LESSON 31: SIGNALLING TECHNIQUES AND SIGNALLING SYSTEMS

Objective

To provide a detailed understanding of the concepts signaling systems.

Introduction

Signaling

A major component of any telephone system is signaling, in which electric pulses or audible tones are used for alerting (requesting service), addressing (e.g., dialing the called party's number at the subscriber set), supervision (monitoring idle lines), and information (providing dial tones, busy signals, and recordings).

Signaling systems must obviously be compatible with switching systems in a network. They must be able to transmit all the signals required to operate the switches. They must also be compatible with transmission systems in the network in order to reach the exchanges that they control.

In general, signaling may occur either within the subscriber loop (that is, within the circuit between the individual telephone instrument and the local office) or in circuits between offices. Interoffice signaling has undergone the more notable evolution, changing over from simple "in-band" methods to fully digitized "out-of-band" methods.

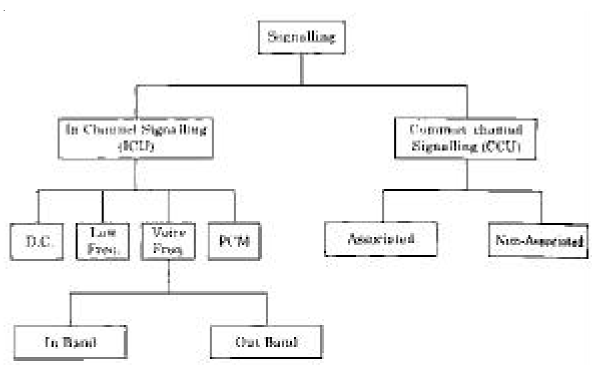


Fig.31.1 Classification of various techniques

In-band signaling

In the earliest days of the telephone network, signaling was provided by means of direct current (DC) between the telephone instrument and the operator. As long-distance circuits and automatic switching systems were placed into service, the use of DC became obsolete, since long-distance circuits could not pass the DC signals. Hence, alternating current (AC) began to be used over interoffice circuits. Until the mid-1970s, interoffice circuits employed what has become known as in-band signaling. In in-band signaling, the same circuits that were used to connect two telephone instruments and serve as the voice path were also used to transmit the AC signals that set up the switches employed in the circuit. Single-frequency tones were used in the switching network to signal availability of a trunk. Once a trunk line became available, multiple-frequency tones were used to pass the address information between switches. Multiple-frequency signaling employed pairs of six tones, similar to the signaling used in Touch-Tone dialing.

Out-of-band signaling

Despite the simplicity of the in-band method, this type of signaling presented a number of problems. First, because the in-band signals by necessity fell within the bandwidth of speech signals, speech signals could at times interfere with the in-band signals. Second, in-band signaling did not always make efficient use of the available telephone circuits. For example, if a called party's telephone instrument was in use, the called party's central office would generate a busy signal that was carried by the already established voice path through the PSTN to the calling party's handset. Hence, a full voice-circuit path through the network was tied up merely to convey a busy signal.

In order to overcome these issues and to speed the call set-up process in long-distance calls, another form of interoffice signaling, known as common channel signaling (CCS), was developed.

Common Channel Signalling

In a network of SPC (stored program control) exchanges, a connection that is made through two exchanges requires call processing by the central processor in each exchange. If a channel associated signaling is used for calls from exchange A to exchange B, it is necessary for the central processor of exchange A to send its outgoing forward signals to the individual speech circuit for transmission to exchange B. At this exchange, the signal must be detected on the speech circuit and passed to a central processor. Similarly, backward signals from processor B must be sent over the speech circuit, detected at exchange A and extended o its processor. This is an inefficient arrangement for signaling between the two processor.

If a high-speed data link is employed between the processors, it can provide a channel for all the signals between exchanges A and B. This is known as **common-channel signaling (CCS).** It gives the following advantages:

1. Information can be exchanges between the processors much more rapidly then when channel-associated signaling is used.
2. As a result, a much wider repertoire of signals can be used and this enables more services to be provided to customers.
3. Signals can be added or changes by the software modification to provide new services.
4. There is no longer any need for line signaling equipment on every junction, which results in a considerable cost saving.

5. Since there is no line signaling, the junctions can be used for calls from B to A in addition to calls from A to B. Both way working requires fewer circuits to carry the traffic then if separate groups of junction are provided from A to B and from B to A.
6. Signals relating to a call can be sent while the call is in progress. This enables customers to alter connections after they have been setup. For e.g., a customer can transfer a call elsewhere, or request a third party to be connected into an existing connection.
7. Signals can be exchanged between processors for functions other than call processing, for example for maintenance or network-management purposes.

