ISUP consists of call processing, supplementary services, and maintenance functions. This chapter is divided into the following sections, which describe the specific components of ISUP:

- Bearers and Signaling
- ISUP and the SS7 Protocol Stack
- ISUP Message Flow
- Message Timers
- Circuit Identification Codes
- Enbloc and Overlap Address Signaling
- Circuit Glare
- Continuity Test
- ISUP Message Format
- Detailed Call Walk-Through
- Circuit Suspend and Resume
- ISUP and Local Number Portability
- ISUP–ISUP Tandem Calls
- Interworking with ISDN
- Supplementary Services
- Additional Call Processing Messages
- Maintenance Messages and Procedures

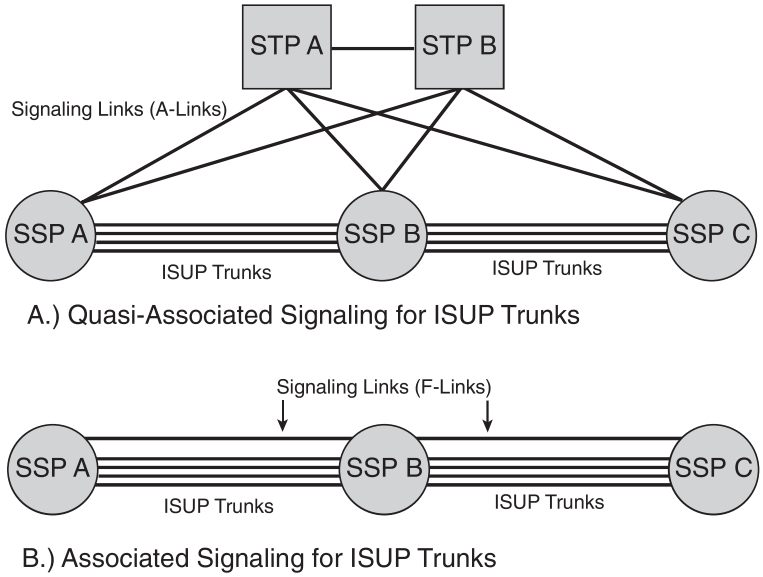
Bearers and Signaling

ISUP allows the call control signaling to be separated from the circuit that carries the voice stream over interoffice trunks. The circuit that carries the voice portion of the call is known within the telephone industry by many different terms. Voice channel, voice circuit, trunk member, and bearer all refer to the digital time slot that transports the voice (fax, modem, or other voiceband data) part of a call. The term "voice circuit" can be somewhat ambiguous in this context because sometimes it is used to refer to the trunk span that is divided into time slots, or to an individual time slot on a span.

The signaling component of the call is, of course, transported over SS7 signaling links. This creates two independent paths for call information between nodes: the voice path and the signaling path. The signaling mode describes the signaling relation between the two paths. Following is a brief review of the associated and quasi-associated signaling modes as they relate to ISUP, which we discussed in earlier chapters.

If the signaling travels on a single linkset that originates and terminates at the same nodes as the bearer circuit, the signaling mode is associated. If the signaling travels over two or more linksets and at least one intermediate node, the signaling mode is quasi-associated. In Figure 8-1, part A shows quasi-associated signaling between SSP A and SSP B and between SSP B and SSP C. In part B of Figure 8-1, the same SSP nodes are shown using associated signaling. Notice that the signaling links in part B terminate at the same point as the trunks. Also, the signaling link is shown as a separate entity in part B to illustrate the signaling mode; however, it is typically just another time slot that is dedicated for signaling on a trunk span.

Figure 8-1 *Signaling Mode Relating to ISUP Trunks*

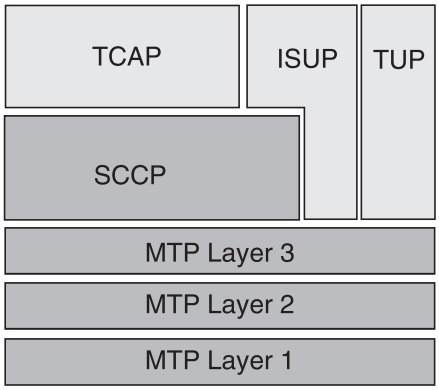


The signaling mode used for ISUP depends greatly on what SS7 network architecture is used. For example, North America uses hierarchical STPs for aggregation of signaling traffic. Therefore, most ISUP trunks are signaled using quasi-associated signaling. Using this mode, the signaling is routed through the STP before reaching the destination SSP. In contrast, while the U.K. uses quasi-associated signaling for some SSPs, they also heavily use associated signaling with directly connected signaling links between many SSPs.

ISUP and the SS7 Protocol Stack

As shown in Figure 8-2, ISUP resides at Level 4 of the SS7 stack with its predecessor, the Telephone User Part (TUP). TUP is still used in many countries, but ISUP is supplanting it over time. TUP also provides a call setup and release that is similar to ISUP, but it has only a subset of the capabilities. TUP is not used in North America because its capabilities are not sufficient to support the more complex network requirements.

Figure 8-2 *ISUP at Level 4 of the SS7 Stack*



As you can see in Figure 8-2, a connection exists between ISUP and both the SCCP and MTP3 levels. ISUP uses the MTP3 transport services to exchange network messages, such as those used for call setup and clear down. The connection to SCCP is for the transport of end-to-end signaling. While SCCP provides this capability, today ISUP end-to-end signaling is usually transported directly over MTP3. The "Interworking with ISDN" section of this chapter further discusses end-to-end signaling and the two different methods using MTP3 and SCCP for transport.

ISUP Standards and Variants

The ITU-T defines the international ISUP standards in the Q.767 and the national standards in the Q.761–Q.764 series of specifications. The ITU-T standards provide a basis from which countries or geographical regions can define regional or national versions of the protocol, which are often referred to as variants. For the U.S. network, the following standards provide the primary specifications for the ISUP protocol and its use in local and long distance networks:

- ANSI T1.113–ANSI ISUP
- Telcordia GR-246 Telcordia Technologies Specification of Signaling System No. 7, Volume 3. (ISUP)
- Telcordia GR-317 LSSGR—Switching System Generic Requirements for Call Control Using the Integrated Services Digital Network User Part (ISDNUP)
- Telcordia GR-394 LSSGR—Switching System Generic Requirements for Interexchange Carrier Interconnection (ICI) Using the Integrated Services Digital Network User Part (ISDNUP)

In Europe, the following ETSI standards provide the basis for the national ISUP variants:

- ETSI ETS 300-121 Integrated Services Digital Network (ISDN); Application of the ISDN User Part (ISUP) of CCITT Signaling System No. 7 for international ISDN interconnections
- ETSI ETS 300-156-x Integrated Services Digital Network (ISDN); Signaling System No. 7; ISDN User Part (ISUP) for the international interface

The ETS 300-121 is version 1, and the ETS 300-156-x (where x represents an individual document number) is a suite of specifications that covers ETSI ISUP versions 2–4.

A multitude of different country requirements have created many ISUP variants. A few of the several flavors are Swedish ISUP, U.K. ISUP, Japanese ISUP, Turkish ISUP, Korean ISUP. Each variant is tailored to the specific national requirements. Although not certain of the exact number of variants that are in existence today, the author has encountered over a hundred different ISUP variants while developing software for switching platforms.

ISUP Message Flow

This section provides an introduction to the core set of ISUP messages that are used to set up and release a call. The ISUP protocol defines a large set of procedures and messages, many of which are used for supplementary services and maintenance procedures. While the ITU Q.763 ISUP standard defines nearly fifty messages, a core set of five to six messages represent the majority of the ISUP traffic on most SS7 networks. The basic message flow that is presented here provides a foundation for the remainder of the chapter. Additional messages, message content, and the actions taken at an exchange during message processing build upon the foundation presented here.

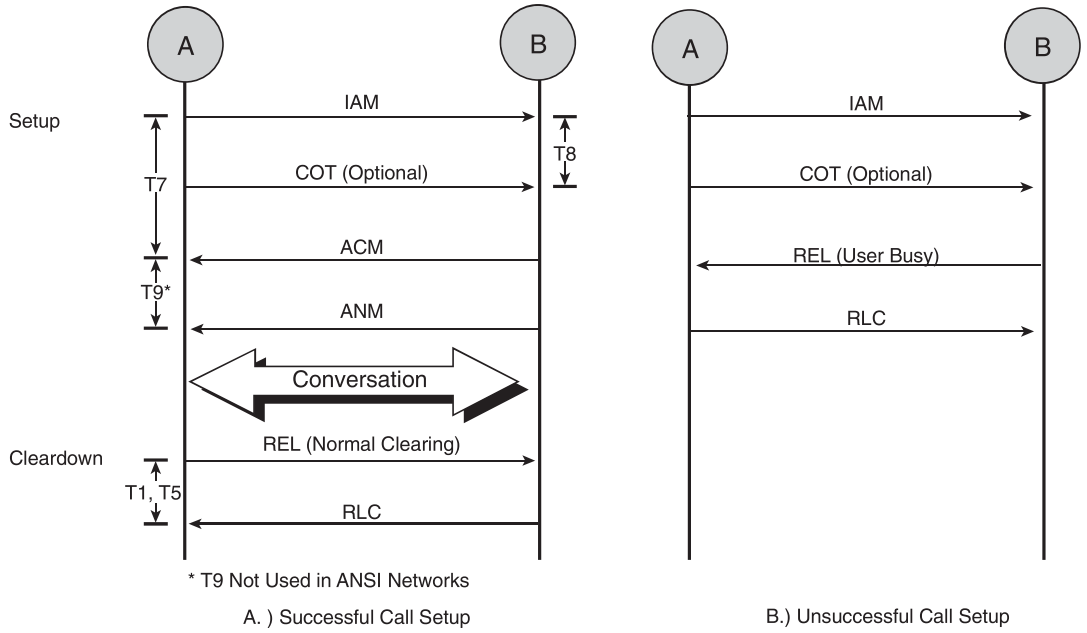
A basic call can be divided into three distinct phases:

- Setup
- Conversation (or data exchange for voice-band data calls)
- Release

ISUP is primarily involved in the set-up and release phases. Further ISUP signaling can take place if a supplementary service is invoked during the conversation phase.

In Figure 8-3, part A illustrates the ISUP message flow for a basic call. The call is considered basic because no supplementary services or protocol interworking are involved. The next section, “Call Setup,” explains the figure’s message timer values.

