



SS7 Tutorial

Table of Contents

SS7 Tutorial.....	1
SS7 Tutorial.....	3
Overview	3
SS7 Protocol Stack.....	6
Message Transfer Part	7
ISDN User Part	13
Signaling Connection Control Part.....	20
Transaction Capabilities Application Part	21
Other SS7 Information	23
Bibliography	23
Comments? Questions?	23

List of Figures

Figure 1. SS7 Signaling Points.....	4
Figure 2. SS7 Signaling Link Types	5
Figure 3. The OSI Reference Model and the SS7 Protocol Stack	6
Figure 4. SS7 Signal Units	8
Figure 5. Message Type Length Indicator Value(s)	9
Figure 6. Service Indicator Values	11
Figure 7. ANSI vs. ITU-T SIO and SIF	12
Figure 8. Basic ISUP Signaling	14
Figure 9. ISUP Message Format.....	16
Figure 10. ANSI and ITU-T Initial Address Message (IAM) Format.....	17
Figure 11. ANSI and ITU-T Address Complete Message (ACM) Format	18
Figure 12. ANSI and ITU-T Answer Message (ANM) Format.....	18
Figure 13. ANSI and ITU-T Release (REL) Message Format.....	19
Figure 14. ANSI and ITU-T Release Complete (RLC) Message Format	19
Figure 15. SCCP Message Format	21

SS7 Tutorial

Overview

Common Channel Signaling System No. 7 (i.e., **SS7** or **C7**) is a global standard for telecommunications defined by the [International Telecommunication Union](#) (ITU) [Telecommunication Standardization Sector](#) (ITU-T). The standard defines the procedures and protocol by which network elements in the public switched telephone network (PSTN) exchange information over a digital signaling network to effect wireless (cellular) and wireline call setup, routing and control. The ITU definition of SS7 allows for national variants such as the [American National Standards Institute](#) (ANSI) and [Bell Communications Research](#) (Telcordia Technologies) standards used in North America and the [European Telecommunications Standards Institute](#) (ETSI) standard used in Europe.

The SS7 network and protocol are used for:

- basic call setup, management, and tear down
- wireless services such as personal communications services (PCS), wireless roaming, and mobile subscriber authentication
- local number portability (LNP)
- toll-free (800/888) and toll (900) wireline services
- enhanced call features such as call forwarding, calling party name/number display, and three-way calling
- efficient and secure worldwide telecommunications

Signaling Links

SS7 messages are exchanged between network elements over 56 or 64 kilobit per second (kbps) bidirectional channels called **signaling links**. Signaling occurs **out-of-band** on dedicated channels rather than **in-band** on voice channels. Compared to in-band signaling, out-of-band signaling provides:

- faster call setup times (compared to in-band signaling using multi-frequency (MF) signaling tones)
- more efficient use of voice circuits
- support for **Intelligent Network** (IN) services which require signaling to network elements without voice trunks (e.g., database systems)
- improved control over fraudulent network usage

Signaling Points

Each signaling point in the SS7 network is uniquely identified by a numeric **point code**. Point codes are carried in signaling messages exchanged between signaling points to identify the source and destination of each message. Each signaling point uses a routing table to select the appropriate signaling path for each message.

There are three kinds of signaling points in the SS7 network (Fig. 1):

- **SSP** (Service Switching Point)
- **STP** (Signal Transfer Point)
- **SCP** (Service Control Point)

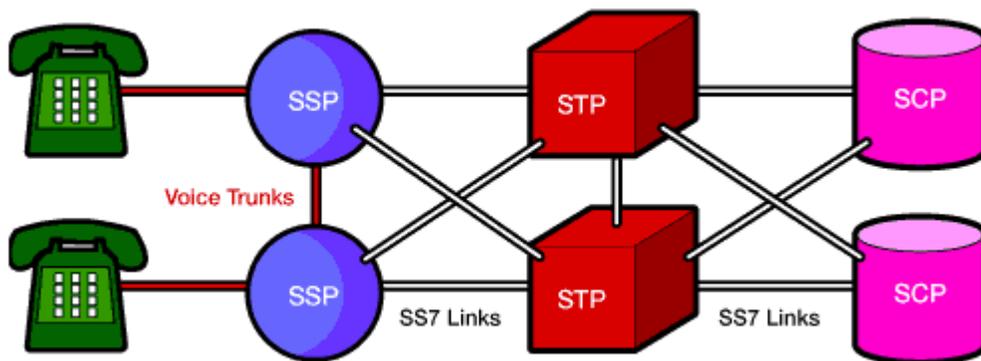


Figure 1. SS7 Signaling Points

SSPs are switches that originate, terminate, or tandem calls. An SSP sends signaling messages to other SSPs to setup, manage, and release voice circuits required to complete a call. An SSP may also send a query message to a centralized database (an **SCP**) to determine how to route a call (e.g., a toll-free 1-800/888 call in North America). An SCP sends a response to the originating SSP containing the routing number(s) associated with the dialed number. An alternate routing number may be used by the SSP if the primary number is busy or the call is unanswered within a specified time. Actual call features vary from network to network and from service to service.

Network traffic between signaling points may be routed via a packet switch called an **STP**. An STP routes each incoming message to an outgoing signaling link based on routing information contained in the SS7 message. Because it acts as a network hub, an STP provides improved utilization of the SS7 network by eliminating the need for direct links between signaling points. An STP may perform **global title translation**, a procedure by which the destination signaling point is determined from digits present in the signaling message (e.g., the dialed 800 number, calling card number, or mobile subscriber identification number). An STP can also act as a "firewall" to screen SS7 messages exchanged with other networks.

Because the SS7 network is critical to call processing, SCPs and STPs are usually deployed in mated pair configurations in separate physical locations to ensure network-wide service in the event of an isolated failure. Links between signaling points are also provisioned in pairs. Traffic is shared across all links in the linkset. If one of the links fails, the signaling traffic is rerouted over another link in the **linkset**. The SS7 protocol provides both error correction and retransmission capabilities to allow continued service in the event of signaling point or link failures.

SS7 Signaling Link Types

Signaling links are logically organized by link type ("A" through "F") according to their use in the SS7 signaling network.

