perform incremental routing using a capability called Global title translation (GTT). GTT frees original signalling points from the burden of having to know every potential destination to which they might have to route a message. A switch can originate a query, for example, and address it to a STP along with request for GTT. The receiving STP can then examine a portion of the message, make a determination as to where the message should be routed and then route it.

STP must maintain a database that enables them to determine where a query should be routed. GTT effectively centralizes the problem and places it in a node (STP) that has been designed to perform this function. In performing GTT, a STP does not need to know the exact final destination of a message. It can instead perform intermediate GTT in which it uses its tables to find another STP further along the route to the destination. That STP, in turn can perform final GTT, routing the message to its actual destination. Intermediate GTT minimizes the need for STPs to maintain extensive information about modes that are far remote from them. GTT also is used at the STP to share load among meted SCPs in both normal and failure scenarios. In these instances, where messages arrive at an STP for final GTT and routing to a database, the STP can select from among available redundant SCPS. It can select an SCP on either a priority basis (referred to as primary backup) or as to equalize the load across all available SCPs.

5.1 Transaction capabilities Application Parts (TCAP)

TCAP defines the messages and protocol used to communicate between applications (deployed as subsystems) in nodes. It is used for database services such as calling card, 800, AIN as well as switch-to-switch services including repeat dialling and call return. Because TCAP message must be delivered to individual application within the nodes the address, they use the SCCP for transport.

5.2 Operation Maintenance and Administration Part (OMAP)

OAMP defines messages and protocol designed to assist administrations of the SS7 network. To date, the most fully developed and deployed of these capabilities are procedures for validating network routing tables and for diagnosing link troubles. OMAP includes messages that use both the MTP and SCCP for routing.

5.3 ISUP (ISDN user part)

The ISUP user part defines the messages and protocol used in the establishment and tear down of voice and data calls over the public switched network and to manage the trunk network on which they rely. Despite its name, ISUP is used for both ISDN and non-ISDN calls. In the North American version of SS7 ISUP messages rely exclusively on MTP to transport messages between connected nodes.

5.3.1 ISUP Requirements

i. It must rely on network service part of SS7
ii. Flexible to accommodate future enhancements of ISDN
iii. It must inter-work with user-network Q.931 call control protocol

Important observations on ISUP

- Q.931 defines call-control protocol over CCS open to use by ISDN subscriber i.e. ISDN subscriber uses Q.931 procedures to set-up calls to other subscriber, with associated user facilities supported in Q.931.
- ISUP refers to signalling facilities employed by the NW on behalf of ISDN user.

The function can be summarized as: -

- ISDN communicates with ISDN subscriber via Q.931 for purpose of call control.
- ISDN uses ISUP internal to the network to implement subscriber call control requests.
- ISUP transactions within NW are as per SS7 protocol suite
- The term ‘user’ in ISUP does not refer to ISDN user (subscriber). It highlights the fact that ISUP is a user of lower layers of SS7.

5.3.2 ISUP MESSAGES

There are eight categories: -

i. Forward set-up messages
ii. General set-up messages
iii. Back-ward set-up messages
iv. Call supervision messages
v. Circuit supervision messages
vi. Circuit group supervision messages
vii. In call modification messages
viii. End-to-end messages

i. Forward set-up messages
   - Used to set-up a circuit, by identifying exchange end points.
   - Desired characteristics of the call can also be specified.

These messages propagate in forward direction – from call originating exchange to destination exchange.

Types of forward set-up messages

- Initial address message
  Sent in forward direction to initialize seizure of an outgoing circuit and to transmit address and related information.
- Subsequent address message
  May be sent following initial address message to convey additional calling party address information.

ii. General set-up messages

- Used during call establishment phase.
- They provide means of transferring any additional information required during call set-up.
- We can also check using these messages that- A circuit that straddles more than one ISDN, maintains desired characteristics across all network

Types of General set-up messages

- Information request: Requests additional call related information.
- Information: Conveys additional call related information
- Continuity: Sent in forward direction to indicate continuity of the preceding speech circuit to the following international exchange.

iii. Back-ward set-up messages

Support call set-up process and initiate accounting and call charging procedures.
Types of back-ward set-up messages

- Address complete: Sent in backward direction to indicate that all the address information required for routing the call to the called party is received.
- Connect: Sent in backward direction to indicate that all the address information required for routing the call to the called party is received and call has been answered.
- Call progress: Indicates that an event has occurred during call set-up that should be relayed to the calling party.

iv. Call supervision messagesThese are additional messages that might be needed in the process of call establishment. These include indications whether call was answered or not and the capability to support manual intervention between ISDNs that cross national boundaries.

Types of call supervision messages

- Answer: Sent in backward direction to indicate call was answered.
- Forward transfer: Sent in forward direction on semi-automatic calls when outgoing international exchange operator wants help of an operator at the incoming international exchange.
- Release: Indicates that the circuit identified in the message is released.

v. Circuit supervision messages

Relate to already established circuit
Three functions are supported:

- Circuit may be released
- Circuit may be suspended and later resumed
- Circuit may be established

Types of Circuit supervision messages

- Delayed release
- Release complete
- Continuity check request
- Reset circuit
- Loop-back acknowledgement
- Blocking
- Un-blocking
- Unequipped circuit identification code
- Blocking ack.
- Unblocking ack.
- Overload
- Suspend
- Resume
- Confusion

vi. Circuit group supervision messages

Perform similar functions as circuit supervision messages for a group of Circuits.