Figure 8-3 *Simple ISUP Message Flow*



Call Setup

A simple basic telephone service call can be established and released using only five ISUP messages. In Figure 8-3, part A shows a call between SSP A and SSP B. The Initial Address Message (IAM) is the first message sent, which indicates an attempt to set up a call for a particular circuit. The IAM contains information that is necessary to establish the call connection—such as the call type, called party number, and information about the bearer circuit. When SSP B receives the IAM, it responds with an Address Complete Message (ACM). The ACM indicates that the call to the selected destination can be completed. For example, if the destination is a subtending line, the line has been determined to be in service and not busy. The Continuity message (COT), shown in the figure, is an optional message that is used for continuity testing of the voice path before it is cut through to the end users. This chapter's "Continuity Test" section discusses the COT message.

Once the ACM has been sent, ringing is applied to the terminator and ring back is sent to the originator. When the terminating set goes off-hook, an Answer Message (ANM) is sent to the originator. The call is now active and in the talking state. For an ordinary call that does not involve special services, no additional ISUP messages are exchanged until one of the parties signals the end of the call by going on-hook.

Call Release

In Figure 8-3, the call originator at SSP A goes on-hook to end the call. SSP A sends a Release message (REL) to SSP B. The REL message signals the far end to release the

bearer channel. SSP B responds with a Release Complete message (RLC) to acknowledge the REL message. The RLC indicates that the circuit has been released.

If the terminating party goes on-hook first, the call might be suspended instead of being released. Suspending a call maintains the bearer connection for a period of time, even though the terminator has disconnected. The terminator can go off-hook to resume the call, providing that he does so before the expiration of the disconnect timer or a disconnect by the originating party. This chapter discusses suspending and resuming a connection in more detail in the section titled "Circuit Suspend and Resume."

NOTE Several different terms are used to identify the two parties who are involved in a telephone conversation. For example, the originating party is also known as the calling party, or the "A" party. The terminating party, or "B" party, are also synonymous with the called party.

Unsuccessful Call Attempt

In Figure 8-3, part B shows an unsuccessful call attempt between SSP A and SSP B. After receiving the IAM, SSP B checks the status of the destination line and discovers that it is busy. Instead of an ACM, a REL message with a cause value of User Busy is sent to SSP A, indicating that the call cannot be set up. While this example shows a User Busy condition, there are many reasons that a call set-up attempt might be unsuccessful. For example, call screening at the terminating exchange might reject the call and therefore prevent it from being set up. Such a rejection would result in a REL with a cause code of Call Rejected.

NOTE Call screening compares the called or calling party number against a defined list of numbers to determine whether a call can be set up to its destination.

Message Timers

Like other SS7 protocol levels, ISUP uses timers as a safeguard to ensure that anticipated events occur when they should. All of the timers are associated with ISUP messages and are generally set when a message is sent or received to ensure that the next intended action occurs. For example, when a REL message is sent, Timer T1 is set to ensure that a RLC is received within the T1 time period.

ITU Q.764 defines the ISUP timers and their value ranges. In Figure 8-3, part A includes the timers for the messages that are presented for a basic call. The "Continuity Test" section

of this chapter discusses the timers associated with the optional COT message. Following are the definitions of each of the timers in the figure:

- **T7 awaiting address complete timer**—Also known as the network protection timer. T7 is started when an IAM is sent, and is canceled when an ACM is received. If T7 expires, the circuit is released.
- **T8 awaiting continuity timer**—Started when an IAM is received with the Continuity Indicator bit set. The timer is stopped when the Continuity Message is received. If T8 expires, a REL is sent to the originating node.
- **T9 awaiting answer timer**—Not used in ANSI networks. T9 is started when an ACM is received, and is canceled when an ANM is received. If T9 expires, the circuit is released. Although T9 is not specified for ANSI networks, answer timing is usually performed at the originating exchange to prevent circuits from being tied up for an excessive period of time when the destination does not answer.
- **T1 release complete timer**—T1 is started when a REL is sent and canceled when a RLC is received. If T1 expires, REL is retransmitted.
- **T5 initial release complete timer**— T5 is also started when a REL is sent, and is canceled when a RLC is received. T5 is a longer duration timer than T1 and is intended to provide a mechanism to recover a nonresponding circuit for which a release has been initiated. If T5 expires, a RSC is sent and REL is no longer sent for the nonresponding circuit. An indication of the problem is also given to the maintenance system.

We list the timers for the basic call in part A of Figure 8-3 to provide an understanding of how ISUP timers are used. There are several other ISUP timers; a complete list can be found in Appendix H, "ISUP Timers for ANSI/ETSI/ITU-T Applications."

Circuit Identification Codes

One of the effects of moving call signaling from CAS to Common Channel Signaling (CCS) is that the signaling and voice are now traveling on two separate paths through the network. Before the introduction of SS7 signaling, the signaling and voice component of a call were always transported on the same physical facility. In the case of robbed-bit signaling, they are even transported on the same digital time slot of that facility.

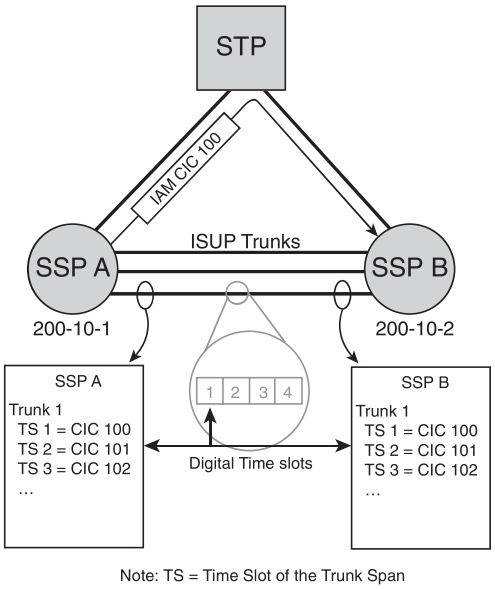
The separation of signaling and voice create the need for a means of associating the two entities. ISUP uses a Circuit Identification Code (CIC) to identify each voice circuit. For example, each of the 24 channels of a T1 span (or 30 channels of an E1 span) has a CIC associated with it. When ISUP messages are sent between nodes, they always include the CIC to which they pertain. Otherwise, the receiving end would have no way to determine the circuit to which the incoming message should be applied. Because the CIC identifies a bearer circuit between two nodes, the node at each end of the trunk must define the same CIC for the same physical voice channel.

TIP Not defining CICs so that they match properly at each end of the connection is a common cause of problems that occur when defining and bringing new ISUP trunks into service.

ITU defines a 12-bit CIC, allowing up to 4096 circuits to be defined. ANSI uses a larger CIC value of 14 bits, allowing for up to 16,384 circuits.

Figure 8-4 shows an ISUP message from SSP A that is routed through the STP to SSP B. For simplicity, only one STP is shown. In the message, CIC 100 identifies the physical circuit between SSP A and B to which the message applies. Administrative provisioning at each of the nodes associates each time slot of the digital trunk span with a CIC. As shown in the figure, Trunk 1, time slot (TS) 1 is defined at each SSP as CIC 100. Trunk 1, time slot 2 is defined as CIC 101, and so on.

Figure 8-4 *CIC Identifies the Specific Voice Circuit*



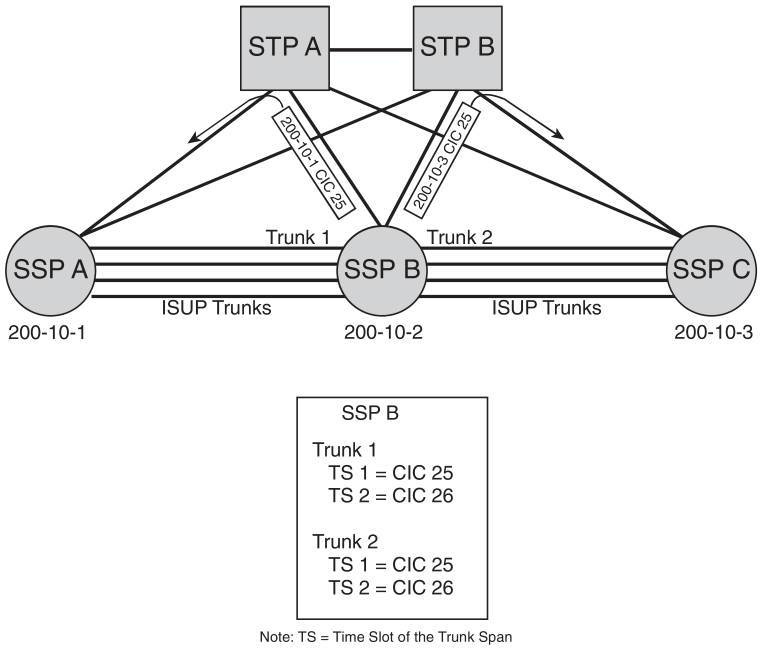
DPC to CIC Association

Since each ISUP message is ultimately transported by MTP, an association must be created between the circuit and the SS7 network destination. This association is created through provisioning at the SSP, by linking a trunk group to a routeset or DPC.

The CIC must be unique to each DPC that the SSP defines. A CIC can be used again within the same SSP, as long as it is not duplicated for the same DPC. This means that you might see CIC 0 used many times throughout an SS7 network, and even multiple times at the same

SSP. It is the combination of DPC and CIC that uniquely identifies the circuit. Figure 8-5 shows an example of three SSPs that are interconnected by ISUP trunks. SSP B uses the same CIC numbers for identifying trunks to SSP A and SSP C. For example, notice that it has two trunks using CIC 25 and two trunks using CIC 26. Since SSP A and SSP C are separate destinations, each with their own unique routeset defined at SSP B, the DPC/CIC combination still uniquely identifies each circuit. SSP B can, in fact, have many other duplicate CIC numbers associated with different DPCs.

Figure 8-5 *Combination of DPC/CIC Provide Unique Circuit ID*



Unidentified Circuit Codes

When a message is received with a CIC that is not defined at the receiving node, an Unequipped Circuit Code (UCIC) message is sent in response. The UCIC message's CIC field contains the unidentified code. The UCIC message is used only in national networks.