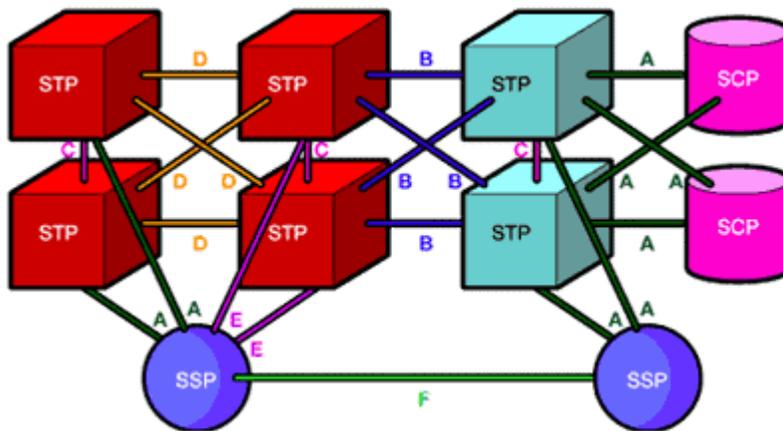


Figure 2. SS7 Signaling Link Types

A Link:	An "A" (access) link connects a signaling end point (e.g., an SCP or SSP) to an STP. Only messages originating from or destined to the signaling end point are transmitted on an "A" link.
B Link:	A "B" (bridge) link connects an STP to another STP. Typically, a quad of "B" links interconnect peer (or primary) STPs (e.g., the STPs from one network to the STPs of another network). The distinction between a "B" link and a "D" link is rather arbitrary. For this reason, such links may be referred to as "B/D" links.
C Link:	A "C" (cross) link connects STPs performing identical functions into a mated pair . A "C" link is used only when an STP has no other route available to a destination signaling point due to link failure(s). Note that SCPs may also be deployed in pairs to improve reliability; unlike STPs, however, mated SCPs are not interconnected by signaling links.
D Link:	A "D" (diagonal) link connects a secondary (e.g., local or regional) STP pair to a primary (e.g., inter-network gateway) STP pair in a quad-link configuration. Secondary STPs within the same network are connected via a quad of "D" links. The distinction between a "B" link and a "D" link is rather arbitrary. For this reason, such links may be referred to as "B/D" links.
E Link:	An "E" (extended) link connects an SSP to an alternate STP. "E" links provide an alternate signaling path if an SSP's "home" STP cannot be reached via an "A" link. "E" links are not usually provisioned unless the benefit of a marginally higher degree of reliability justifies the added expense.
F Link:	An "F" (fully associated) link connects two signaling end points (i.e., SSPs and SCPs). "F" links are not usually used in networks with STPs. In networks without STPs, "F" links directly connect signaling points.

SS7 Protocol Stack

The hardware and software functions of the SS7 protocol are divided into functional abstractions called "levels". These levels map loosely to the **Open Systems Interconnect (OSI)** 7-layer model defined by the [International Standards Organization \(ISO\)](http://www.iso.org).

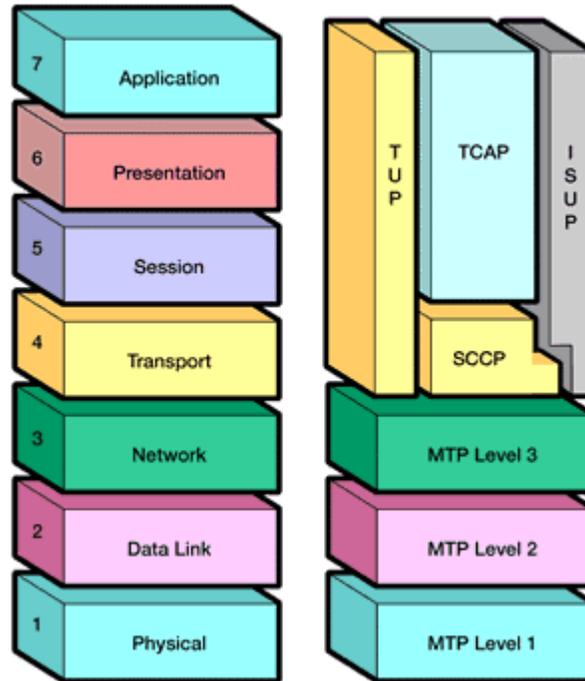


Figure 3. The OSI Reference Model and the SS7 Protocol Stack

Message Transfer Part

The Message Transfer Part (MTP) is divided into three levels. The lowest level, **MTP Level 1**, is equivalent to the OSI Physical Layer. MTP Level 1 defines the physical, electrical, and functional characteristics of the digital signaling link. Physical interfaces defined include **E-1** (2048 kb/s; 32 64 kb/s channels), **DS-1** (1544 kb/s; 24 64kb/s channels), **V.35** (64 kb/s), **DS-0** (64 kb/s), and **DS-0A** (56 kb/s).

MTP Level 2 ensures accurate end-to-end transmission of a message across a signaling link. Level 2 implements flow control, message sequence validation, and error checking. When an error occurs on a signaling link, the message (or set of messages) is retransmitted. MTP Level 2 is equivalent to the OSI Data Link Layer.

MTP Level 3 provides message routing between signaling points in the SS7 network. MTP Level 3 re-routes traffic away from failed links and signaling points and controls traffic when congestion occurs. MTP Level 3 is equivalent to the OSI Network Layer.

ISDN User Part (ISUP)

The ISDN User Part (ISUP) defines the protocol used to set-up, manage, and release trunk circuits that carry voice and data between terminating line exchanges (e.g., between a calling party and a called party). ISUP is used for both ISDN and non-ISDN calls. However, calls that originate and terminate at the same switch do not use ISUP signaling.

Telephone User Part (TUP)

In some parts of the world (e.g., China, Brazil), the Telephone User Part (TUP) is used to support basic call setup and tear-down. TUP handles analog circuits only. In many countries, ISUP has replaced TUP for call management.

Signaling Connection Control Part (SCCP)

SCCP provides connectionless and connection-oriented network services and **global title translation** (GTT) capabilities above MTP Level 3. A **global title** is an address (e.g., a dialed 800 number, calling card number, or mobile subscriber identification number) which is translated by SCCP into a destination point code and **subsystem number**. A subsystem number uniquely identifies an application at the destination signaling point. SCCP is used as the transport layer for TCAP-based services. **Transaction Capabilities Applications Part (TCAP)**

TCAP supports the exchange of non-circuit related data between applications across the SS7 network using the SCCP connectionless service. Queries and responses sent between SSPs and SCPs are carried in TCAP messages. For example, an SSP sends a TCAP query to determine the routing number associated with a dialed 800/888 number and to check the personal identification number (PIN) of a calling card user. In mobile networks (**IS-41** and **GSM**), TCAP carries **Mobile Application Part (MAP)** messages sent between mobile switches and databases to support user authentication, equipment identification, and roaming.

Operations, Maintenance and Administration Part (OMAP) and ASE

OMAP and ASE are areas for future definition. Presently, OMAP services may be used to verify network routing databases and to diagnose link problems.

Message Transfer Part

The Message Transfer Part (MTP) is divided into three levels:

MTP Level 1

The lowest level, MTP Level 1, is equivalent to the [OSI Physical Layer](#). MTP Level 1 defines the physical, electrical, and functional characteristics of the digital signaling link. Physical interfaces defined include **E-1** (2048 kb/s; 32 64 kb/s channels), **DS-1** (1544 kb/s; 24 64 kp/s channels), **V.35** (64 kb/s), **DS-0** (64 kb/s), and **DS-0A** (56 kb/s).

MTP Level 2

MTP Level 2 ensures accurate end-to-end transmission of a message cross a signaling link. Level 2 implements flow control, message sequence validation, and error checking. When an error occurs on a signaling link, the message (or set of messages) is retransmitted. MTP Level 2 is equivalent to the [OSI Data Link Layer](#).

An SS7 message is called a **signal unit** (SU). There are three kinds of signal units: **Fill-In Signal Units** (FISUs), **Link Status Signal Units** (LSSUs), and **Message Signal Units** (MSUs) (Fig. 4).