The error rate for CCS must be very low and the reliability required is much greater than for channel associated signaling. Failure in data link would prevent any calls from being made between exchanges A and B, whereas failure of a line equipment, or even of an inter-register signaling system, would result only in the loss of a small fraction of traffic. The CCS is used; the successful exchange of signals over a circuit proves that is working. CCS does not inherently provide this checking facility, so a separate means (e.g. automatic routing testing) must be provided to ensure the integrity of the speech circuits.

CCS systems use message-based signaling. Successive messages exchanged between the processors usually relate to different calls. Each message must therefore contain a label, called the circuit identity code, that indicates that to which speech circuit, and thus to which call, it belongs. Since messages pass directly between central processors, no connection is required to an incoming junction before an address signal is received. The address signal can therefore be the first message sent and there is no need for a seize signal. In a multi-link connection, signaling takes place from one transit exchange to the next without involving the originating exchange. Thus, link-by-link signaling is inherent with CCS.

In CCS systems messages from the processor queue for transmission over the signaling link. The number of speech circuits that can be handled by a CCS system is therefore determined by the acceptable delay. A signaling link operating at 64kbps normally provides signaling for upto 1000 to 1500 speech circuits. However, more may be handled (with increased delays) when the load of a link that has failed is added to the existing to the existing load on a back-up link.

Common channel signaling (CCS) may be implemented in two ways:

(1) Channel associated mode

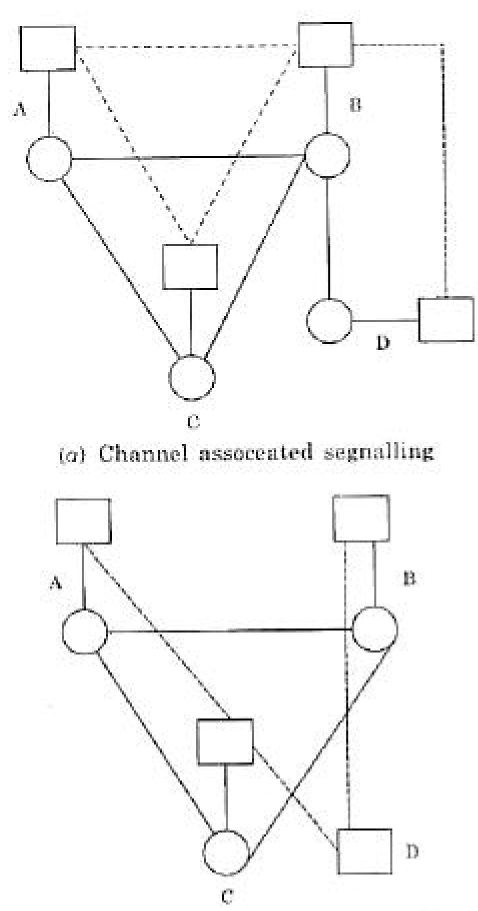
(2) Channel non-associated mode.

In channel-associated mode of CCS, the common signalling channel closely tracks the trunk groups along the entire length of a connection.

In channel non-associated mode of CCS, there is no close or simple assignment of control channels to trunk groups. Both modes of CCS are illustrated in Fig.31.2

Channel Associated Mode of operation of CCS:

In the channel associated mode of operation of CCS shown in Fig.31.2 (a), the signalling paths for the speech paths for the speech paths, A-B-A-C-B and B-D are A-B-A-C-B and B-D respectively. The signalling in CCS associated mode is still done on a separate signalling channel, only does the speech path. Network topologies of the signalling network and the speech network are the same.



(b) Channel Non-associated segnalling

Fig.31.2. Illustration of operation of common channel signalling (CCS).

It has the following advantages:

(1) Its implementation is economic

(2) Assignment of trunk groups to the signalling channel is simple.

A Common Channel Signalling (CCS) consists of two types of nodes:

(1) Signalling Transfer Points (STPs)

(2) Signalling Points (SPs)

The STPs is usually have a connection with the switching centers although this is not essential. Since signalling originates from the control subsystems of the switching centers. These control subsystems are referred to as SPs from the CCS signalling viewpoint.

Signalling point (SP) is capable of handling control messages directly addressed to it. But it is incapable of routing messages.

The STP is capable of routing messages and could also perform the functions of a signalling point (SP)

The use of CCS for inter-exchange signaling has been followed by its application to customers line in Integrated Services Digital Networks (ISDN).