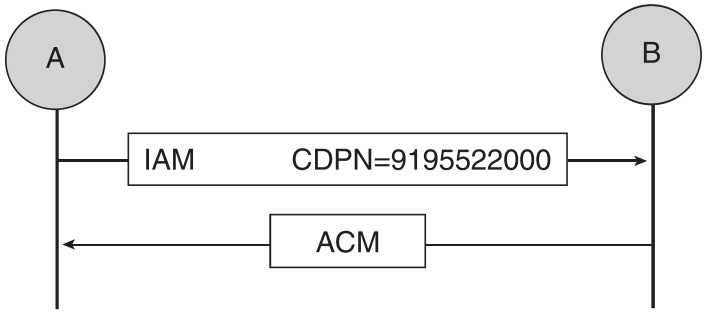
Enbloc and Overlap Address Signaling

The Called Party Number (CdPN) is the primary key for routing a call through the network. When using ISUP to set up a call, the CdPN can be sent using either enbloc or overlap signaling. In North America, enbloc signaling is always used; in Europe, overlap signaling is quite common, although both methods are used.

Enbloc Signaling

The enbloc signaling method transmits the number as a complete entity in a single message. When using enbloc signaling, the complete number is sent in the IAM to set up a call. This is much more efficient than overlap signaling, which uses multiple messages to transport the number. Enbloc signaling is better suited for use where fixed-length dialing plans are used, such as in North America. Figure 8-6 illustrates the use of enbloc signaling.

Figure 8-6 *Enbloc Address Signaling*



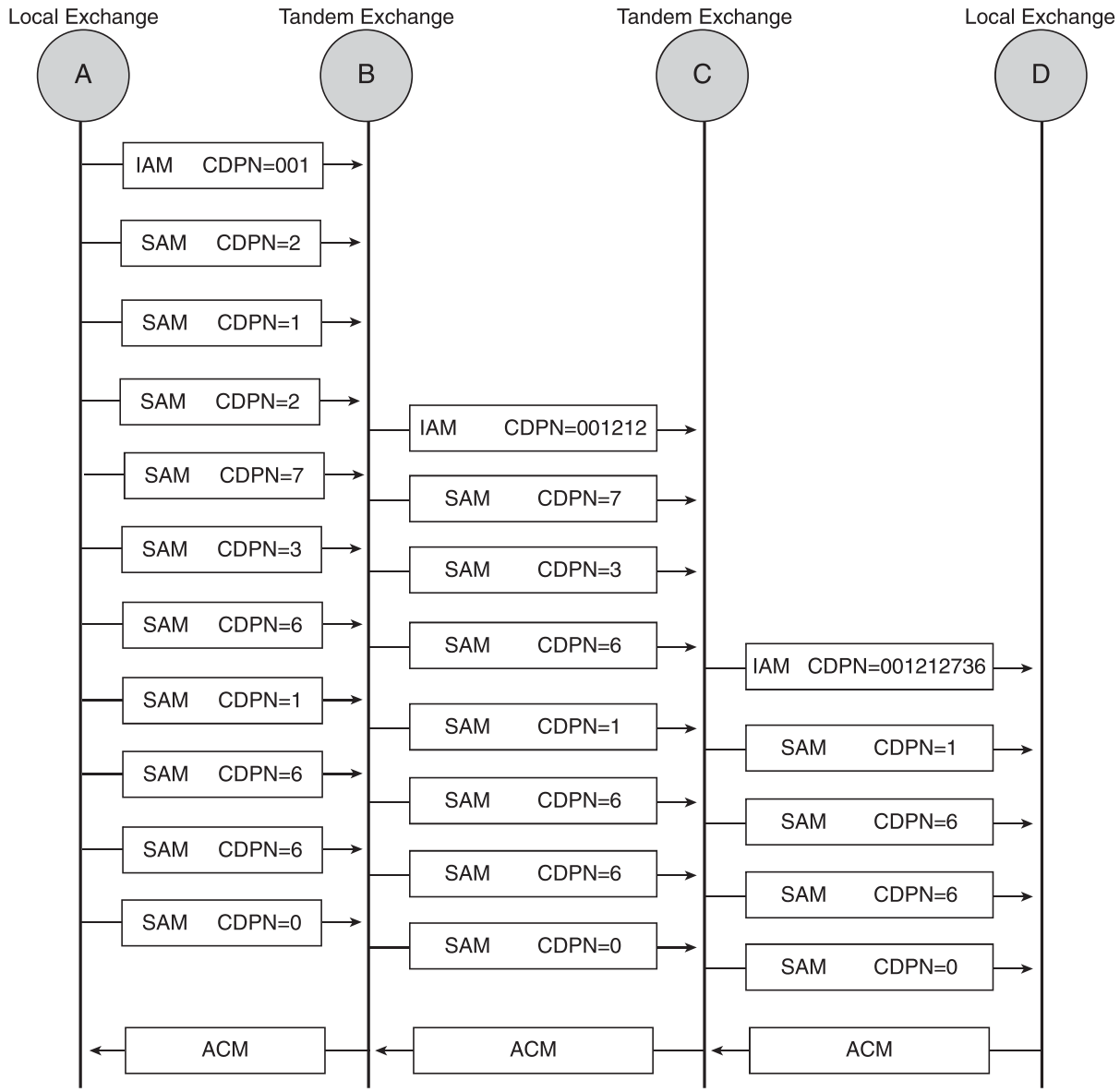
Overlap Signaling

Overlap signaling sends portions of the number in separate messages as digits are collected from the originator. Using overlap signaling, call setup can begin before all the digits have been collected. When using the overlap method, the IAM contains the first set of digits. The Subsequent Address Message (SAM) is used to transport the remaining digits. Figure 8-7 illustrates the use of overlap signaling. Local exchange A collects digits from the user as they are dialed. When enough digits have been collected to identify the next exchange, an IAM is sent to exchange B. When tandem exchange B has collected enough digits to identify the next exchange, it sends an IAM to exchange C; exchange C repeats this process. After the IAM is sent from exchange C to exchange D, the destination exchange is fully resolved. Exchange D receives SAMs containing the remaining digits needed to identify the individual subscriber line.

When using dialing plans that have variable length numbers, overlap signaling is preferable because it decreases post-dial delay. As shown in the preceding example, each succeeding call leg is set up as soon as enough digits have been collected to identify the next exchange.

As discussed in Chapter 5, "The Public Switched Telephone Network (PSTN)," interdigit timing is performed as digits are collected from a subscriber line. When an exchange uses variable length dial plans with enbloc signaling, it must allow interdigit timing to expire before attempting to set up the call. The exchange cannot start routing after a specific number of digits have been collected because that number is variable. By using overlap signaling, the call is set up as far as possible, waiting only for the final digits the subscriber dials. Although overlap signaling is less efficient in terms of signaling bandwidth, in this situation it is more efficient in terms of call set-up time.

Figure 8-7 *Overlap Address Signaling*



Circuit Glare (Dual-Seizure)

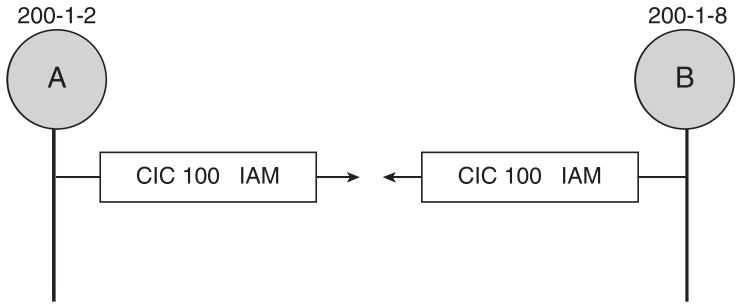
Circuit glare (also known as dual-seizure) occurs when the node at each end of a two-way trunk attempts to set up a call over the same bearer at the same time. Using ISUP signaling, this occurs when an IAM for the same CIC is simultaneously sent from each end. Each end sends an IAM to set up a call before it receives the IAM from the other end. You will recall from our discussion of the basic ISUP message flow that once an IAM is sent, an ACM is expected. When an IAM is received after sending an IAM for the same CIC, glare has occurred.

Resolving Glare

When glare is detected, one node must back down and give control to the other end. This allows one call to complete, while the other call must be reattempted on another CIC. There are different methods for resolving which end takes control. For normal 64-kb/s connections, two methods are commonly used. With the first method, the point code and CIC numbers are used to determine which end takes control of the circuit. The node with the higher-numbered point code takes control of even number CICs, and the node with the lower-numbered point code takes control of odd numbered CICs. This provides a fair mechanism that allows each node to control approximately half of the calls encountering glare. In the United States, an example of this use would be two peer End Office exchanges. The second method of glare resolution is handled by prior agreement between the two nodes about which end will back down when glare occurs. One node is provisioned to always back down, while the other node is provisioned to take control. A typical example of this arrangement in the U.S. network would be a hand-off between non-peer exchanges, such as an IXC to AT. The method to use for glare resolution can usually be provisioned at the SSP, typically at the granularity level of the trunk group.

Figure 8-8 illustrates a glare condition when SSP A and B have both sent an IAM before receiving the IAM from the other end. Assuming that the point code/CIC method of resolving glare is being used, SSP B takes control of the circuit because the CIC is even numbered and SSP B has a numerically higher point code.

Figure 8-8 *Glare Condition During Call Setup*



Avoiding Glare

When provisioning trunks, glare conditions can be minimized by properly coordinating the trunk selection algorithms at each end of a trunk group. A common method is to perform trunk selection in ascending order of the trunk member number at one end of the trunk group, and in descending order at the other end. This minimizes contention to the point of selecting the last available resource between the two ends. Another method is to have one end use the "Most Idle" trunk selection while the other end uses the "Least Idle" selection. The idea is to have an SSP select a trunk that is least likely to be selected by the SSP at the other end of the trunk group.

Continuity Test

Continuity testing verifies the physical bearer facility between two SSPs. When CAS signaling is used, a call setup fails if the voice path is faulty. Using ISUP signaling, it is possible to set up a call using the signaling network without knowing that the bearer connection is impaired or completely broken.

The voice and signaling channels are usually on separate physical facilities, so a means of verifying that the voice facility is connected properly between the SSPs is needed. Many digital voice transmission systems provide fault detection on bearer facilities, which are signaled to the connected switching system using alarm indication bits within the digital information frame. However, these bits are not guaranteed to be signaled transparently through interconnecting transmission equipment, such as a Digital Access Cross Connect system (DACS) or digital multiplexers. Some networks require these alarm indications to be passed through without disruption, therefore, reducing the need for continuity testing.

Continuity testing can be considered part of the ISUP maintenance functions. It can be invoked to test trunks manually, as part of routine maintenance and troubleshooting procedures. Continuity testing can also be provisioned to take place during normal call setup and it has an impact on the flow of call processing. During call processing, the originating exchange determines whether a continuity test should be performed. Network guidelines vary concerning whether and how often continuity testing is performed. The determination is typically based on a percentage of call originations. For example, in the United States, the generally accepted practice is to perform continuity testing on 12 percent of ISUP call originations (approximately one out of eight calls). This percentage is based on Telcordia recommendations.