Types of circuit group supervision messages

- Circuit group blocking
- Circuit group un-blocking
- Blocking ack.
- Unblocking ack.
- Circuit group reset
- Reset ack.
- Overload
- Circuit group query
- Circuit group query response

vii. In-call modification messages

Used to alter characteristics or associated NW facilities of an active call.

Types of in call modification messages

- Call modification request
- Call modification completed
- Call modification reject
- Facility request
- Facility accepted
- Facility rejected

viii. End-to-end messages

Include pass along and end-to-end user Messages.

- Pass-along: - To transfer information between SPs.
- End-to-end: - To send user-to-user signalling information independent of call-control messages.

5.4 ISUP message fields

ISUP message has the following fields: -

- Routing label
  - Part of MTP header
  - Contains source code, destination code, SLS.
  - Circuit identification code
  - Specifies circuit to which message relates.
- Message type
  - Identifies message type.
- Mandatory fixed par
  - Position, Length, order of parameters depends on type.
  - Mandatory variable part
- Pointers locate parameters.
  - Optional part
- Pointers also locate optional parameters

5.5 SS7 applications in GSM

The additional protocol layers with the PSTN protocol layer are –

i. Base station subsystem application part (BSSAP)
ii. Mobile application part (MAP)
iii. Traction capabilities application part

BSSAP – This layer is used when an MSC communicate with BSC and mobile station.

5.6 SS7 layer in GSM elements

Protocol stack in the MSC

Since the MTP is the foundation on which SS7 is built, this will be required in every element that is capable of precessing SS7. The MSC is the element in GSM networks that is responsible for call control; TUP/ISP sits on top of the MTP. The MSC/ VLR is also responsible for location updates and communication with the BSC and the HLR. For this reason it also needs to have BSSAP and MAP that sit on top of the SCCP. The MSC also has the TCAP to provide services for the MAP. It can thus be seen that the MSC/VLR has all the SS7 protocol implemented in it.

Functions of higher layers of SS7

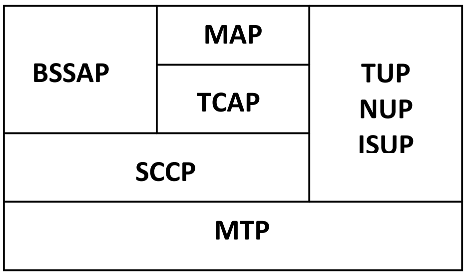


Fig 5.1 Protocol stack in the MSC

i. BSSAP

This layer is used when an MSC communicates with the BSC and the mobile station. Since the mobile station and the MSC have to communicate via the BSC, there must be a virtual connection; therefore the service of the SCCP is also needed. The authentication verification procedure and assigning a new TMSI all take place with the standard sets of messages of the BSSAP. Communication between the MSC and the BSC also uses the BSSAP protocol layer.

ii. MAP

While a mobile terminated call is being handled, the MSRN (Mobile subscriber roaming number) has to be requested from the HLR without routing the call to the HLR. Therefore, for these cases another protocol layer is added to the SS7 and is called the Mobile Application Part. MAP is used for signalling communication between NSS element.

iii. Transaction capabilities application part (TCAP)

In MAP signalling, one MSC sends a massage to an HLR, and that message requests (or invokes) a certain result. The HLR sends the result back, which may be the final result or some other result or some other messages might also follow (or it might not be last result). These invocations and results that are sent back and forth between multiple elements using MAP need some sort of secretary to manage the transactions. This secretary is called the transaction capabilities application part. This completes the SS7 protocol stack in the GSM network and their functions.

iv. Massage transfer part (MTP)

We have so far established that signalling is used for setting up calls, and that there are standard sets of messages, which are sent back and forth to help facilitate this. The part responsible for taking these messages from one network element to another network element is known as the messages transfer part (MTP). The entire SS7 is built on the foundation of this MTP, which consists of three sub layers as shown in the figure bellow

The lowest level, MTP layer 1 (physical connections), defines the physical and electrical characteristics. MTP layer 2(data link control) helps in error free transmission of the signalling messages between adjacent elements. MTP layer 3 (network layer) is responsible for taking the message from any element in a signalling network to any other element.

Objective:

1. The protocol used for MSC to communicate with the BSC and the mobile station is BSSAP

2. MAP is used for signalling communication between NSS element.

3. The secretary to manage the transactions of MAP is transaction capabilities application part (TCAP).

4. OAMP defines messages and protocol designed to assist administrations of the SS7 network.

5. The ISUP defines the messages and protocol used in the establishment and tear down of voice and data calls over the public switched network

Subjective:

1. What are the higher level protocols in SS7 suit and explain their functions?

2. What are the additional protocols available in SS7 suit specific to GSM networks? Explain briefly.