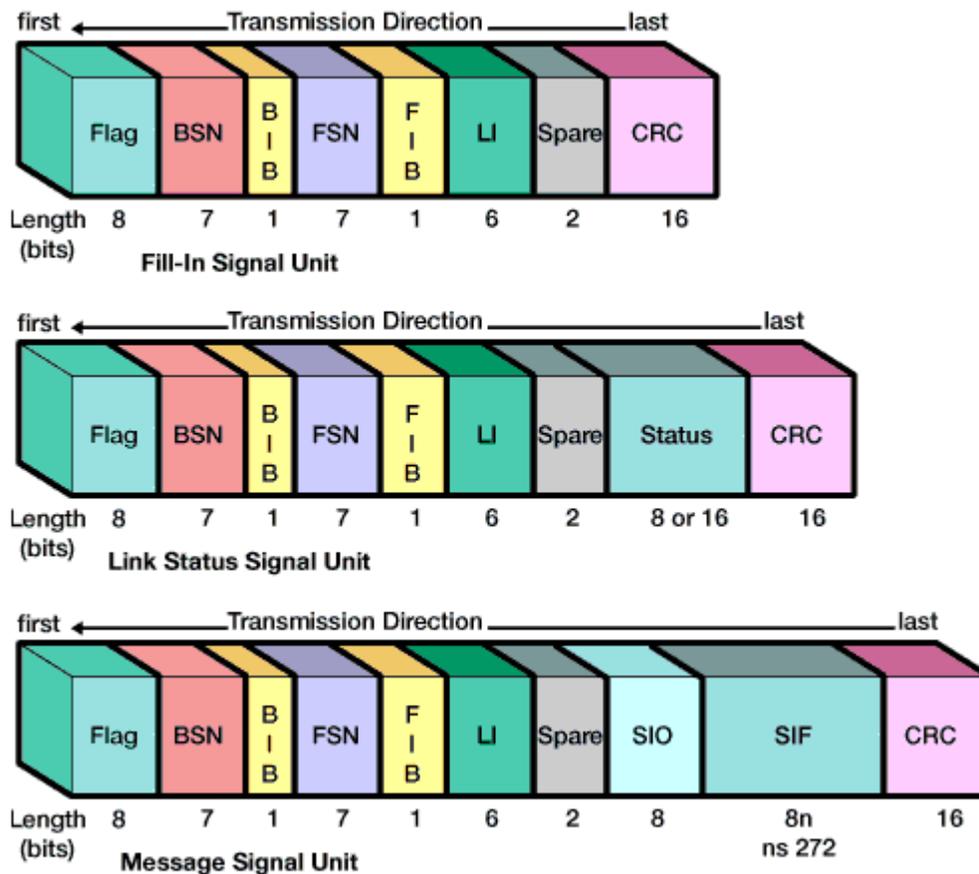


Figure 4. SS7 Signal Units

Fill-In Signal Units (FISUs) are transmitted continuously on a signaling link in both directions unless other signal units (MSUs or LSSUs) are present. FISUs carry basic level 2 information only (e.g., acknowledgment of signal unit receipt by a remote signaling point). Because a [CRC checksum](#) is calculated for each FISU, signaling link quality is checked continuously by both signaling points at either end of the link. (Note: In the ITU-T Japan variant, signaling link quality is checked by the continuous transmission of [flag](#) octets (8-bit bytes) rather than FISUs; FISUs are sent only at predefined timer intervals (e.g., once every 150 milliseconds).

Link Status Signal Units (LSSUs) carry one or two **octets** (8-bit bytes) of link status information between signaling points at either end of a link. The link status is used to control link alignment and to indicate the status of a signaling point (e.g., local processor outage) to the remote signaling point.

Message Signal Units (MSUs) carry all call control, database query and response, network management, and network maintenance data in the signaling information field ([SIF](#)). MSUs have a **routing label** which allows an originating signaling point to send information to a destination signaling point across the network.

The value of the **LI** (Length Indicator) field determines the signal unit type:

LI Value	Signal Unit Type
0	Fill-In Signal Unit (FISU)
1..2	Link Status Signal Unit (LSSU)
3..63	Message Signal Unit (MSU)

Figure 5. Message Type Length Indicator Value(s)

The 6-bit LI can store values between zero and 63. If the number of octets which follow the LI and precede the CRC is less than 63, the LI contains this number. Otherwise, the LI is set to 63. An LI of 63 indicates that the message length is equal to *or greater than* 63 octets (up to a maximum of 273 octets). The maximum length of a signal unit is 279 octets: 273 octets (data) + 1 octet (flag) + 1 octet (BSN + BIB) + 1 octet (FSN + FIB) + 1 octet (LI + 2 bits spare) + 2 octets (CRC).

Flag

The flag indicates the beginning of a new signal unit and implies the end of the previous signal unit (if any). The binary value of the flag is **0111 1110**. Before transmitting a signal unit, MTP Level 2 removes "false flags" by adding a zero-bit after any sequence of five one-bits. Upon receiving a signal unit and stripping the flag, MTP Level 2 removes any zero-bit following a sequence of five one-bits to restore the original contents of the message. Duplicate flags are removed between signal units.

BSN (Backward Sequence Number)

The BSN is used to acknowledge the receipt of signal units by the remote signaling point. The BSN contains the sequence number of the signal unit being acknowledged. (See description under **FIB** below.)

BIB (Backward Indicator Bit)

The BIB indicates a negative acknowledgment by the remote signaling point when toggled. (See description under **FIB** below.)

FSN (Forward Sequence Number)

The FSN contains the sequence number of the signal unit. (See description under **FIB** below.)

FIB (Forward Indicator Bit)

The FIB is used in error recovery like the BIB. When a signal unit is ready for transmission, the signaling point increments the FSN (forward sequence number) by 1 (FSN = 0..127). The CRC (cyclic redundancy check) checksum value is calculated and appended to the forward message. Upon receiving the message, the remote signaling point checks the CRC and copies the value of the FSN into the BSN of the next available message scheduled for transmission back to the initiating signaling point. If the CRC is correct, the backward message is transmitted. If the CRC is incorrect, the remote signaling point indicates negative acknowledgment by toggling the BIB prior to sending the backward message. When the originating signaling point receives a negative acknowledgment, it retransmits all forward messages, beginning with the corrupted message, with the FIB toggled.

Because the 7-bit FSN can store values between zero and 127, a signaling point can send up to 128 signal units before requiring acknowledgment from the remote signaling point. The BSN indicates the last in-sequence signal unit received correctly by the remote signaling point. The BSN acknowledges all previously received signal units as well. For example, if a signaling point receives a signal unit with BSN = 5 followed by another with BSN = 10 (and the BIB is not toggled), the latter BSN implies successful receipt of signal units 6 through 9 as well.

SIO (Service Information Octet)

The SIO field in an MSU contains the 4-bit subservice field followed by the 4-bit service indicator. FISUs and LSSUs do not contain an SIO.

The **subservice field** contains the network indicator (e.g., national or international) and the message priority (0..3 with 3 being the highest priority). Message priority is considered only under congestion conditions, not to control the order in which messages are transmitted. Low priority messages may be discarded during periods of congestion. Signaling link test messages receive a higher priority than call setup messages.

The **service indicator** specifies the MTP user (Fig. 6), thereby allowing the decoding of the information contained in the [SIF](#).

Service Indicator	MTP User
0	Signaling Network Management Message (SNM)
1	Maintenance Regular Message (MTN)
2	Maintenance Special Message (MTNS)
3	Signaling Connection Control Part (SCCP)
4	Telephone User Part (TUP)
5	ISDN User Part (ISUP)
6	Data User Part (call and circuit-related messages)
7	Data User Part (facility registration/cancellation messages)

Figure 6. Service Indicator Values

SIF (Signaling Information Field)

The SIF in an MSU contains the **routing label** and signaling information (e.g., SCCP, [TCAP](#), and ISUP message data). LSSUs and FISUs contain neither a routing label nor an SIO as they are sent between two directly connected signaling points. For more information about routing labels, refer to the description of MTP Level 3 below.