Loopback and Transceiver Methods

The actual circuit testing can be performed using either the loopback or the transceiver method. The loopback method is performed on four-wire circuits using a single tone, and the transceiver method is used for two-wire circuits using two different tones. The primary difference between the two methods is related to the action that takes place at the terminating end. When using either method, a tone generator is connected to the outgoing circuit at the originating exchange. Using the loopback method, the terminating exchange connects the transmit path to the receive path, forming a loopback to the originator. The originator measures the tone coming back to ensure that it is within the specified parameters. When the transceiver method is used, the transmit and receive path are connected to a tone transceiver that measures the tone coming from the originating exchange and sends a different tone back to the originating exchange. The tone frequencies vary between countries. The following tones are used for the continuity test in North America:

- 2010 Hz from the originating exchange
- 1720 Hz from the terminating exchange (transceiver method only)

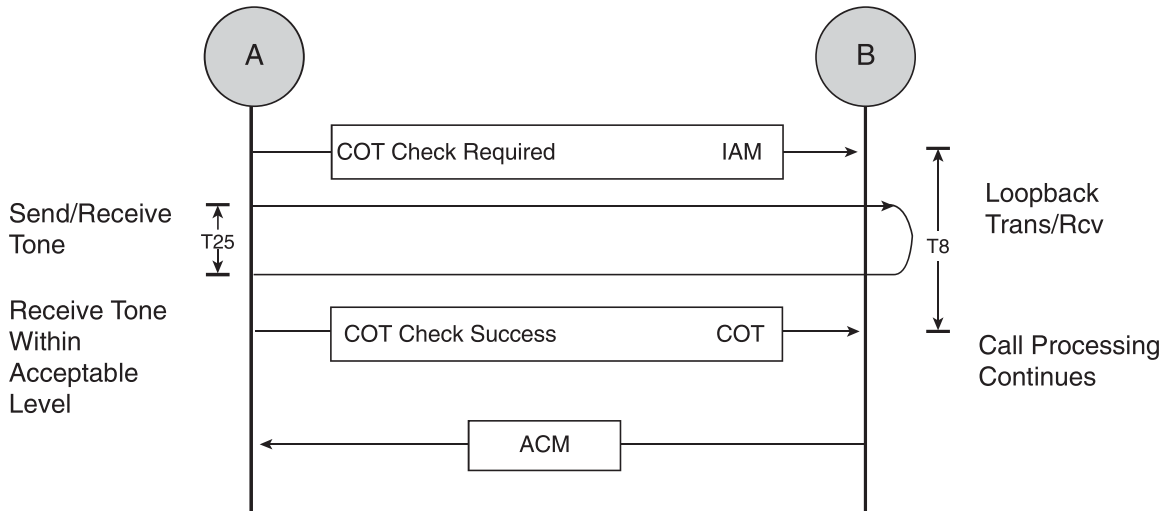
Another example of the COT tone frequency is 2000 Hz, which is used in the U.K.

Continuity Check Procedure

The Initial Address Message contains a *Continuity Check Indicator* as part of the *Nature of Connection* field. When an ISUP trunk circuit is selected for an outgoing call and the exchange determines that a continuity check should be performed, the Continuity Check Indicator is set to true. A tone generator is connected to the outgoing circuit, and the IAM is sent to the SSP at the far end of the trunk. Timer T25 is started when the tone is applied, to ensure that tone is received back within the T25 time period. When the SSP at the far end receives the IAM with the Continuity Check Indicator set to true, it determines whether to create a loopback of the transmit and receive path, or to connect a transceiver. The transceiver receives the incoming tone and generates another tone on the outgoing circuit. The determination of whether to use a loopback or transceiver is typically based on provisioned data at the receiving exchange. Upon receipt of the IAM, Timer T8 is started at the terminating exchange, awaiting the receipt of a COT message to indicate that the test passed. The terminating exchange does not apply ringing to the called party or send back ACM until the COT message has been received with a continuity indicator of continuity check successful to indicate that the bearer connection is good.

The originating exchange measures the received tone to ensure that it is within an acceptable frequency range and decibel level. Next it sends a COT message to the terminating exchange to indicate the test results. If the test passes, the call proceeds as normal; if the test fails, the CIC is blocked, the circuit connection is cleared, and the originating exchange sends a Continuity Check Request (CCR) message to request a retest of the failed circuit. While ISUP maintenance monitors the failed circuit's retest, ISUP call processing sets the call up on another circuit. Figure 8-9 shows a successful COT check using the loopback method.

Figure 8-9 *Successful COT Check Using the Loopback Method*



ISUP Message Format

The User Data portion of the MTP3 Signaling Information Field contains the ISUP message, identified by a Service Indicator of 5 in the MTP3 SIO field. Each ISUP message follows a standard format that includes the following information:

- **CIC**—The Circuit Identification Code for the circuit to which the message is related.
- **Message Type**—The ISUP Message Type for the message (for example, an IAM, ACM, and so on).
- **Mandatory Fixed Part**—Required message parameters that are of fixed length.
- **Mandatory Variable Part**—Required message parameters that are of variable length. Each variable parameter has the following form:
  - Length of Parameter
  - Parameter Contents

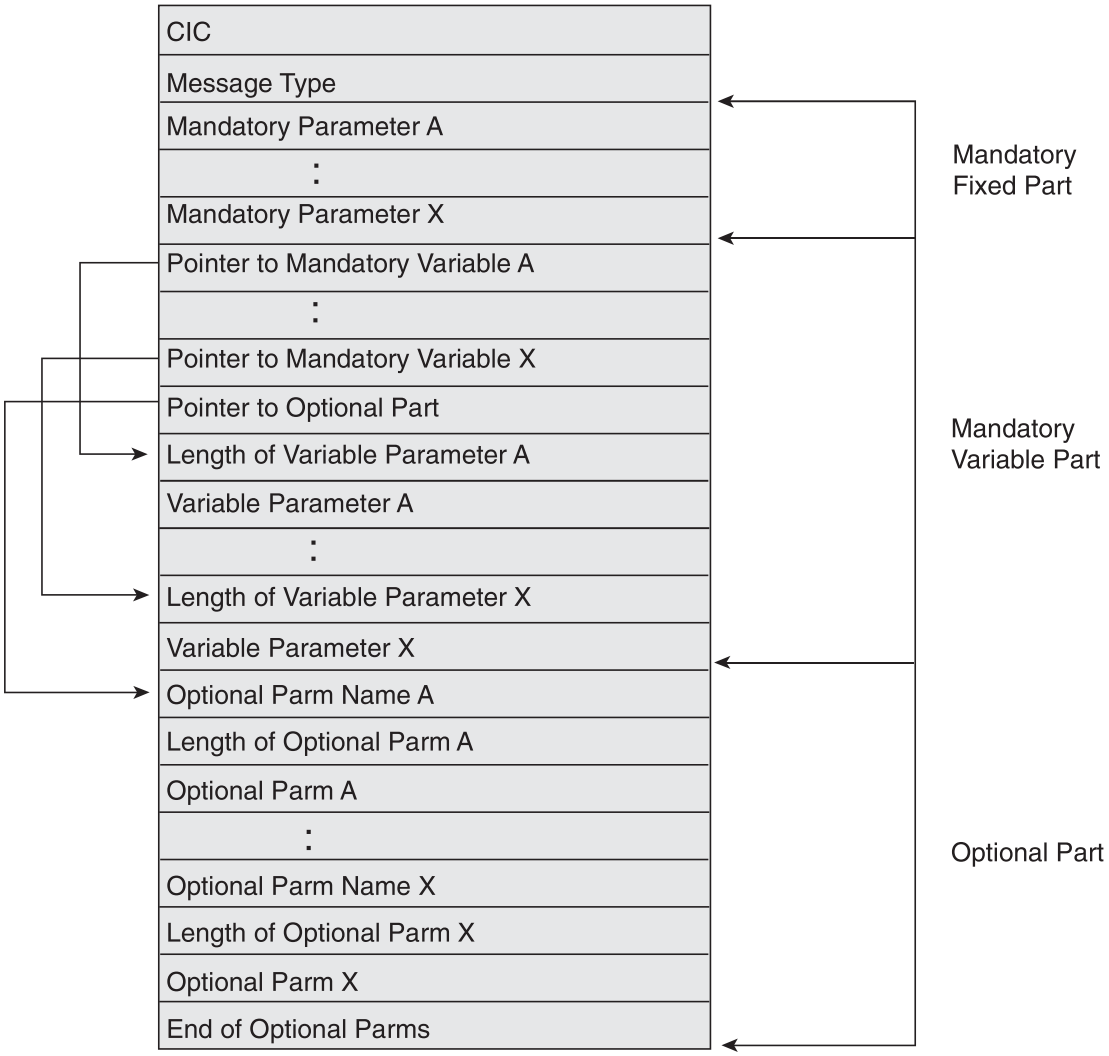
Because the parameter is not a fixed length, a field is included to specify the actual length.

- **Optional Part**—Optional fields that can be included in the message, but are not mandatory. Each optional parameter has the following form:
  - Parameter Name
  - Length of Parameter
  - Parameter Contents

Figure 8-10 shows the ISUP message structure, as described here. This message structure provides a great deal of flexibility for constructing new messages. Each message type defines the mandatory parameters that are necessary for constructing a message. The mandatory fixed variables do not contain length information because the ISUP standards specify them to be a fixed length. Because the mandatory variable parameters are of variable lengths, pointers immediately follow the mandatory fixed part to point to the beginning of each variable parameter. The pointer value is simply the number of octets from the pointer field to the variable parameter length field.

In addition to the mandatory fields, each message can include optional fields. The last of the pointer fields is a pointer to the optional part. Optional fields allow information to be included or omitted as needed on a per-message basis. The optional fields differ based on variables such as the call type or the supplementary services involved. For example, the Calling Party Number (CgPN) field is an optional parameter of the IAM, but is usually included to provide such services as Caller ID and Call Screening.

Figure 8-10 *ISUP Message Format*



A single message can include many optional parameters. The optional part pointer field only points to the first parameter. Because the message might or might not include the parameters, and because the parameters can appear in any order, the first octet includes the name of each parameter in order to identify it. The parameter length follows the name to indicate how many octets the parameter contents include. When the parameter name is coded as zero, it signals the end of the optional parameters. During parsing of an incoming ISUP message, optional parameters are processed until the *end of optional parameters* marker is reached. If the message does not have any optional parameters, the pointer to the optional part is coded to zero.