The first version of CCS was developed between 1964 and 1968 by the **International Telegraph and Telephone Consultative Committee (CCITT),** a United Nations body that establishes worldwide telecommunications standards. The first system was standardized internationally as CCITT-6 signaling; within North America, CCITT-6 was modified by AT&T and became known as common channel interoffice signaling, CCIS. CCIS was first installed in the Bell System in 1976. In CCIS an "out-of-band" circuit (that is, a separate circuit from that used to establish the voice connection) was dedicated to serve as a data link, carrying address information and certain other information signals between the processors employed in telephone switches.

Although CCITT-6 was standardized by an international body, it was never universally deployed. Recognizing this shortcoming as well as the still-growing amount of international traffic within the worldwide telephone network, the CCITT, between 1980 and 1991, developed a successor version known as CCITT-7. Within North America, CCITT-7 has been implemented as Signaling System 7, or SS7. Additional features are provided in SS7 to support the integrated services digital network (ISDN) and to form the foundation of a future intelligent network.

The transmission bearers used for a CCS network are channels in the transmission bearer network. The first generation of CCS systems (CCITT no.6) used modems to transmit at 2.4kbps or 4.8kbps over a analog telephone channels. A 4kbps channel could also be provided over a 1.5 Mbps PCM system. Current CCS systems (CCITT no.7) use a 64kbps channel provided by time slot 16 in a 2 Mbps PCM system or time-slot 24 in a 1.5Mbps system.

CCITT signaling system no.6

The first CCS system signaling to be standardized internationally was a CCITT no.6 system. This was designed for the use in analog networks and used bit rates of 2.4kbps and 4.8 kbps. It used fixed-size signal units of 28 bits (20 information bits + 8 parity-check bits). A later version for use in digital networks added four padding bits to be compatible with 8-bit PCM time-slots. However, this pioneer system has now been superseded by CCITT signaling system no.7.

CCITT signaling system no.7

The concept of CCS has been further expanded in CCITT Signalling System No.7. *i.e.* SS7. It was first defined in 1980 and revised in 1984 and 1988. SS7 has been designed to be an open ended CCS standard that can be used over a variety of digital circuit switched networks. While the network being controlled is circuit switched, the control signalling itself uses a packet switched operation, but the actual implementation can be in circuit switched nodes as additional functions. Here the case when channel associated signalling mode is chosen for implementation.

Specific attention has been given to the requirements of ISDN while design SS7. The internal control and network intelligence essential to an ISDN are provided by SS7. Although, SS7 is suitable for operation over analog channels and at speeds less than 64kb/s. It is primarily optimised to work with digital SPC exchanges.

These SPC exchanges use 64 kb/s digital channels. SS7 is suitable for operation over both terrestrial and satellite links.

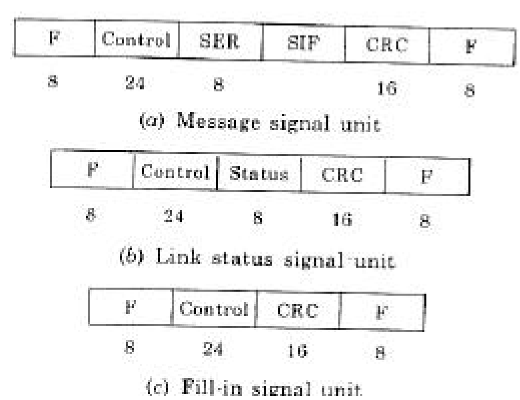
The protocol architecture of SS7 has four levels as shown in Fig. 31.3. The lower three levels referred to as the message transfer part (MTP) provide a reliable service for routing messages through the SS7 network. The lowest layer, signalling data link, is concerned with the physical and electrical characteristics of the signalling links between STPs and SP. All signalling data links in SS7 are full duplex links dedicated to SS7 traffic.

The main purpose of the second layer, viz. the signalling link, is to turn a potentially unreliable physical link into a reliable data link. The signalling link layer must ensure that

1. There are no losses or duplication of control message,
2. Messages are delivered in the same order in which they originate, and
3. There is a match between the receiver capacity and the transmission rates. (This function implies that the receiver is capable of exercising flow control over the sender.)

There are three types of signalling units (SUs) defined in SS7:

1. Message signal unit (MSU)
2. Link status signal unit (LSSU)
3. Fill-in signal unit (FISU)



(a) Message signal unit

(b) Link status signal unit

(c) Fill-in signal unit

Fig.31.3 Architecture of SS7.

The formats of these SUs are shown in Fig 31.4. All the SUs begin and end with a flag field, which has the unique bit pattern 01111110. The flags act as delimiters for the SUs. A common flag may be used as the closing flag for one SU and the opening flag for the next if the SUs are transmitted in continuum.