CRC (Cyclic Redundancy Check)

The CRC value is used to detect and correct data transmission errors. For more information, see the description for BIB above.

MTP Level 3

MTP Level 3 provides message routing between signaling points in the SS7 network. MTP Level 3 is equivalent in function to the OSI Network Layer.

MTP Level 3 routes messages based on the routing label in the signaling information field (SIF) of message signal units. The routing label is comprised of the **destination point code** (DPC), **originating point code** (OPC), and **signaling link selection** (SLS) field. Points codes are numeric addresses which uniquely identify each signaling point in the SS7 network. When the destination point code in a message indicates the receiving signaling point, the message is distributed to the appropriate user part (e.g., ISUP or SCCP) indicated by the service indicator in the SIO. Messages destined for other signaling points are transferred provided that the receiving signaling point has message transfer capabilities (like an STP). The selection of outgoing link is based on information in the DPC and SLS.

An ANSI routing label uses 7 octets; an ITU-T routing label uses 4 octets (Fig. 7).

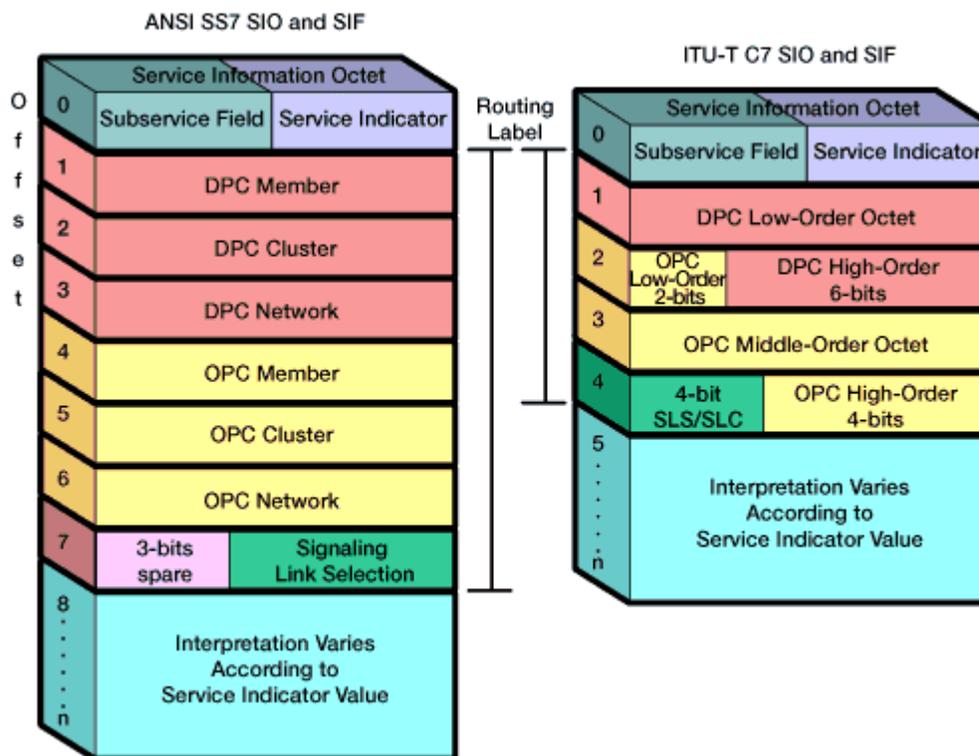


Figure 7. ANSI vs. ITU-T SIO and SIF

ANSI point codes use 24-bits (three octets); ITU-T point codes typically use 14-bits. For this reason, signaling information exchanged between ANSI and ITU-T networks must be routed through a gateway STP, protocol converter, or other signaling point which has both an ANSI and an ITU-T point code. (Note: China uses 24-bit ITU-T point codes which are incompatible with both ANSI and other ITU-T networks). Interaction between ANSI and ITU-T networks is further complicated by different implementations of higher level protocols and procedures.

An ANSI point code consists of network, cluster, and member octets (e.g., 245-16-0). An octet is an 8-bit byte which can contain any value between zero and 255. Telcos with large networks have unique network identifiers while smaller operators are assigned a unique cluster number within networks 1 through 4 (e.g., 1-123-9). Network number 0 is not used; network number 255 is reserved for future use.

ITU-T point codes are pure binary numbers which may be stated in terms of zone, area/network, and signaling point identification numbers. For example, the point code 5557 (decimal) may be stated as 2-182-5 (binary 010 10110110 101).

Signaling Link Selection (SLS)

The selection of outgoing link is based on information in the DPC and Signaling Link Selection field. The SLS is used to:

- Ensure message sequencing. Any two messages sent with the same SLS will always arrive at the destination in the same order in which they were originally sent.
- Allow equal load sharing of traffic among all available links. In theory, if a user part sends messages at regular intervals and assigns the SLS values in a round-robin fashion, the traffic level should be equal among all links (within the combined linkset) to that destination.

In ANSI networks, the size of the SLS field was originally 5 bits (32 values). In configurations with two links in each linkset of a combined linkset (totaling 4 links), 8 SLS values were assigned to each link to allow an equal balance of traffic.

A problem arose when growing networks provisioned linksets beyond 4 links. With a 5 bit SLS, a configuration with 5 links in each linkset of a combined linkset (totaling 10 links) results in an uneven assignment of 3 SLS values for 8 links and 4 SLS values for the remaining 2 links. To eliminate this problem, both ANSI and Bellcore moved to adopt an 8-bit SLS (256 values) to provide better loadsharing across signaling links.

In ITU-T implementations, the SLS is interpreted as the **signaling link code** in MTP messages. In ITU-T Telephone User Part message, a portion of the circuit identification code is stored in the SLS field.

MTP Level 3 re-routes traffic away from failed links and signaling points and controls traffic when congestion occurs. However, a detailed discussion of this topic is outside the scope of this tutorial.

MTP Levels 2 and 1 can be replaced by **ATM** (Asynchronous Transfer Mode), a simple broadband protocol which uses fixed-length 53 octet **cells**. MTP Level 3 interfaces to ATM using the **Signaling ATM Adaptation Layer** (SAAL). This interface is currently an area of ongoing study.

ISDN User Part

The ISDN User Part (ISUP) defines the protocol and procedures used to set-up, manage, and release trunk circuits that carry voice and data calls over the public switched telephone network (PSTN). ISUP is used for both ISDN and non-ISDN calls. Calls that originate and terminate at the same switch do not use ISUP signaling.

Basic ISUP Call Control

Figure 8 depicts the ISUP signaling associated with a basic call.

1. When a call is placed to an out-of-switch number, the originating SSP transmits an ISUP **initial address message (IAM)** to reserve an idle trunk circuit from the originating switch to the destination switch (**1a**). The IAM includes the [originating point code](#), [destination point code](#), **circuit identification code** (circuit "5" in Fig. 8), dialed digits and, optionally, the calling party number and name. In the example below, the IAM is routed via the home STP of the originating switch to the destination switch (**1b**). Note that the same signaling link(s) are used for the duration of the call unless a link failure condition forces a switch to use an alternate signaling link.

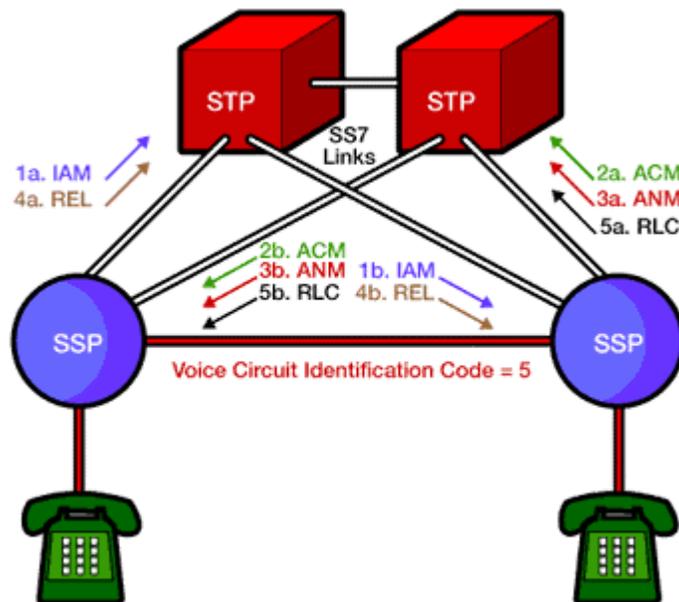


Figure 8. Basic ISUP Signaling

2. The destination switch examines the dialed number, determines that it serves the called party, and that the line is available for ringing. The destination switch rings the called party line and transmits an ISUP address complete message (ACM) to the originating switch (2a) (via its home STP) to indicate that the remote end of the trunk circuit has been reserved. The STP routes the ACM to the originating switch (2b) which rings the calling party's line and connects it to the trunk to complete the voice circuit from the calling party to the called party.

In the example shown above, the originating and destination switches are directly connected with trunks. If the originating and destination switches are not directly connected with trunks, the originating switch transmits an IAM to reserve a trunk circuit to an intermediate switch. The intermediate switch sends an ACM to acknowledge the

- circuit reservation request and then transmits an IAM to reserve a trunk circuit to another switch. This process continues until all trunks required to complete the voice circuit from the originating switch to the destination switch are reserved.
3. When the called party picks up the phone, the destination switch terminates the ringing tone and transmits an ISUP **answer message** (ANM) to the originating switch via its home STP (**3a**). The STP routes the ANM to the originating switch (**3b**) which verifies that the calling party's line is connected to the reserved trunk and, if so, initiates billing.
 4. If the calling party hangs-up first, the originating switch sends an ISUP **release message** (REL) to release the trunk circuit between the switches (**4a**). The STP routes the REL to the destination switch (**4b**). If the called party hangs up first, or if the line is busy, the destination switch sends an REL to the originating switch indicating the release cause (e.g., normal release or busy).
 5. Upon receiving the REL, the destination switch disconnects the trunk from the called party's line, sets the trunk state to idle, and transmits an ISUP **release complete message** (RLC) to the originating switch (**5a**) to acknowledge the release of the remote end of the trunk circuit. When the originating switch receives (or generates) the RLC (**5b**), it terminates the billing cycle and sets the trunk state to idle in preparation for the next call.

ISUP messages may also be transmitted during the connection phase of the call (i.e., between the ISUP Answer (ANM) and Release (REL) messages).

ISUP Message Format

ISUP information is carried in the [Signaling Information Field](#) (SIF) of an MSU. The SIF contains the **routing label** followed by a 14-bit (ANSI) or 12-bit (ITU) **circuit identification code** (CIC). The CIC indicates the trunk circuit reserved by the originating switch to carry the call. The CIC is followed by the **message type** field (e.g., IAM, ACM, ANM, REL, RLC) which defines the contents of the remainder of the message (Fig. 9).

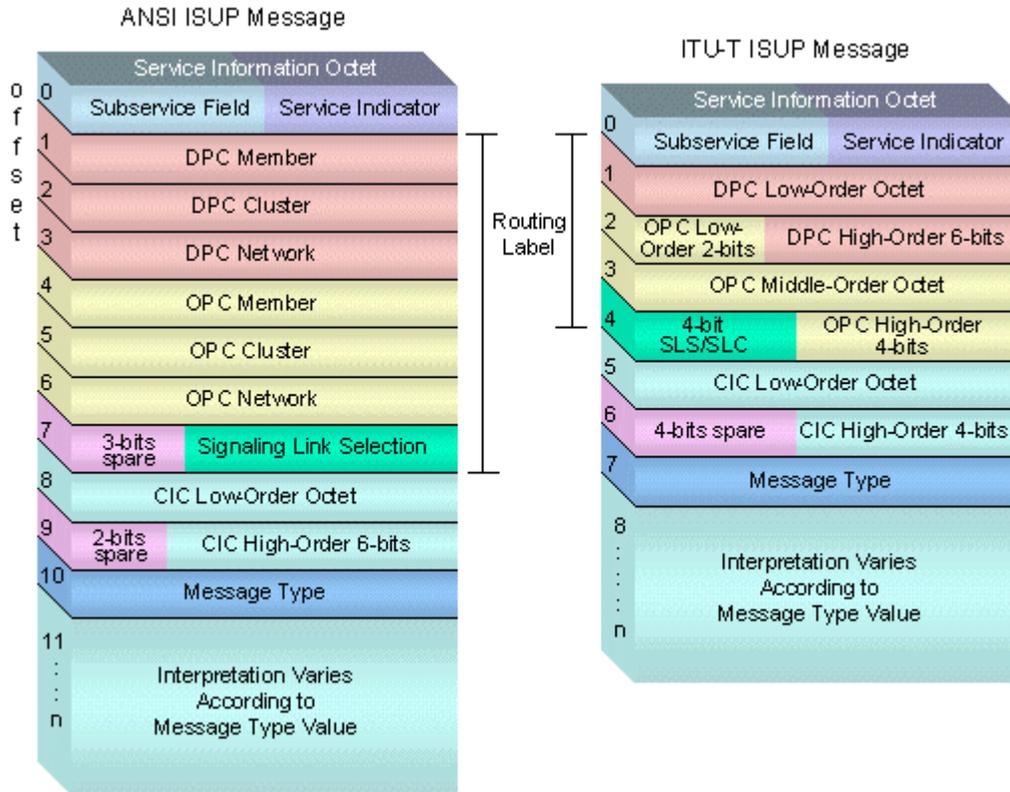


Figure 9. ISUP Message Format

Each ISUP message contains a **mandatory fixed part** containing mandatory fixed-length parameters. Sometimes the mandatory fixed part is comprised only of the message type field. The mandatory fixed part may be followed by the **mandatory variable part** and/or the **optional part**. The mandatory variable part contains mandatory variable-length parameters. The optional part contains optional parameters which are identified by a one-octet parameter code followed by a length indicator ("octets to follow") field. Optional parameters may occur in any order. If optional parameters are included, the end of the optional parameters is indicated by an octet containing all zeros.

Initial Address Message

An Initial Address Message (IAM) is sent in the "forward" direction by each switch needed to complete the circuit between the calling party and called party until the circuit connects to the destination switch. An IAM contains the called party number in the mandatory variable part and may contain the calling party name and number in the optional part.