Basic Call Message Formats

Here, we examine the six messages shown in the basic call setup because they comprise the core message set for basic call setup and release, and are therefore used frequently. There are slight variations in the messages used based on the individual network. For example, Europe uses the SAM frequently and the COT message more rarely. In North America, SAM is not used at all, but COT is used more often. This section considers the following messages:

- Initial Address Message (IAM)
- Subsequent Address Message (SAM–ITU Networks only)
- Continuity Message (COT)
- Address Complete Message (ACM)
- Answer Message (ANM)
- Release Message (REL)
- Release Complete Message (RLC)

The following sections show only the mandatory fields for each message. Keep in mind that many optional parameters can also be included. In each of the figures, the fixed mandatory fields with sub-fields have been expanded to show what they are. For the sake of brevity in the figures, the variable subfields have not been expanded. All of the ISUP Message formats and parameters are documented in ITU-T Q.763. ANSI T1.113 documents the North American version of the messages.

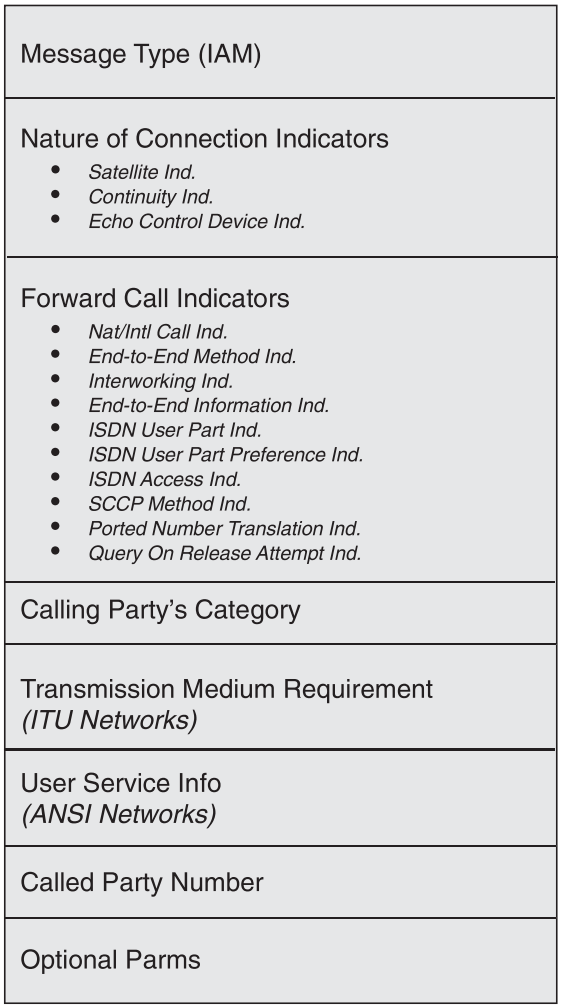
Initial Address Message (IAM)

The IAM contains the information needed to set up a call. For a basic call, it is the first message sent and is typically the largest message in terms of size. Figure 8-11 shows the mandatory fields that the message includes. In addition to the mandatory fields, the ITU-T Q.764 lists more than 50 optional parameters that can be included in the IAM. The mandatory parameters for ITU and ANSI are the same, with the exception of the Transmission Medium Requirements parameter. In ANSI networks, the User Service Info field is used instead.

As shown in Figure 8-11, the *Nature of Connection Indicators (NOC)* pass information about the bearer circuit connection to the receiving node. The indicators consist of the following subfields:

- **Satellite Indicator**—Specifies whether one or more satellites have been used for the circuit connection that is being set up. This information is useful when setting up calls to prevent an excessive number of satellite hops, which can reduce the quality of calls.
- **Continuity Indicator**—Designates whether to perform a continuity check on the circuit being set up.
- **Echo Control Device Indicator**—Specifies whether echo suppression is used on the circuit. Echo suppression is used to increase the quality of voice calls by reducing echo, but it can damage data and fax calls because it subtracts a portion of the voice-band signal.

Figure 8-11 *IAM Message Format*



The *Forward Call Indicators (FCI)* contain information that specifies both the preferences about call setup in the forward direction and the conditions encountered so far in setting up the call. They include the following subfields:

- **National/International Call Indicator**—Indicates whether the call is coming in as National or International. International calls are specified by ITU international procedures, and national calls are processed according to national ISUP variant standards.
- **End-to-End Method Indicator**—Indicates the method used for signaling end-to-end information. SCCP and pass-along are the two end-to-end methods that are used. The pass-along method traverses each node in the connection to deliver information to the correct node. The SCCP method uses connectionless signaling to send information directly to its destination.

- **Interworking Indicator**—Indicates whether the connection has encountered interworking with non-SS7 facilities (for example, MF trunks). Interworking with non-SS7 facilities can limit or prohibit the capability of supplementary services or certain call types that require SS7 signaling.
- **End-to-End Information Indicator**—Indicates whether any end-to-end information is available.
- **ISDN User Part Indicator**—Indicates whether ISUP has been used for every leg of the connection. Note that this is not the same as the Interworking Indicator. It is possible to have an SS7-signaled circuit, but not use ISUP (for example, TUP signaling); however, if interworking has been encountered, this indicator is set to *ISDN User Part not used all the way.*
- **ISDN User Part Preference Indicator**—Specifies whether an ISUP facility is required or preferred when choosing an outgoing circuit. Some supplementary services or call types are not possible over non-ISUP facilities. If ISUP is required but not available, the call is released because the requested facility's preference cannot be met. If the preference indicator is set to *preferred*, an ISUP facility is chosen, if available; however, the call is still set up as long as a facility is available, even if it is not ISUP.
- **ISDN Access Indicator**—Indicates whether the originating access is ISDN or non-ISDN. ISDN provides a much richer interface to services that is not available on plain analog lines. This indicator suggests that the ISDN interface is available so that end-to-end signaling, backward requests for information, and so on can be carried out.
- **SCCP Method Indicator**—Indicates which method, if any, is used for SCCP end-to-end signaling. SCCP might use connection-oriented, connectionless, or both.

The *Calling Party's Category* specifies a general category into which the calling party is classified—such as an ordinary calling subscriber, operator, payphone, or test call.

The *Transmission Medium Requirement* (*TMR*) is not applicable to ANSI networks and is only supported in ITU-T networks. It contains the requirements for the bearer circuit capabilities (speech, 3.1-kHz audio, 64-Kb unrestricted, and so forth) that are needed for the call being set up. For example, a video conference might require a 384-Kbs unrestricted circuit to guarantee an acceptable level of video quality.

User Service Information (*USI*) is used in ANSI networks instead of the ITU-T specified TMR. It contains the requirements for the bearer circuit capabilities (speech, 3.1-kHz audio, and 64-Kbs unrestricted) along with additional information such as layer 1 codec, circuit, or packet transfer mode and other bearer-related specifics.

The *Called Party Number (CdPN)* is the destination number that the calling party dials. The CdPN contains the following fields:

- **Odd/Even Indicator**—Indicates an odd or even number of digits in the CdPN.
- **Nature of Address Indicator**—Indicates the type of number (for example, National Significant Number or International). The receiving switch uses this indicator during translations to apply the number's proper dial plan.

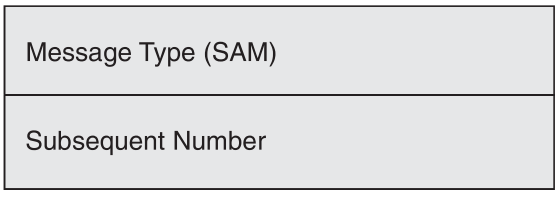
The *Internal Network Number Indicator* (*INN*), which is not used for ANSI, specifies whether routing to an internal network number is permitted. This field is used to block routing to specific numbers that should not be directly accessible from outside of the network. For example, if a premium rate number is translated to an internal number, the subscriber is blocked from dialing the internal number to ensure that the appropriate premium rate charges are collected.

- **Numbering Plan Indicator**—Specifies the type of number plan used. The E.164 ISDN numbering plan is commonly used for voice calls.
- **Address Signals**—The actual digits that comprise the called number. This includes digits 0–9 and the overdecadic digits (A–F), however, the overdecadic digits are not supported in all networks. Each digit is coded as a four-bit field.

Subsequent Address Message (SAM–ITU Networks Only)

Shown in Figure 8-12, the SAM is used to send subsequent address signals (digits) when using overlap signaling for call setup. It has one mandatory variable parameter: the *subsequent number*. One or more SAMs can be sent after an IAM to carry subsequent digits for call setup that are part of a destination's complete telephony number.

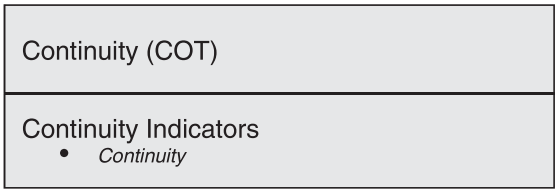
Figure 8-12 *SAM Message Format*



Continuity Message (COT)

As shown in Figure 8-13, the COT message contains the results of a continuity test. It has only one field: the *Continuity Indicators*. This field uses a single bit to indicate whether a continuity test passed or failed. The test's originator sends the message to the far end of the circuit that is being tested.

Figure 8-13 *COT Message Format*



Address Complete Message (ACM)

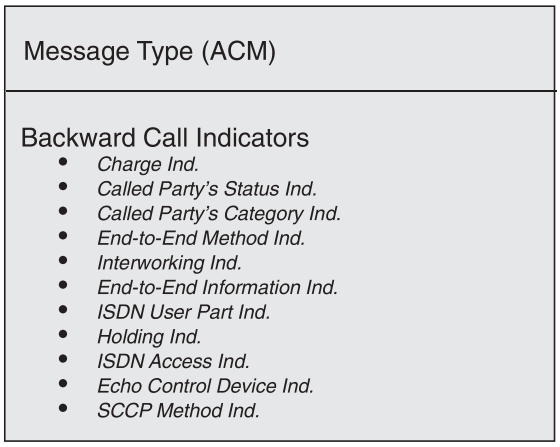
As shown in Figure 8-14, a destination node sends the ACM to indicate that a complete CdPN has been received. When enbloc signaling is used to set up the call, the ACM is sent after receiving the IAM; when overlap signaling is used, it is sent after the last SAM is received. In addition to indicating the successful reception of the CdPN, the ACM sends Backward Call Indicators (BCI*)* to signal information about the call setup. It is not mandatory for an ACM to be sent when setting up a call. It is permissible to send an ANM after receiving an IAM; this is sometimes referred to as "fast answer."