It is possible that the unique flag bit pattern appears inside the SU, thus destroying synchronisation. To avoid this problem, a technique known as bit stuffing is used. In brief, in this technique, the transmitter inserts an extra 0 whenever it comes across five consecutive 1 's in the data part of the SU. The receiver on detecting five 1 's deletes the zero following it. Thus only the flag pattern can contain six 1's.

All the SUs in SS7 contain a 16-bit error-checking field.

The control field consists of five sub fields as shown in Fig. 31.4 (d). The backward sequence number (BSN) and the backward indicator (BI) bit together permit piggybacked acknowledgement of the SUs received from the other side. A negative acknowledgement is indicated by inverting the BI bit which remains unchanged for all subsequent positive acknowledgements. The forward sequence number (FSN) identifies the SU uniquely using modulo 128 count. A retransmission is indicated by inverting the forward indicator (FI) bit which remains unchanged until another retransmission occurs. The length field specifies the length of the following information fields in octets. This provides a crosscheck on closing flag. It also serves as a SU type indicator. A value of 0 indicates a FISU, a value of 1 implies a LSSU and a value of 3 to 63 specifies a MSU.

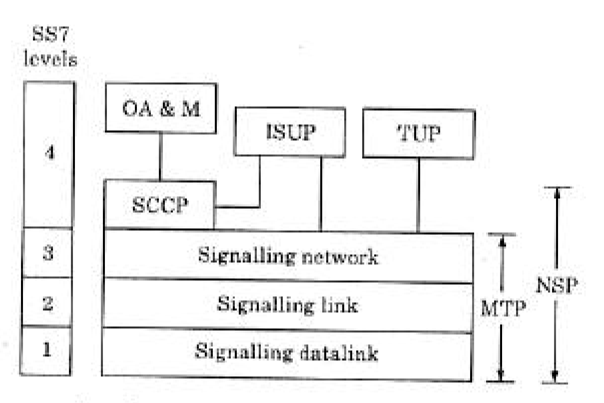


Fig.31.4 Formats of signaling units

The service information octet specifies the type of message and whether the message relates to a national or international network. This signalling information field (SIF) contains level 3 and 4 information. This consists of an address label and user data from the telephone user part (TUP), ISDN user part (ISUP) or operations, administration and maintenance (OA&M) part. The address label field is 32 bits long with a 14-bit signalling link selection subfield. The link selection subfield is used to select a path .from among alternative routes.

The level 3 signalling network functions relate to message handling and network management. Message handling involves discrimination, routing and distribution of messages. The discrimination function analyses the destination code in the address label to decide whether a message is to be routed to another node or distributed to one of the user parts in the local node. The particular user part to which a message is to be distributed at the destination node is decided based on an analysis of the type of message information in the SER field.

The discrimination function is needed only in STPs. Routing function may be invoked by the discrimination function or by a local level 4 entity. The routing decision is based on the value in the signalling link selection subfield. With a 4-bit link selection subfield, a total of 16 different routes may be defined through the network. In general, all of the control signals associated with a single call will follow the same route. This guarantees that the control messages arrive in sequence at the destination.

The main objective of network management function is to monitor signalling links and overcome link failures or degradations. The goal is to achieve an unavailability figure of not more than 10 minutes per year per route. This is attempted through redundancy of links and dynamic rerouting. The emphasis on the network management in SS7 stems from the fact that the functioning of the common signalling network is critical and a failure in the signalling network may affect all the subscribers. Since failure recovery and restoration actions may involve multiple networks across national boundaries, it is essential to lay down failure and congestion recovery procedures.

The signalling connection control part (SCCP) residing in level 4 was added to SS7 specifications in 1984. The SCCP and MTP together are referred to as the network service part (NSP). The main purpose of SCCP is to enhance the limited routing, distribution and addressing capabilities of the third layer, to meet the needs of those user parts, requiring enriched information transfer facilities. The other modules in level 4 are invoked as per the requirements of a service. The TUP is invoked in response to actions by a subscriber at a telephone. The control signalling associated with TUP deal with establishment, maintenance, and termination of telephone calls. Similarly, ISUP deals with ISDN subscriber calls and related functions. Finally, the OA&M module deals with messages relating to network management, operations and maintenance.

The SS7 architectural principles are similar to the ones in ISO open system interconnection (OSI) architecture proposed for data networks. Many of the functions in each of these layers are common, in these two architectures. In fact, SS7 architecture came into existence much after the ISO-OSI model and is largely based on the work done for ISO-OSI model.