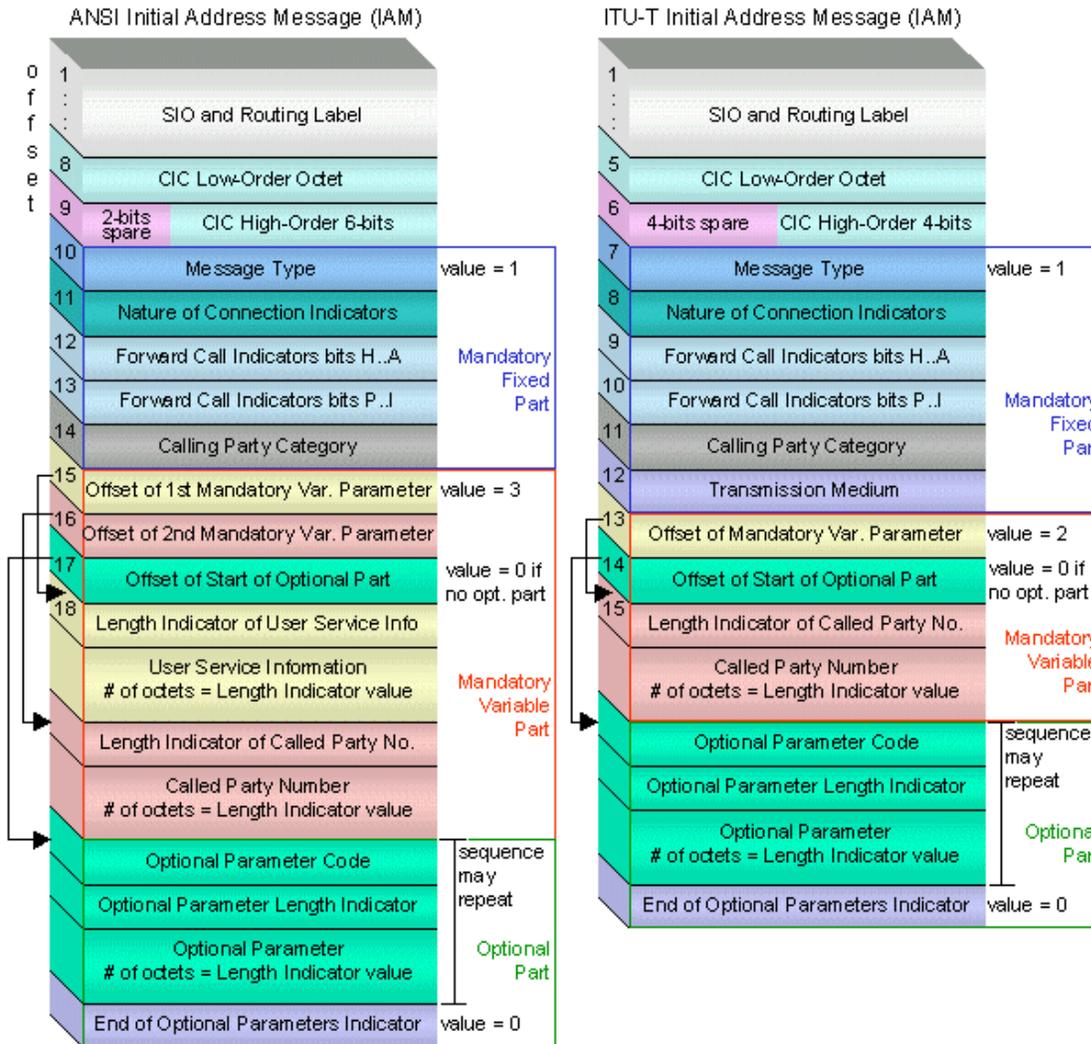


Figure 10. ANSI and ITU-T Initial Address Message (IAM) Format

Address Complete Message

An Address Complete Message (ACM) is sent in the "backward" direction to indicate that the remote end of a trunk circuit has been reserved.

The originating switch responds to an ACM message by connecting the calling party's line to the trunk to complete the voice circuit from the calling party to the called party. The originating switch also sends a ringing tone to the calling party's line.

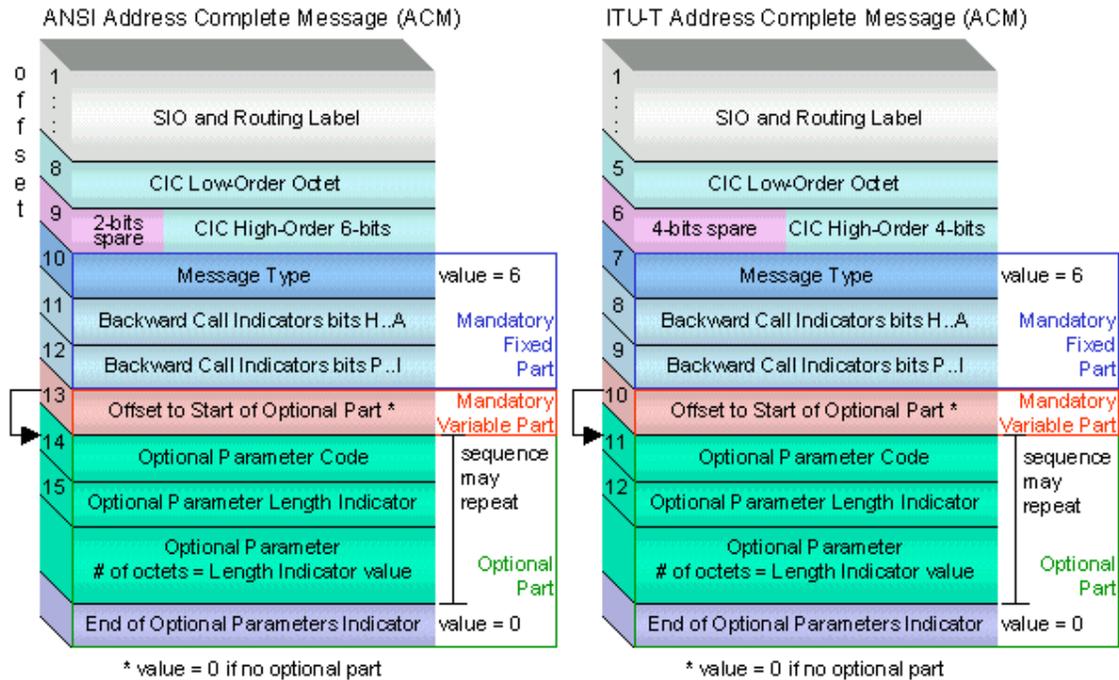


Figure 11. ANSI and ITU-T Address Complete Message (ACM) Format

When the called party answers, the destination switch terminates the ringing tone and sends an Answer Message (ANM) to the originating switch. The originating switch initiates billing after verifying that the calling party's line is connected to the reserved trunk.

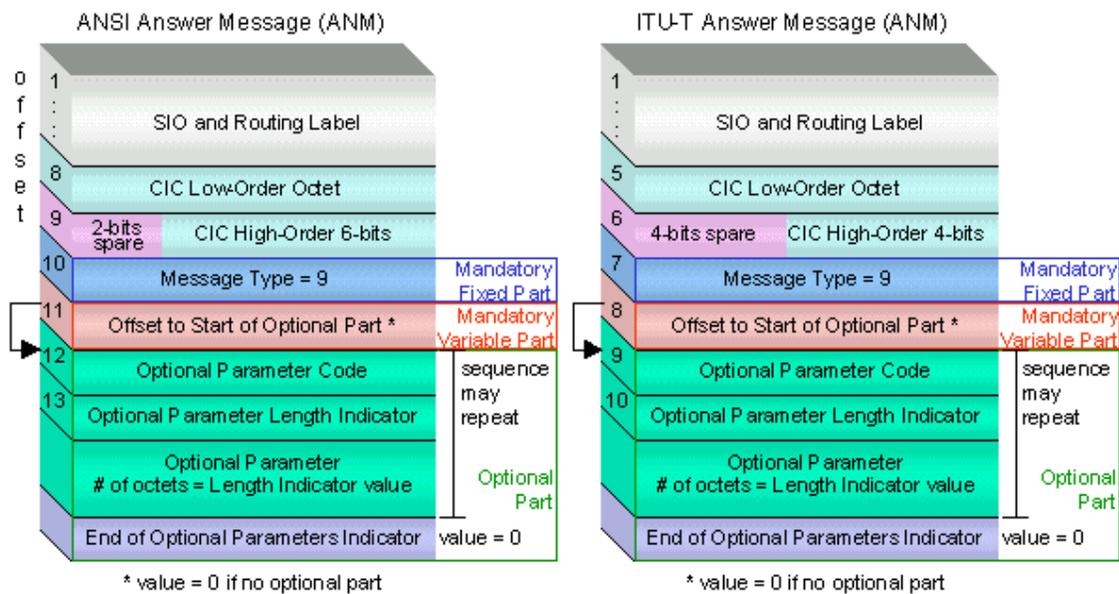


Figure 12. ANSI and ITU-T Answer Message (ANM) Format

Release Message

A Release Message (REL) is sent in either direction indicating that the circuit is being released due to the **cause indicator** specified. An REL is sent when either the calling or called party "hangs up" the call (cause = 16). An REL is also sent in the backward direction if the called party line is busy (cause = 17).

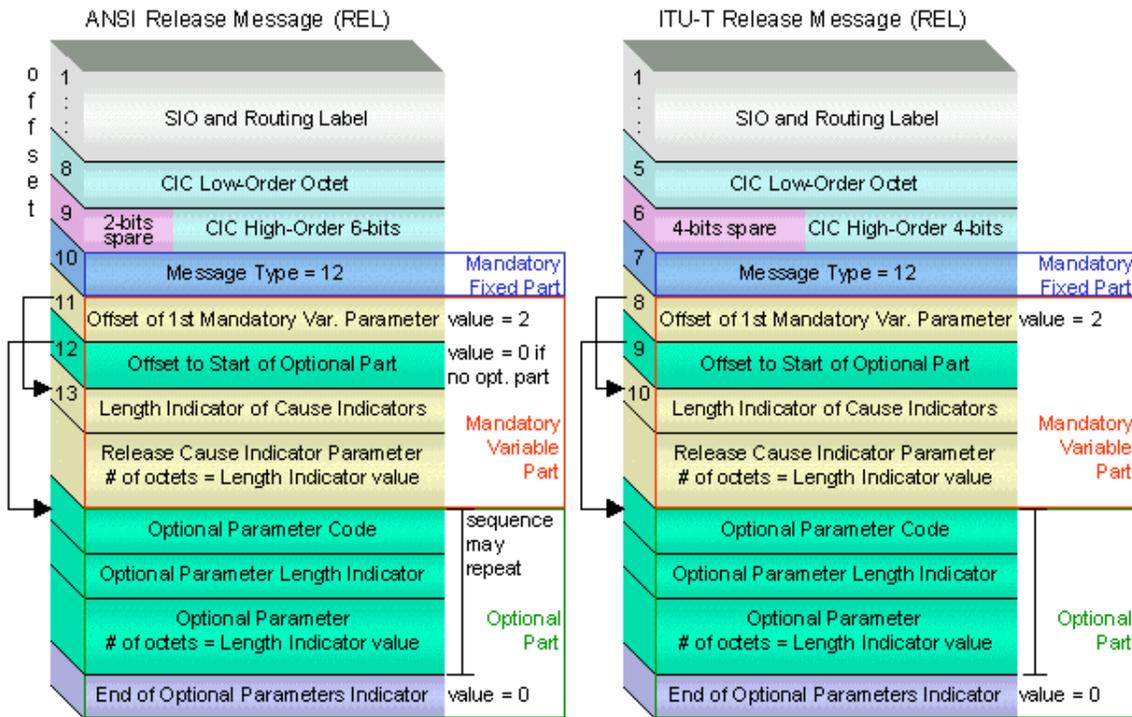


Figure 13. ANSI and ITU-T Release (REL) Message Format

Release Complete Message

A Release Complete Message (RLC) is sent in the opposite direction of the REL to acknowledge the release of the remote end of a trunk circuit and end the billing cycle as appropriate.

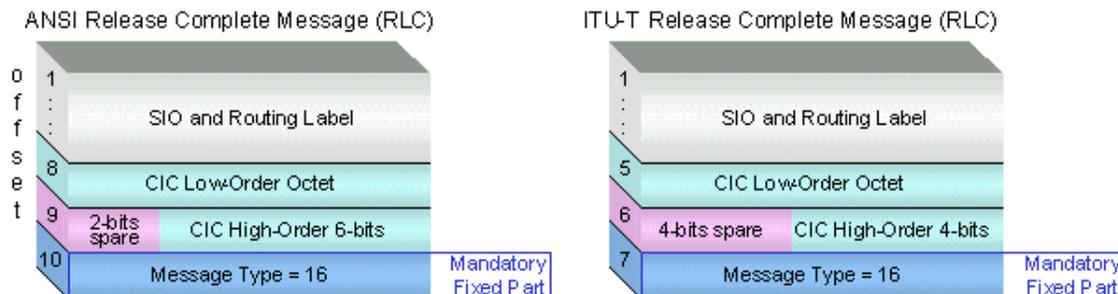


Figure 14. ANSI and ITU-T Release Complete (RLC) Message Format

Telephone User Part

In some parts of the world (e.g., China), the **Telephone User Part** (TUP) supports basic call processing. TUP handles analog circuits only; digital circuits and data transmission capabilities are provided by the **Data User Part**.

Signaling Connection Control Part

SCCP provides connectionless and connection-oriented network services above [MTP Level 3](#). While MTP Level 3 provides point codes to allow messages to be addressed to specific signaling points, SCCP provides **subsystem numbers** to allow messages to be addressed to specific applications (called **subsystems**) at these signaling points. SCCP is used as the transport layer for [TCAP](#)-based services such as freephone (800/888), calling card, local number portability, wireless roaming, and **personal communications services** (PCS). **Global Title Translation**

SCCP also provides the means by which an [STP](#) can perform **global title translation** (GTT), a procedure by which the destination signaling point and subsystem number (SSN) is determined from digits (i.e., the **global title**) present in the signaling message.

The global title digits may be any sequence of digits (e.g., the dialed 800/888 number, calling card number, or mobile subscriber identification number) pertinent to the service requested. Because an STP provides global title translation, originating signaling points do not need to know the destination point code or subsystem number of the associated service. Only the STPs need to maintain a database of destination point codes and subsystem numbers associated with specific services and possible destinations.

SCCP Message Format

The Service Indicator of the Service Information Octet (SIO) is coded 3 (binary 0011) for SCCP. SCCP messages are contained within the Signaling Information Field ([SIF](#)) of an MSU. The SIF contains the routing label followed by the SCCP message contents. The SCCP message is comprised of a one-octet **message type** field which defines the contents of the remainder of the message (Fig. 15).

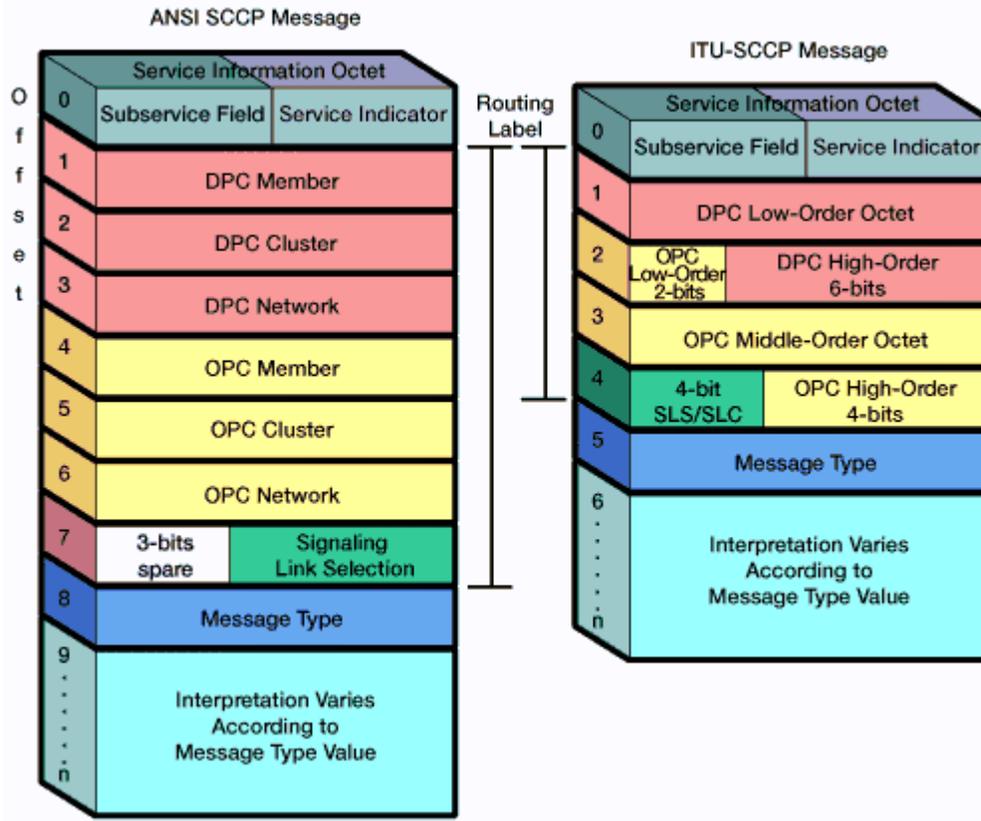


Figure 15. SCCP Message Format

Each SCCP message contains a **mandatory fixed part** (mandatory fixed-length parameters), **mandatory variable part** (mandatory variable-length parameters), and an **optional part** which may contain fixed-length and variable-length fields. Each optional part parameter is identified by a one-octet parameter code followed by a length indicator ("octets to follow") field. Optional parameters may occur in any order. If optional parameters are included, the end of the optional parameters is indicated by an octet containing all zeros.

Transaction Capabilities Application Part

TCAP enables the deployment of advanced *intelligent network* services by supporting non-circuit related information exchange between signaling points using the SCCP connectionless service. An SSP uses TCAP to query an SCP to determine the routing number(s) associated with a dialed 800, 888, or 900 number. The SCP uses TCAP to return a response containing the routing number(s) (or an error or reject component) back to the SSP. Calling card calls are also validated using TCAP query and response messages. When a mobile subscriber roams into a new **mobile switching center** (MSC) area, the integrated **visitor location register** requests service profile information from the subscriber's **home location register** (HLR) using **mobile application part** (MAP) information carried within TCAP messages.

TCAP messages are contained within the SCCP portion of an MSU. A TCAP message is comprised of a *transaction portion* and a *component portion*.

Transaction Portion

The transaction portion contains the package type identifier. There are seven package types:

- **Unidirectional:** Transfers component(s) in one direction only (no reply expected).
- **Query with Permission:** Initiates a TCAP transaction (e.g., a 1-800 query). The destination node may end the transaction.
- **Query without Permission:** Initiates a TCAP transaction. The destination node may *not* end the transaction.
- **Response:** Ends the TCAP transaction. A response to an 1-800 query with permission may contain the routing number(s) associated with the 800 number.
- **Conversation with Permission:** Continues a TCAP transaction. The destination node may end the transaction.
- **Conversation without Permission:** Continues a TCAP transaction. The destination node may *not* end the transaction.
- **Abort:** Terminates a transaction due to an abnormal situation.

The transaction portion also contains the **Originating Transaction ID** and **Responding Transaction ID** fields which associate the TCAP transaction with a specific application at the originating and destination signaling points respectively.

Component Portion

The component portion contains *components*. There are six kinds of components:

- **Invoke (Last):** Invokes an operation. For example, a Query with Permission transaction may include an Invoke (Last) component to request SCP translation of a dialed 800 number. The component is the "last" component in the query.
- **Invoke (Not Last):** Similar to the Invoke (Last) component except that the component is followed by one or more components.
- **Return Result (Last):** Returns the result of an invoked operation. The component is the "last" component in the response.
- **Return Result (Not Last):** Similar to the Return Result (Last) component except that the component is followed by one or more components.
- **Return Error:** Reports the unsuccessful completion of an invoked operation.
- **Reject:** Indicates that an incorrect package type or component was received.

Components include **parameters**, which contain application-specific data carried unexamined by TCAP.

Other SS7 Information

For **general information** about SS7, refer to [Bell Atlantic's Signaling System 7 tutorial](#) on the International Engineering Consortium's [Web ProForum](#) web site.

For **detailed information** about SS7, contact:

- [International Telecommunication Union](http://www.itu.int/) (ITU) - <http://www.itu.int/>
- [American National Standards Institute](http://www.ansi.org/) (ANSI) - <http://www.ansi.org/>
- [Telcordia Technologies](http://www.telcordia.com/) (formerly Bellcore) - <http://www.telcordia.com/>
- [European Telecommunications Standards Institute](http://www.etsi.org/) (ETSI) - <http://www.etsi.org/>

For a more extensive list, refer to our web site at <http://www.pt.com/telecomlinks.html>.

Bibliography

The following table lists several important SS7 standards documents which were used in the preparation of this tutorial:

SS7 Level	ITU Standard	ANSI Standard	JTC (Japan) Standard
MTP Level 2	ITU Q.701 - Q.703, 1992	ANSI T1.111.2-3, 1992	JT-Q.701 - JT-Q.703, 1992
MTP Level 3	ITU Q.704 - Q.707, 1992	ANSI T1.111.4-7, 1992	JT-Q.704 - JT-Q.707, 1992
SCCP	ITU Q.711 - Q.714, 1992	ANSI T1.112, 1992	JT-Q.711 - JT-Q.714, 1992
TUP	CCITT Q.721 - Q.724, 1988	N/A	N/A
ISUP	ITU Q.761 - Q.764, 1992	ANSI T1.113, 1992	JT-Q.761 - JT-Q.764, 1992
TCAP	ITU Q.771 - Q.775, 1992	ANSI T1.114, 1992	JT-Q.771 - JT-Q.775, 1992

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