Many of the fields contained in the Backward Call Indicators are the same as those in the Forward Call Indicators (FCI), which are contained in the IAM. While the FCI signals the call indicators in the forward direction to provide information on the call setup to the terminating access (and intermediate nodes), the BCI signals similar information in the backward direction to the originator.

Here we discuss only the fields that are unique to the BCI. The remaining fields are the same as those we discussed for the FCI, except that they are representative of the call from the terminating end. For example, the ISDN Access Indicator specifies whether the "terminator" is ISDN.

- **Charge Indicator**—Indicates whether a call should be charged as determined by the charging exchange.
- **Called Party's Status Indicator**—Indicates whether the subscriber is free.
- **Called Party's Category Indicator**—Indicates the general category of the called party, an ordinary subscriber, or payphone.
- **Holding Indicator**—Indicates whether holding is required. This indicator can be used for special services, such as Operator Signaling Services or Malicious Call Tracing, to indicate that the incoming connection should be held. No specification for ANSI networks exists.

Figure 8-14 *ACM Message Format*



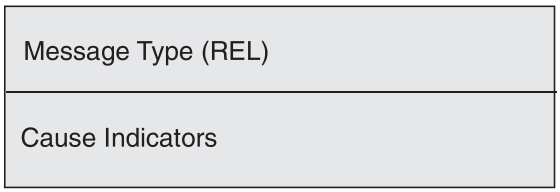
Answer Message (ANM)

The ANM is sent to the previous exchange when the called party answers (off-hook). Although it might contain many optional parameters, the ANM does not contain any mandatory fields other than the message type.

Release Message (REL)

As shown in Figure 8-15, the REL message indicates that the circuit is being released. When a RLC has been received in response, the circuit can be returned to the idle state for reuse. The REL message can be sent in either direction. It contains a single mandatory *Cause Indicators* field to indicate why the circuit is being released.

Figure 8-15 *REL Message Format*



Cause Indicators specify the cause information associated with the circuit being released. The Cause Indicators contain the general location in the network (such as local, remote, or transit) in which the circuit was released. The Coding Standard indicates which standard is used for decoding the Cause Value (such as ANSI, ITU). ANSI and ITU define some cause values differently, and ANSI also has additional values the ITU does not include.

The *Cause Value* contains an integer that represents the reason the circuit is being released. This value can be further decomposed into a class and a value. The most significant three bits of the Cause Value field represent the class. Each class is a general category of causes; for example, binary values of 000 and 001 are *normal event* class, and a value of 010 is *resource unavailable*. So, a cause value of 1 (unallocated number) is in the *normal event* class and a cause value of 34 (no circuit available) is in the *resource unavailable* class. Appendix M, "Cause Values," contains a complete list of the ITU and ANSI cause values.

The *Diagnostics* field is only applicable to certain cause values. It provides further information pertaining to the circuit release (for example, Transit Network Identity, Called Party Number [CdPN]) for those cause values.

Release Complete Message (RLC)

The RLC message is sent to acknowledge a REL message. Upon receipt of an RLC, a circuit can return to the idle state.

Detailed Call Walk-Through

Earlier in this chapter, we presented an ISUP message flow in order to illustrate the exchange of messages to establish and release an ISUP call. Now that we have discussed more of the ISUP details, we will build on that illustration. This section provides more detail about the call processing that was driven by the ISUP message events used in the earlier example. Although this chapter's primary focus is the ISUP protocol, it is important to understand how ISUP is applied in its normal domain of trunk call processing.

Call Setup

Refer back to Figure 8-3, where a call originates from a line at SSP A and terminates to a line at SSP B over an interexchange ISUP trunk. When call processing has completed translations of the called number at SSP A, the translations' results indicates that the call requires routing to an interexchange trunk group. The provisioned signaling type for the selected trunk group determines whether ISUP signaling or some other signaling, such as Multifrequency (MF), is used. When the signaling type is determined to be ISUP, the trunk circuit to be used for the outgoing call is reserved for use.

The SSP populates the IAM with information about the call setup, such as the CIC, CdPN, Call Type, CgPN, and PCM Encoding scheme. The IAM information is placed in the User Data field of the MTP3 SIF. The MTP3 information is populated based on the SS7 network information that is associated with the selected trunk group. As previously noted, each switching exchange contains a provisioned association (usually static) between routesets and trunkgroups. The IAM is then transmitted onto a signaling link toward the destination identified in the message by the DPC. If quasi-associated signaling is used, the message's

next-hop node is an STP that will route the message to the intended SSP. If associated signaling is used, the IAM is transmitted directly to the SSP that is associated with the trunk being set up. SSP A starts timer T7, which is known as the *network protection timer,* or the *awaiting ACM timer*, to ensure that an ACM is received in response to the IAM.

When SSP B receives the MTP3 message, it recognizes it as an ISUP message by the SIO's Service Indicator bit. Then the message is passed to ISUP for processing, during which it extracts the message information. An IAM indicates a request to set up a call so SSP B enters the call processing phase for a trunk origination. The CdPN and Calling Party Category fields provide key pieces of information from the IAM for SSP B to complete number translations for this simple call.

NOTE The CdPN is commonly used to enter number translations processing; however, depending on call specifics, other fields can be used for translation. For example, calls involving ported numbers can use the Generic Address Parameter during number translation to determine the outgoing call destination.

In this example, the number translates to a subtending line of SSP B, which checks the line to determine whether it is available. An ACM is built and sent to SSP A, notifying that the call can be completed and is proceeding. At this point, the speech path in the backward direction (from SSP B to SSP A) should be cut through to allow the ring-back tone to be sent over the bearer channel from the terminating exchange to the originating exchange. This indicates that the terminator is being alerted.

NOTE Note that the terminating office does not always send the ring-back tone. For example, ISDN can use the ACM message to notify the originating phone terminal to provide the ring-back tone.

Ringing is now applied to the terminating set, while ring back occurs at the originating set. Answer timing is usually applied at the originating switch to limit the amount of time an originator waits for answer.

When the terminating subscriber goes off-hook, an ANM is sent back to the originator to indicate that an answer has occurred. By this point, the voice path should be cut through in the forward direction to allow the conversation to take place. Note that the voice path can be cut through before receiving the ANM, but it must be cut through no later than the ANM. The call is now in the active, or talking, state. This is often a point of interest for billing procedures that require capturing the time at which a call conversation begins. For an

ordinary call, no further signaling messages are exchanged for the duration of the conversation. When either of the parties goes on-hook, it initiates signaling for the release of the call. The following section discusses Call release.

Call Release

When either the originating or terminating subscriber goes on-hook, it signals an attempt to disconnect the call. In Figure 8-3, the originator at SSP A goes on-hook. SSP A recognizes the signal to disconnect the call and sends a Release message (REL) to SSP B. SSP B responds by sending a Release Complete message (RLC) as an acknowledgement. The trunk member is freed and placed back into its idle queue to be used for another call.

Terminal Portability

The ITU defines terminal portability in Q.733.4 for allowing the called or calling party to hang up a phone and resume a conversation at another phone that is connected to the same line. When the two parties are connected over an inter-exchange ISUP trunk, suspend and resume messages are used to maintain the trunk connection until the on-hook party has gone off-hook. Terminal portability requirements for the called party exist in many countries; however, terminal portability for the calling party is not supported as often. ANSI networks do not support terminal portability for the calling party.

Circuit Suspend and Resume

In Figure 8-3, the originating subscriber goes on-hook first. The originator is normally considered in control of the call, so the circuit is released when the originator goes on-hook. If the terminator goes on-hook while the originator remains off-hook, there are two methods of handling the disconnection.

The first method is for the terminating exchange to release the call by sending a REL message to the originating exchange. This is no different than the scenario presented for a release initiated at the originating exchange; the originating switch responds with an RLC and the circuit is idled at each SSP.

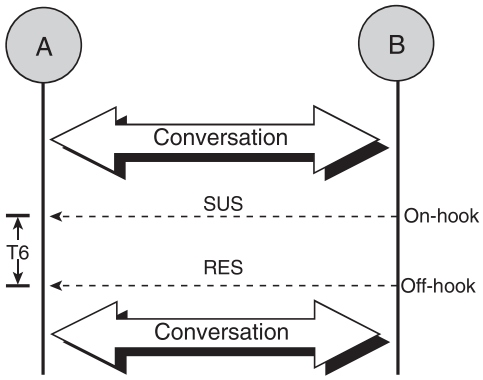
The other method is for the terminating exchange to send a Suspend (SUS) message in the backward direction when it receives a disconnect indication from the terminating line. The SUS message provides notification that the terminating party has disconnected but that the circuit connection is still being maintained. Suspending the call allows the person who receives the call an opportunity to pick up on another phone extension.

When the SUS is received, the originating exchange starts a suspend timer (Timer T6, or Timer T38 in the case of an international exchange). If the terminating party reconnects

(off-hook) before the suspend timer expires, a Resume (RES) message is sent in the backward direction, allowing the conversation to continue.

Figure 8-16 shows an example of a Suspend (SUS) and Resume (RES) being sent from the terminating exchange. If the suspend timer expires, a REL is sent in the forward direction. In the event that the originator goes on-hook during the time the circuit is suspended, the originating exchange sends a REL forward and normal call clearing takes place. The terminating exchange responds with a RLC.

Figure 8-16 *ISUP Suspend/Resume*



Support for SUS/RES varies, based on factors such as the type of service and the local network policies. For example, in the United States, SUS/RES is only supported for non-ISDN service.

ISUP and Local Number Portability

Local Number Portability (LNP) is the concept of having phone numbers that remain the same for the subscriber, regardless of whether the subscriber changes service providers or geographic location. Historically, phone numbers have been associated with a particular geographic region or a particular service provider. The actual use of LNP in the network exists today, but only to a small degree. It is being expanded in phases and will take some time before it is ubiquitous across all networks and locations. This section examines the different mechanisms used to provide portability services and how these mechanisms relate to setting up calls with ISUP.

Chapter 11, "Intelligent Networks (IN)" provides an overview of the various phases identified under the umbrella of Number Portability (NP), such as service provider portability and location portability. Some of the mechanisms used for NP employ Intelligent Network (IN) databases, so we cover NP in part both in the Chapter 11 and in this chapter.

When NP is implemented, numbers are transitioned from physical addresses that identify an exchange location to virtual addresses that identify a subscriber. A means of mapping must be used to derive a physical address in the network from the called number because the number no longer identifies a physical destination. The network in which the physical number existed before portability was introduced is called a *donor* network. Each time a number is ported and becomes a virtual address, the network has "donated" a number that previously belonged to that network. We use the term "donor" or "donor network" several times during the discussion of NP. The network in which the physical number now resides is called the *recipient* network.

Currently, four mechanisms are defined for implementing NP:

- All Call Query (ACQ)
- Query on Release (QOR)
- Dropback or Release to Pivot (RTP)
- Onward Routing (OR)

Each method has its merits in terms of resource efficiencies, maintainability, and competitive fairness among network operators, but those topics are outside of the scope of the book. The details of how each mechanism is implemented also vary from country to country. The following section provides a general understanding of NP and how it affects the ISUP call flow and messages.

All Call Query (ACQ)

ACQ sends an IN query to a centrally administered database to determine the call's physical address or routing address. Chapter 11 discusses the details of the IN query. The way the routing number returned by the query is used varies based on national standards. The following example illustrates how the routing number is used in North America.

The number returned from the database is a Location Routing Number (LRN) that identifies the exchange serving the called number. Each exchange in the network is assigned an LRN. The IAM sent after the database query is performed contains the LRN in the CdPN field. The call is routed on the CdPN using switching translations to reach the destination exchange. The IAM also includes a Generic Address Parameter (GAP) with the original dialed number (the virtual address). This allows the destination exchange to set up the call to the intended subscriber because the LRN can only identify the exchange. The Forward Call Indicators of the IAM include a Ported Number Translation Indicator (PNTI), which indicates that a query for the ported number has been performed.

Query On Release (QOR)

QOR routes the call from the originator to the donor network's ported number in the same manner used prior to NP. The donor network releases the call back with a cause value of *Number Portability QOR number not found* (ITU causes value 14, ANSI causes value 27 in the REL message). The originating network then performs a query to an NP database to determine what routing number to use in the IAM in order to reach the recipient network.

Dropback (Also Known as Release to Pivot)

Dropback, or *Release to Pivot* (*RTP*), routes the call to the ported number in the donor network, just like QOR. However, instead of having the originating network query for the number, the donor exchange provides the routing number for the ported number when it releases back to the originator.

Onward Routing (OR)

Onward Routing (*OR*) also routes the call to the donor network's ported number. It differs from QOR and RTP in that it does not release the call back to the originating network. Rather, it references an internal database to determine the new routing number that is associated with the ported number and uses the new number to route the call.

Using the QOR and RTP mechanisms, an IAM is sent and an REL received back from the donor network, therefore, requiring a subsequent call attempt. The ACQ and OR do not release back or require subsequent call attempts. The OR mechanism creates additional call legs because the call is being connected through the donor network rather than being directly set up to the recipient network.

ISUP-ISUP Tandem Calls

Previous scenarios have focused on line-ISUP and ISUP-line calls. ISUP processing at a tandem switch occurs in the same sequence as the line to ISUP calls we discussed previously. However, in the case of ISUP-ISUP calls, the trigger for call processing events on the originating and terminating side are incoming ISUP messages.

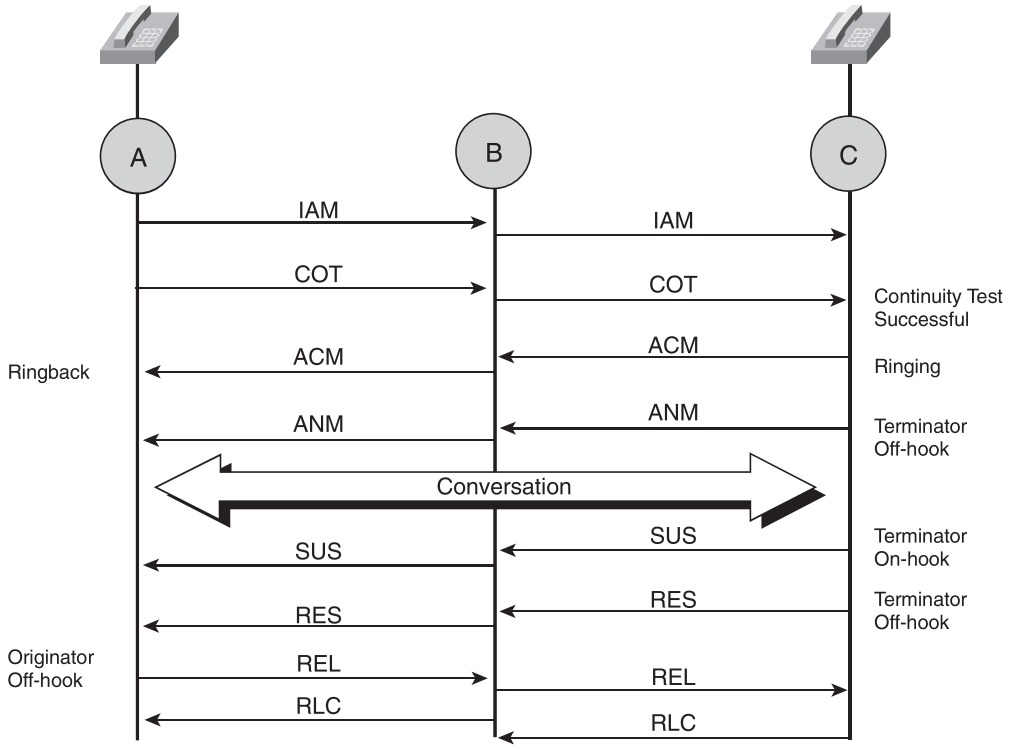
This section discusses the following three areas that are related to ISUP processing at a tandem node:

- ISUP Message Processing
- Continuity Testing
- Transporting Parameters

ISUP Message Processing at a Tandem

In Figure 8-17, the call origination at SSP B is based on an incoming ISUP origination (IAM) from another exchange. The fields that are necessary for number translation, such as CdPN, are extracted from the IAM and used to process the call at the tandem node to determine the outgoing destination. The translation and routing process results in the selection of an outgoing ISUP trunk. An IAM is sent in the forward direction to SSP C, updating fields in the message as necessary. For example, a new CdPN might be inserted as a result of translations. The NOC field is updated based on information such as whether a satellite is being used for the voice circuit or whether a continuity check is being performed.

Figure 8-17 *ISUP-ISUP Tandem Calls*



When the ACM and ANM are received at SSP B, they are propagated to SSP A, updating fields such as the BCI as necessary. Each leg of the call cuts through the speech path in the same manner discussed in the "Detailed Call Walk-Through" section of this chapter.

When SSP A sends an REL message, SSP B responds with an RLC. It does not need to wait for the RLC to be sent from SSP C. Next, SSP B sends an REL to SSP C and waits for RLC to complete the release of that leg of the call. Keep in mind that even though some messages in a multi-hop ISUP call are propagated, the entire call actually consists of independent circuit segments. The release procedure is a reminder of this fact because the RLC can be sent immediately after receiving a REL.

Continuity Testing

When a call is set up across multiple exchanges, continuity testing is performed independently on each leg of the call. If a call traverses three trunks across four different exchanges and continuity is done on a statistical basis, it will likely only be performed on some of the trunks involved in the call. While the actual continuity test is performed independently on each call leg, the end-to-end call setup is dependent on each leg passing the test. If a continuity test is successfully performed on the second leg of the call (SSP B to SSP C), the results are not reported until the COT results have been received from the previous leg of the call (SSP A to SSP B). If a previous leg of the call connection cannot be set up successfully, there is no need to continue. For example, if SSP A reports a COT failure, it would attempt to establish a new connection in the forward direction by selecting another circuit to set up the call. There is no need to continue the previous connection from SSP B to SSP C because the new call attempt from SSP A will come in as a new origination to SSP B.

Transporting Parameters

A tandem node can receive ISUP parameters that are only of interest to the destination exchange. This is particularly true of many optional parameters, which are passed transparently in the outgoing messages across tandem nodes. However, the tandem might update some fields during call processing, based on new information encountered while processing. For example, a tandem node that selects an outgoing ISUP facility over a satellite connection would update the NOC *Satellite Indicator* field in the outgoing IAM. This distinction is made because the tandem node might be required to have knowledge of how to process some parameters, but not others. When parameters are passed across a tandem node without processing the information, it is sometimes referred to as "ISUP transparency." Since the parameters do not need to be interpreted by the tandem, they are considered transparent and are simply relayed between the two trunks.

Interworking with ISDN

ISDN uses a common channel (the D channel) for access signaling; this compliments the common channel network signaling ISUP uses and provides a complete digital signaling path between end users when ISDN is used for network access and ISUP is used throughout the core network. The ISUP/ISDN interworking specifications for ITU-T, ETSI, and Telcordia are found in the following standards:

- ITU-T Q.699—Interworking of Signaling Systems—Interworking Between Digital Subscriber Signaling System No. 1 and Signaling System No. 7
- ETSI EN 300-899-1 Integrated Services Digital Network (ISDN); Signaling System No. 7; Interworking Between ISDN User Part (ISUP) Version 2 and Digital Subscriber Signaling System No. one (DSS1); Part 1: Protocol Specification
- Telcordia GR-444 Switching System Generic Requirements Supporting ISDN Access Using the ISDN User Part

A correlation exists between the ISDN messages from the user premises and the ISUP messages on the network side of the call. Figure 8-18 illustrates this correlation using an ISDN-to-ISDN call over an ISUP facility. Table 8-1 lists the message mapping that occurs between the two protocols for the basic call setup shown in the diagram.

Figure 8-18 *ISUP-ISDN Interworking*

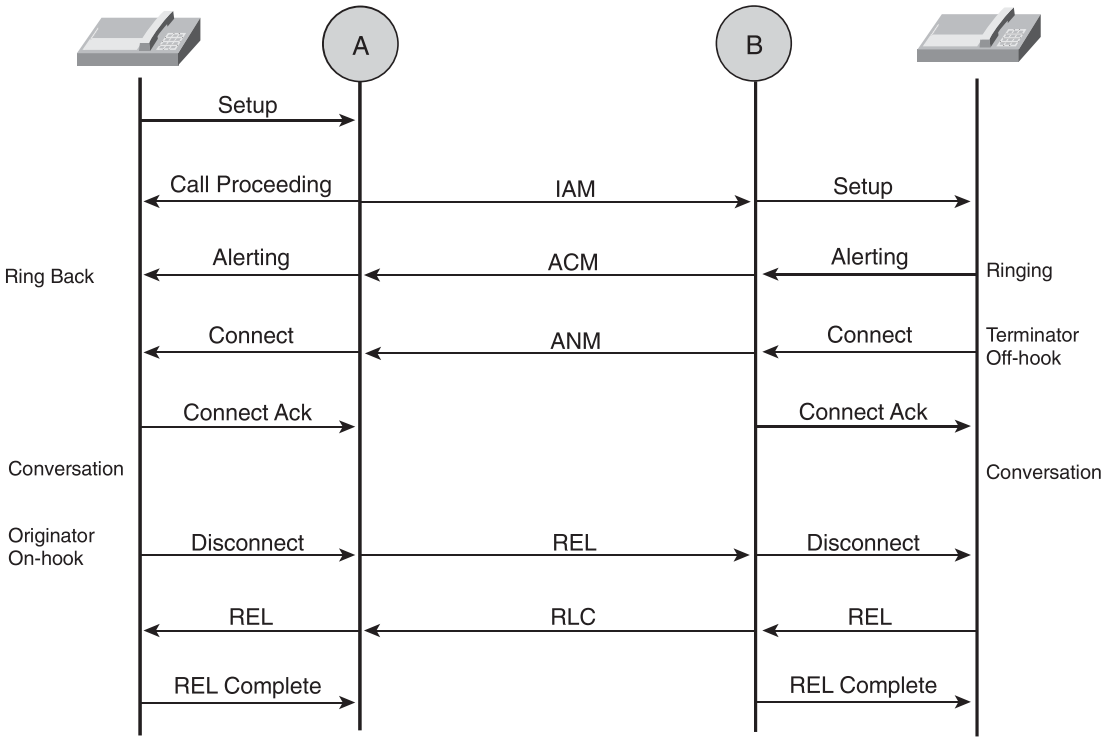


Table 8-1 *Message Mapping Between ISDN and ISUP*

ISDN	ISUP
Setup	IAM
Alerting	ACM (or CPG)
Connect	ANM (or CON)
Disconnect	REL
Release	RLC

Many of the fields within these messages also have direct mappings. For example, the bearer capability field in the ISDN Setup message maps to the ANSI User Service Info or the IAM's ITU Transmission Medium Requirements field. There are fields that have no

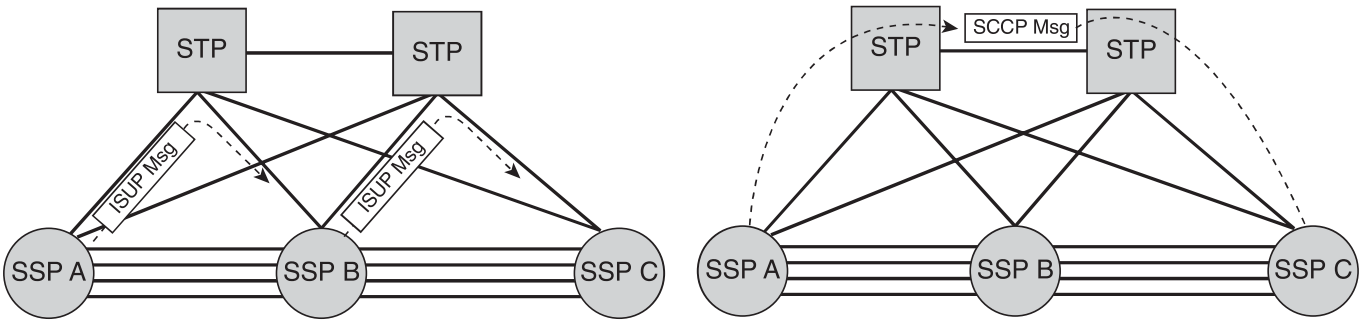
direct mapping, such as the NOC Indicators and FCIs in the IAM. Many of the fields that do not have direct mapping contain network-specific information that would not be useful for the ISDN signaling endpoint.

End-to-End Signaling

The ability to perform end-to-end signaling is accomplished using ISDN access signaling and ISUP network signaling. End-to-end signaling is the passing of information across the network that is only pertinent to the two communicating endpoints. Generally, this means that the two phone users are connected across the network. The network itself can be viewed as a communications pipe for the user information.

There are two different methods for end-to-end signaling over ISUP: the *Pass Along Method (PAM)* and the *SCCP Method.* As shown in Figure 8-19, PAM exchanges end-to-end signaling by passing along information from one node to the next, based on the physical connection segments. The SCCP method uses a call reference to pass end-to-end data between endpoints without having to pass through each individual hop. PAM is the method that is currently used throughout the network for end-to-end signaling.

Figure 8-19 *ISUP End-to-End Signaling*



A.) PAM End-to-End Signaling

B.) SCCP End-to-End Signaling

ISDN Signaling Indicators in the IAM

The following set of fields in the IAM *FCI* comprises what is known as the *Protocol Control Indicator (PCI)*:

- End-to-end method indicator
- Interworking indicator
- IAM segmentation indicator
- ISDN User Part indicator

These fields provide information about the protocol communication across the ISUP connection. The Protocol Control Indicator fields are of particular importance to ISDN because they identify whether ISDN signaling can be exchanged across the network. If the Interworking Indicator is set to *interworking encountered*, it indicates that a non-SS7 connection (such as MF signaling) has been used in a circuit connection. It also indicates that SS7 signaling cannot be exchanged across this connection because it would prevent an ISDN terminal from being able to relay signaling across the network that depended on an SS7 connection all the way.

The ISDN User Part indicator field indicates whether ISUP has been used for every call leg up to the current exchange. If this field is set to *ISDN User Part not used all the way*, it might not be possible to pass ISDN information across the network.

The ISDN User Part preference indicator field indicates to the receiving node whether the call needs an outgoing ISUP connection.

The preference field might contain the following values:

- ISDN User Part preferred
- ISDN User Part required
- ISDN User Part not required

For calls originating from an ISDN set, the preference field is set to *ISDN User Part preferred* unless specified otherwise by different services. If it is available during outgoing trunk selection, an ISUP facility is chosen; an ISUP facility is "preferred," but not necessarily required. If an ISUP facility is not available, the call is still set up if a non-ISUP facility is available. If a call is being established that requires the ability to pass service information—such as end-to-end signaling—across the network, the preference field is set to *ISDN User Part required*. A call with a preference of "required" is not set up unless an ISUP facility is available. For example, setting up a multichannel ISDN video connection would not be possible without end-to-end ISUP signaling.

Although the PCI provides information about the connection across the network, it does not specify the actual protocol of the access signaling. The FCI includes the *ISDN access indicator* bit to indicate whether the originating terminal is an ISDN set.

Supplementary Services

Supplementary services are one of the ISUP advantages noted in this chapter's introduction. ISUP provides many messages and parameters that are explicitly created for the support of supplementary services across the network. The introduction of ISUP has helped to greatly standardize widely used services, allowing them to operate across networks and between vendors more easily. Service specifications still vary between different networks based on differences in locales and market needs. ISUP provides the flexibility to accommodate these differences using a rich message set and a large set of optional parameters.

The ITU-T defines a core set of widely used ISDN services in the Q.730–Q.739 series of specifications using ISUP network signaling. The actual specification of these services at the national level can vary. In addition, national networks and private networks offer many services outside of those that are specified by the ITU-T. In the United States, Telcordia has defined a large number of services in various Generic Requirements (GR) specifications for U.S. network operators.

The list of services implemented on modern telephony switches has grown quite long. However, the purpose of this section is not to explore the services themselves, but to provide examples of how ISUP is used to support them. Two examples of common services have been chosen to discuss how ISUP provides support for them: Calling Line Identification and Call Forwarding Unconditional.

Calling Line Identification (CLI) Example

ITU Q.731 specifies Calling Line Identification (CLI). Calling party information can be used at the terminating side of a call in many different ways. Following are a few examples:

- Calling Number Delivery (CND)
- Calling Name Delivery (CNAMD)
- Incoming Call Screening
- Customer Account Information Retrieval (Screen Pops)

Being able to identify the calling party allows the called party to make decisions before answering a call. For example, an end user can use call screening to allow them to choose which calls they wish to accept. A business might use the incoming number to speed the retrieval of customer account information to call centers. If the called party subscribes to Calling Name Delivery, the CgPN is used at the terminating exchange to retrieve the name associated with the number.

CLI is specifically defined by the ITU-T as:

- Calling Line Identification Presentation (CLIP)
- Calling Line Identification Restriction (CLIR)

The ISUP CdPN parameter contains an *Address Presentation Restricted indicator* that specifies whether the calling party identification can be presented to the called party. The Address Presentation Restricted indicator has the following possible values:

- Presentation allowed
- Presentation restricted
- Address not available
- Reserved for restriction by the network

If the terminating party subscribes to the CLI service, the terminating exchange uses this indicator's value to determine whether the number can be delivered. The number is delivered only if the value is set to Presentation allowed. If the connection encounters non-SS7 interworking, the address information might not be available for presentation. In addition, transit network operators might not transport the information in some cases, depending on regulatory policies. While the actual display to the end-user varies depending on location, it is quite common to see restricted addresses displayed as "private" and unavailable addresses displayed as "unknown" or "out of area."

In some networks, if the CLI is not present in the IAM, it might be requested from the calling party using an Information Request (INR) message. The originating exchange delivers the requested CLI using an Information (INF) message.

Call Forwarding Example

Call Forwarding is part of a larger suite of services known as Call Diversion services. There are many variations of Call Forwarding. The ITU-T in the Q.732 specification defines the standard set of Call Forwarding variations as follows:

- Call Forward Unconditional (CFU)
- Call Forward No Reply (CFNR)
- Call Forward Busy (CFB)

Other variations of Call Forwarding exist within localized markets. For example, Call Forwarding Selective is another variation that allows forwarding for calls that originate from selective calling numbers. For this example, we have chosen Call Forward Unconditional to illustrate the use of ISUP signaling.

In Figure 8-20, the ITU-T message flow is shown for CFU at SSP B. The ANSI message flow differs slightly from that shown for ITU. A subscriber at SSP B has forwarded their calls to a number at SSP C. When SSP B attempts to terminate the call and encounters the Call Forward service, a new IAM is sent to SSP C. Keep in mind that a call might be forwarded multiple times before reaching its destination. The additional parameters included in the IAM for Call Forwarding convey information about the first and last instances of forwarding. In our example, the IAM to SSP C contains the following parameters, specific to the call redirection:

- Redirection Information (RI)
- Redirecting Number (RN)
- Original Called Number (OCN)

The inclusion of the RI parameter varies among different networks, so it might or might not be present. The RI parameter contains the following information fields: