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Travis Russell

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In memory of my father, Travis L. Russell, who taught me that the only barriers to success are those we create ourselves.

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Preface

On September 11, 2001, a series of events changed the lives of Americans forever. The way we think, the way we work, and even the way we play were altered. These unthinkable events have certainly changed my focus on networks and specifically SS7.

After a meeting with local government regarding infrastructure protection, I was speaking to a colleague of mine when it occurred to both of us that we Americans have a difficult time with acts of terrorism because we ourselves are not devious enough to contemplate the many ways we could be attacked. It is time to start thinking deviously.

Now more than ever, it is important to understand the SS7 protocol, its many procedures, and, yes, even its vulnerabilities. I hope this book helps with that study. Although I will not put in print the vulnerabilities, I hope one can learn of these vulnerabilities through understanding the protocol itself and use this knowledge in a constructive manner.

Acknowledgments

Over the last few years I have traveled all over the country and met readers of this book's previous edition at telephone companies everywhere. The encouragement you provide is what keeps this book going. Thank you for your continuous support.

Many of you probably wonder where I go when I have questions about SS7. Tekelec is full of expertise in this area, but there are three colleagues I would like to give special mention to here: Dave Sprague, Venkat Ravishankar, and Dan Bantakul are a constant source of information, and I always enjoy our discussions on the various protocols we work with.

Of course, behind every man is a woman and family. Thanks again to my special wife and kids during this project. Your support is what keeps me writing!

Introduction

First, there was ISDN, and then came portable 800 numbers. Soon, cellular providers were talking of tying their networks together so that subscribers could roam from one cellular network to another without special roaming numbers. And now we are talking over the Internet using our computers as telephones.

But during all of these discussions, nothing (or at least very little) has been mentioned about how all of this is possible. How is it possible that many networks can communicate with one another, and which control mechanism is used to maintain the connections and data communications?

Behind the scenes is a quiet, highly reliable, fault-tolerant data communications network that links the world's telephone networks together and allows them to share vital signaling and control information. This one network, which controls telecommunications networks around the world, is being prepared to control the broadband networks that will form the framework for tomorrow's information highway.

This data communications network is so robust, so sophisticated, that few really know much about it. Only recently has anything been written about this network, and what information does exist has been sparse and ambiguous. The very standards that define the network are written to answer the needs of so many different audiences that they remain vague and reliant on reader interpretation.

This network, often dubbed the world's largest data communications network, is known as *Signaling System 7* (SS7). It started as a way to access 800 databases here in the United States, although its functions provide much more. Soon SS7 was being used to send signaling information from exchange to exchange. Today, the same network is being used to control central office switching equipment from remote locations.

SS7 is really a control network, as well as a signaling network. This is important to understand because as the information highway rolls out, and as the *Advanced Intelligent Network* (AIN) is implemented, SS7 will be relied on almost exclusively as a means for telephone companies and other service providers to share database information and switching control without human intervention.

Already we are seeing SS7 play an important role in *Local Number Portability* (LNP), a new requirement for all telephone service providers (both wireline and wireless) defined in the Telecommunications Act of 1996. SS7 continues to grow in usage as the industry discovers new ways to utilize this powerful technology.

Without SS7, AIN is not possible, cellular roaming is not seamless, 800 numbers are not portable, and the many features and services we take for granted today (such as caller ID) would not be feasible.

When someone asks me how to break into the telecommunications field, I provide one suggestion: Learn SS7. Every telephone company needs it, and every manufacturer develops around it, yet there is a lack of SS7 expertise in this country. That is what this book is all about.

As I began studying SS7 protocols and began providing seminars around the country, I came to the conclusion that there is no reference material available that provides a comprehensive view of SS7; nothing explains the various acronyms and the true applications of this network.

Hence, the reason for this book. If you are looking for more than a reference book—a tutorial on SS7, a textbook that provides real-world applications, and a futuristic look at the telephone networks—read on.

The intent of this book is to provide a comprehensive introduction to this fascinating network and its protocols, as well as a reference for those already familiar with SS7. Although the various standards are always the best source for specific details, this book provides something the standards do not: explanations as to when and why procedures are used and what they mean to the network, the subscriber, and the service provider.

Although it is based on ANSI and Bellcore standards, the text is of value to anyone seeking knowledge about SS7 networks. Although the message types and the protocol parameters may be different between countries, the principles and applications are universal.

I hope you find this to be as fascinating as I did while writing this book. Truly, this is an exciting period for the telecommunications industry as new services and applications are defined almost monthly. Keeping up with all of the new technologies and advances can be a dizzying experience. Yet the future is today, and this technology is a key player in making it all happen.

Travis Russell

Acronym List

A-links	Access links
AAL	ATM adaptation layer
ABOM	A-bis Operations and Maintenance
AC	Authentication center
ACD	Automatic call distribution
ACG	Automatic Code Gap
ACM	Address complete message
AE	Application entity
AERM	Alignment error rate monitor
AIN	Advanced Intelligent Networks
AK	Data acknowledgment
AMI	Alternate mark inversion
ANI	Automatic number identification
ANM	Answer message
ANSI	American National Standards Institute
ASE	Application service element
ASN-1	Abstract Syntax Notation One
ASP	Application Service Part
ATM	Asynchronous transfer mode
B-links	Bridge links
BIB	Backward indicator bit
BISDN	Broadband ISDN
BISUP	Broadband ISUP
BITS	Building integrated timing system
BLA	Blocking acknowledgment (message)
BLO	Blocking (message)
BOP	Bit-oriented protocol
BRI	Basic rate ISDN

BSC	Base station controller
BSDB	Business service database
BSN	Backward sequence number
BSS	Base station subsystem
BSSMAP	Base Station Subsystem Mobile Application Part
BTA	Basic trading area
BTS	Base transceiver station
BTSM	Base Transceiver Station Management
C-links	Cross-links
CAE	Communications Applications Environment
CBA	Changeback acknowledgment (message)
CBD	Changeback declaration (message)
CC	Connection confirmed (message)
CCE	Consistency check end (message)
CCEA	Consistency check end acknowledgment (message)
CCR	Continuity check request (message)
CCRA	Consistency check request acknowledgment (message)
CCS	Common channel signaling
CD	Carrier detect
CESID	Callers Emergency Service Identification
CFN	Confusion (message)
CGB	Circuit group blocking (message)
CGBA	Circuit group blocking acknowledgment (message)
CGU	Circuit group unblocking (message)
CGUA	Circuit group unblocking acknowledgment (message)
CIC	Circuit identification code
CLLI	Common Language Location Identifier
CMC	Call modification completed (message)
CMR	Call modification request (message)
CMRJ	Call modification reject (message)
CMSDB	Call management services database (message)
CON	Connect (message)
COO	Changeover order (message)
COT	Continuity test (message)
CPG	Call progress (message)
CPU	Central processor unit
CQM	Circuit query message
CQR	Circuit query response (message)
CR	Connection request (message)

CRA	Circuit reservation acknowledgment (message); consistency check request (message)
CRC	Cyclic redundancy check
CREF	Connection refusal (message)
CRG	Charge information (message)
CRM	Circuit reservation message
CRST	Cluster route set test (message)
CSA	Canadian Standards Association
CTIA	Cellular Telecommunication Industry Association
CTS	Clear to send
CVM	Circuit validation test (message)
DCE	Data communications equipment
DN	Directory number
DPC	Destination point code
DPNSS	Digital Private Network Signaling System
DRS	Delayed release (message)
DSR	Data set ready
DSU	Data service unit
DT1	Data form 1
DT2	Data form 2
DTAP	Direct Transfer Application Part
DTE	Data terminal equipment
DTMF	Dual-tone multifrequency
DTR	Data terminal ready
DUP	Data User Part
EA	Expedited data acknowledgment (message)
EAS	Exchange access signaling
ED	Expedited data (message)
EIA	Electronic Industries Association
EIR	Equipment identification register
ERR	Error (message)
FA	Framework Advisory
FAA	Facility accepted (message)
FAR	Facility request (message)
FCC	Federal Communications Commission
FDDI	Fiber Distributed Data Interface
FIB	Forward indicator bit
FISU	Fill-in signal unit
FOT	Forward transfer (message)

FR	Family of Requirement
FRJ	Facility rejected (message)
FSN	Forward sequence number
GMT	Greenwich Mean Time
GPS	Global Positioning System
GRA	Circuit group reset acknowledgment (message)
GRS	Circuit group reset (message)
GSM	Groupe Special Mobile
GUI	Graphical user interface
HLR	Home location register
IAA	IAM acknowledgment (message)
IAM	Initial address message
IAR	IAM reject (message)
IN	Intelligent Network
INA	Information network architecture
INF	Information (message)
INR	Information request (message)
IP	Intelligent peripheral
IS-41	Interim Standard-41
ISDN	Integrated Services Digital Network
ISDNUP	Integrated Services Digital Network User Part
ISNI	Intermediate Signaling Network Identification
ISO	International Standards Organization; International Organization for Standardization
ISUP	ISDN User Part
IT	Inactivity test (message)
ITU	International Telecommunications Union
ITU-D	ITU Telecommunication Development Sector
ITU-RS	ITU Radiocommunication Sector
ITU-TSS	International Telecommunications Union—Telecommunications Standardization Sector
LAN	Local area network
LAPD	Link Access Procedure on the D Channel
LATA	Local Access Transport Area
LI	Length indicator
LIDB	Line Information Database
LLI	Logical link identifier
LPA	Loopback acknowledgment (message)
LSB	Least significant bit

LSSU	Link status signal unit
MAP	Mobile Application Part
MF	Multifrequency
MIN	Mobile identification number
MLPP	Multilevel Precedence and Preemption
MSC	Mobile switching center
MSU	Message signal unit
MTA	Major trading area
MTP	Message Transfer Part
NISDN	Narrowband ISDN
NNI	Network-to-network interface
NOF	Network Operations Forum
NRC	Network Reliability Council
NRM	Network resource management
NRZ	Nonreturn to zero
OAMP	Operations, Administration, Maintenance, and Provisioning
OLM	Overload message
OMAP	Operations, Maintenance, and Administration Part
OMC	Operations and Maintenance Center
OOS	Out of service
OPC	Origination point code
OPDU	Operations Protocol Data Unit
OS	Operations system
OSI	Open Systems Interconnection
OSS	Operations Support System
PAM	Pass-along message
PBX	Private branch exchange
PCR	Preventive cyclic retransmission
PCS	Personal Communications Service
PIN	Personal identification number
POP	Point-of-presence
POTS	Plain Old Telephone Service
PRI	Primary rate ISDN
PRS	Primary reference source
PSC	PCS switching center
PSTN	Public Switched Telephone Network
PVC	Permanent virtual circuit
QoS	Quality of Service
RAO	Regional accounting office

RBOC	Regional Bell Operating Company
RD	Receive data
REL	Release (message)
RES	Resume (message)
RJ	Registered jack
RLC	Release complete (message)
RLSD	Released (message)
RSA	Regional service area
RSC	Reset confirmation (message); reset circuit (message)
RSR	Reset request (message)
RTS	Request to send; ready to send
SAM	Subsequent address message
SAP	Service access point
SAT	Supervisory audio tone
SBR	Subsystem backup routing (message)
SCCP	Signaling Connection Control Part
SCE	Service creation environment
SCLC	SCCP connectionless control
SCMG	SCCP management
SCOC	SCCP connection-oriented control
SCP	Service control point
SCRC	SCCP routing control
SDLC	Signaling data link connection
SDU	Service data unit
SF	Status field; single frequency
SGM	Segmentation message
SIB	Status indicator busy
SIE	Status indicator emergency
SIF	Service information field
SIN	Status indicator normal
SIO	Status indicator out of alignment; Service indicator octet
SIOS	Status indicator out of service
SIPO	Status indicator processor outage
SLC	Signaling link code
SLS	Signaling link selection
SLTA	Signaling link test acknowledgment
SLTM	Signaling link test message
SMDR	Station detailed message recording
SMS	Service management system

SNR	Subsystem normal routing
SOG	Subsystem out-of-service-grant
SONET	Synchronous Optical Network
SOR	Subsystem out-of-service request
SR	Special Report
SRCT	Signaling route-set congestion test
SRST	Signaling route-set test
SRT	Subsystem routing-status test
SS7	Signaling System 7
SSA	Subsystem allowed
SSP	Service switching point; subsystem prohibited
SST	Subsystem status test
ST	*Science and Technology* (publication)
STP	Signal transfer point
SUERM	Signal unit error rate monitor
SUS	Suspend message
T1A6	T1 Advisory Group
TA	Technical Advisory
TCA	Transfer cluster allowed
TCAP	Transaction Capabilities Application Part
TCP	Transfer cluster prohibited
TCR	Transfer cluster restricted
TD	Transmitted data
TFA	Transfer allowed
TFC	Transfer controlled
TFP	Transfer prohibited
TFR	Transfer restricted
TIA	Telecommunication Industry Association
TR	Technical Reference
TRA	Traffic restart allowed
TRW	Traffic restart waiting
TUP	Telephone User Part
UBA	Unblocking acknowledgment (message)
UCIC	Unequipped circuit identification code
UDT	Unitdata
UDTS	Unitdata Service
UL	Underwriters Laboratories
UNI	User-to-network interface
UPA	User part available

UPT	User part test
USIS	User-to-user indicators
USR	User-to-user information (message)
VLR	Visitor location register
VPCI	Virtual path connection identifiers
WATS	Wide Area Telephone Service
XUDT	Extended Unitdata
XUDTS	Extended Unitdata Service

Chapter

1

Signaling System #7

Signaling System #7 (SS7) has become one of the most important assets within any carrier's network. Already deemed important for interconnecting calls from one network to the next, SS7 has also become a network rich in user data.

SS7 is really a control protocol, used to provide instructions to the various elements within a telephony network. These instructions may be how to route a call through the network, what features a caller has subscribed to, or, in the case of number portability, which carrier will be used to handle the call.

In order to provide this level of instruction, a great deal of information must be sent from one element to another. Everything from the caller's telephone number to his or her calling card number, as well as other pertinent data, is sent through the network to the various network elements involved in connecting the caller to his or her destination. If there were a means of trapping all this information and storing it for analysis (which, of course, there is), carriers would find a rich resource for identifying the users of their network.

The data can be used for determining new marketing campaigns, the success of new feature offerings, and much more. In fact, I often refer to this data as the "three W's:" who is using the network, when they are using the network, and why they are using the network. This data is crucial to the success of any business to ensure they are meeting the needs of their customers.

Many carriers are just now realizing the benefits of mining the data traversing the SS7 network and are utilizing this data to maintain revenue assurance in all aspects of the business. SS7 has even become an important revenue source for carriers who have learned to tap its links and interface to back-office billing systems.

What was once an obscure, little-known technology has become one of the industry's most prized possessions. However, SS7 will not live forever. Particular aspects of SS7 will continue to thrive throughout the signaling networks, but already the lower layers of the SS7 protocols are being replaced by protocols based on the *Transmission Control Protocol/Internet Protocol* (TCP/IP). Although the applications (circuit connection and database access) will not change, the transport mechanism is already changing.

Still, SS7 is a long way from obsolescence and is extremely important to understand if one plans on making a career out of telecommunications. To fully understand how and why SS7 is used, one first must understand the basics of telephony as well as basic signaling.

Introduction to Telephony Signaling

Why is signaling needed? To understand the answer to this question, you must first understand the mechanics of a telephone call. When a subscriber picks up a telephone receiver, an electrical signal is sent over a wire to a telephone switch. The telephone switch detects the electrical current on this wire and interprets this "signal" as a request for a dial tone.

But let's say the subscriber wants to transmit data over this same line using packet switching rather than an analog modem. The information sent to the telephone switch has to define the transmission as digital data and not voice before the switch can determine how to handle the call. This is only one portion of the call.

To transmit the data to another network, the switch must determine first how the data is to be routed (to what destination) and which circuits to use to reach the destination. After this has been determined, some form of request must be sent to the telephone switch on the other end of the circuit to request a connection. This continues all the way through the network, with the same requirements at each leg of the call. Telephone switches need the capability to signal one another and share information regarding the type of transmission, how the transmission is to be routed (call destination), and what the contents of the transmission are (audio, video, data, and so on).

If there is to be special handling or routing for a call, the telephone switches involved in routing the call must be able to obtain these instructions. Rather than store routing instructions for every single telephone number in the world within each and every telephone switch, each network is responsible for their own network database. The telephone switches then need the capability to connect and communicate with these databases to obtain the special instructions.

This is a high-level view of what signaling networks really do. They enable telephone switches (and now packet switches) to communicate directly with one another and share information needed to process any type of call autonomously. SS7 was originally designed for the analog telephone network, but has continually undergone changes and enhancements to accommodate the ever-changing world of telecommunications. Today SS7 is used for data, video, voice, audio, and even *Voice over IP* (VoIP) networks.

Ever since the beginning of the telephone, signaling has been an integral part of telephone communications. The first telephone devices depended on the receiving party standing next to the receiver. Early telephones did not have ringers like today's telephones and used crude speakers to project the caller's voice into the room. If the person being called was not within close proximity of the speaker, he or she would have no indication of an incoming call.

After the formation of the Bell Telephone Company, Alexander Graham Bell's faithful assistant Thomas Watson invented the telephone ringer. The ringer served one purpose: to alert the called party of an incoming call. When the called party lifted the receiver, DC battery and ground was used to indicate the called party had answered the telephone and completed the circuit. Although not having an immediate impact, this method became important when the first telephone exchange was created. Lifting the receiver and allowing a DC current to flow through the phone and back through the return of the circuit would light a lamp on the operator's switchboard. This signaled the operator when someone needed to place a telephone call and was often accompanied by a buzzer.

Once the operator answered the switchboard, information was exchanged with the operator. The person calling was identified so a record of the call could be made (for billing if the call was long distance), and the person being called was identified so the operator knew how to connect the call. This is not much different than telephone signaling today, although signaling has evolved over the decades to include significantly more information than these early methods could.

Consider the typical long-distance telephone call today. When a caller dials the area code and prefix of the telephone number, the local exchange must determine how to route the call. In addition, billing information must be passed to a central database. If the caller is using a contemporary digital facility (such as T-1 or an *Integrated Services Digital Network* [ISDN]), information regarding the digitization of the line must also be provided.

Early signaling methods were analog and had a limited number of states, or values, that could be represented. They were also limited to audible tones because they used the same circuit for both signaling and voice. The tones would sometimes interfere with the call in progress, and sometimes the voice transmission itself would be interpreted as part of the signaling and release the call.

Another problem with early signaling methods was the fact that the circuit used for the call would be busy from the time the caller started dialing until the call was completed. Since the signaling was sent through the same circuit as the voice transmission, it was necessary to connect the facility end to end even if there was no voice transmission. For example, if the number being called was busy, the facility would still be connected end to end so a busy tone could be sent through the circuit to the caller. This is not an efficient use of facilities, and as the demand grew for telephone service, it placed a heavy burden on telephone companies with limited facilities.

Many telephone companies in metropolitan areas such as Los Angeles were facing substantial investments to add new facilities to support the millions of customers that were creating an enormous amount of traffic. The telephone companies had to find a way to consolidate their facilities, making more efficient use of what they had. In addition, they needed a service that would vastly improve their network's capability and support the many new services being demanded by subscribers.

Europe had already begun the process of digitizing the network in the early 1960s. One of the first steps was to remove signaling from the voice network and place the signaling on a network all its own. This way, the call setup and teardown procedures required with every call could be faster than the previous methods, and voice and data circuits could be reserved for use when a connection was possible, rather than maintaining the connection even when the destination was busy. *Common Channel Signaling* (CCS) paved the way for services the early pioneers of signaling never dreamed of. CCS is the concept that ISDN and SS7 are based upon.

The concept is simple. Rather than use voice trunks for signaling and voice transmission, they are used only when a connection is established and voice (or data) transmission takes place. For instance, when a call is placed to a distant party using conventional signaling, the signaling for that call begins from the time the caller lifts the receiver and goes off-hook until the caller goes back on-hook. After the end office has received the dialed digits, an outgoing trunk to the destination end office is seized, based on a routing table entry and the digits dialed.

The voice circuit remains connected and in use by the telephone switch, even if the distant party never answers the call, until the calling party hangs up. Meanwhile, other circuits are being tied up in a similar fashion. By removing the signaling from the voice network, and placing it on a network of its own, the voice circuit remains available for a longer period of time. This means the availability of voice circuits is higher and the need for additional circuits decreases. The facilities used for signaling provide signaling for many voice circuits, hence the name CCS.

Other efficiencies are provided by SS7 (albeit not always implemented). Take the case of service tones and intercept recordings. Typically, when a caller dials a number that is busy, he or she receives a busy tone from the destination (terminating) switch. This, of course, requires the use of the voice circuit (or at least one side of the circuit) for the audible tone. The same is true for intercept recordings.

With SS7, the caller's local office could provide these tones and recordings at the command of the distant office. This implementation would offer even more efficiencies than the present method used throughout North America (voice circuits are two-way transmission circuits, with a transmit and receive, so tones and recordings are sent in one direction).

SS7 does provide much faster connections and teardowns than conventional signaling methods. Even if voice circuits do get connected, with the speed of the signaling network, circuits can be disconnected and quickly connected again for a new call. This is especially true of long-distance calls where many segments of circuits are bridged between multiple switches to connect a call end to end. Each segment is connected and released individually, making it available for other calls.

Signaling today uses existing digital circuits. By using existing digital circuits, signaling for many circuits can be consolidated onto one signaling link, using a fraction of the bandwidth required if conventional signaling were used. One digital data link can carry the signaling information for thousands of trunks and maintain thousands of telephone calls. When TCP/IP is used as the transport for signaling, this number increases significantly.

With SS7, the amount of information that can be provided is virtually unlimited. This opens the potential for many more uses of the signaling data. Signaling takes place in two parts of the telephone network: between the subscriber and the local end office, and from switching office to switching office within the telephone company network. The signaling requirements are similar, though interoffice signaling can be more demanding.

As the telephone network grew more sophisticated, the signaling methods grew as well. Signaling between the subscriber and the central office now includes the calling party number, which is forwarded to the called party and displayed before the phone is even answered. Interoffice signaling now includes information obtained from regional databases, pertaining to the type of service a subscriber may have or billing information.

In fact, the first usage of SS7 in the United States was not for call setup and teardown, but for accessing remote databases. The opposite is true of Europe and other international communities, where C7 is used for call setup and teardown, but the concept of centralized databases for custom call routing is still new. In the 1980s, the U.S. telephone companies offered a new service called *Wide Area Telephone Service* (WATS), which used a common 800 area code regardless of the destination of the call. This posed a problem for telephone-switching equipment, which uses the area code to determine how to route a call through the *Public Switched Telephone Network* (PSTN).

To overcome this problem, a second number is assigned to every 800 number. This second number is used by the switching equipment to actually route the call through the voice network. But the number must be placed in a centralized database where all central offices can access it. When an 800 number is dialed, the telephone company switching equipment uses a data communications link to access this remote database and look up the actual routing number. The access is in the form of a message packet, which queries the network for the number. The database then responds with a response message packet, providing the routing telephone number as well as billing information for the 800 number. The switching equipment can then route the call using conventional signaling methods.

SS7 provides the data communications link between switching equipment and telephone company databases. Shortly after the 800 number implementation, the SS7 network was expanded to provide other services, including call setup and teardown. Still, the database access capability has proven to be the biggest advantage behind SS7 and is widely used today to provide routing and billing information for all telephone services, including 800 numbers, 900 numbers, 911 services, custom calling features, caller identification, and many new services still being defined.

800 numbers at one time belonged to one service provider. If subscribers wanted to change service providers, they had to surrender their 800 number. This was due to the location of the routing information. All routing information for 800 numbers is located in a central database within the carrier's network and accessed via the SS7 network. SS7 is now used to allow 800 numbers to become transportable and to provide subscribers the option of keeping their 800 numbers even when they change service providers.

When someone dials an 800 number today, the telephone switch sends a query to a network database to first determine which carrier the 800 number belongs to. After the switch receives this information, it can direct a query to the network of the carrier owning the 800 number and translate the 800 number into an actual routing number. This concept was later extended to support number portability.

Without SS7, number portability would be impossible. *Local Number Portability* (LNP) is a service mandated by the *Federal Communications Commission* (FCC) in

1996 that requires telephone companies to support the porting of a telephone number. The intent of this mandate was to open the various markets to competition and provide customers with more incentive to switch service providers without any disadvantage (such as losing the telephone number already printed on stationary).

If customers wish to change their service from *Plain Old Telephone Service* (POTS) to ISDN, they would normally be forced to change telephone numbers. This is because of the way telephone numbers are assigned in switching equipment, with switches assigned ranges of numbers.

The same is true if a subscriber decides to change to a new carrier offering local service in his or her area. Changing telephone companies would require surrendering your old telephone number and receiving a new number from the new carrier. With number portability, subscribers can keep their old numbers and still change carriers.

This requires the use of a database to determine which switch in the network is now assigned the number, which is very similar to the way 800 numbers are routed. Future implementations of LNP will support subscribers moving from one location to another without changing their telephone number (even if they move to a new area code). This obsoletes the former numbering plan and the way calls are routed through the telephone network.

Another use for this type of database is being developed today. As more and more people begin using their computers and the Internet for making telephone calls, the need to route calls to these devices becomes more complex. The Internet does not use telephone numbers as part of the addressing, but uses IP addresses instead. To route calls to these devices, a database will be required that is capable of translating IP addresses to some form of a telephone number that can be used within the traditional PSTN.

In addition to database access, the SS7 protocol provides the means for switching equipment to communicate with other switching equipment at remote sites. For example, if a caller dials a number that is busy, the caller may elect to invoke a feature such as automatic callback. When the called party becomes available, the network will ring the caller's phone. When the caller answers, the called party phone is then rung. This feature relies on the capabilities of SS7 to send messages from one switch to another switch, allowing the two systems to invoke features within each switch without setting up a circuit between the two systems.

Cellular networks use many features requiring switching equipment to communicate with each other over a data communications network. Seamless roaming is one such feature of the cellular network that relies on the SS7 protocol.

Cellular providers use the SS7 network to share subscriber information from their *home location registers* (HLRs), so cellular subscribers no longer have to register with other service providers when they travel to other areas. Cellular providers can access each other's databases and share the subscriber information so that subscribers can roam seamlessly from one network to another.

Before deploying SS7, cellular providers were dependent on X.25 networks to carry *Interim Standard-41* (IS-41) signaling information through their network. This did not enable them to interconnect through the PSTN because the X.25 network was not compatible with the PSTN signaling network (SS7). Wireless carriers own some of the largest SS7 networks deployed today.

Today SS7 has been deployed throughout the *Bell Operating Companies* (BOCs) network and is being deployed by almost all independent telephone companies and interexchange carriers as well. This makes SS7 the world's largest data communications network, linking telephone companies, cellular service providers, and long-distance carriers together into one large information-sharing network.

Calling card validation is another important function of these databases and provides security against telephone fraud. Personal identification numbers are kept in a subscriber database and verified every time a call is placed using a calling card.

We have discussed the basics of signaling previously. Several forms of signaling are used for different aspects of a telephone call. Conventional DC signaling, in-band signaling, out-of-band signaling, digital signaling, and CCS are all used for different applications. It is important to understand all aspects of signaling, how it is used, and the relationships between the various signaling methods.

To fully understand what SS7 is about, one must understand the conventional signaling methods used prior to SS7 in telephone networks. The following discussion explains the signaling methods used prior to SS7.

onventional Signaling

Conventional signaling relies on many different types of mechanisms, depending mostly on the location within the network. *Dual-tone multifrequency* (DTMF) is used between the subscriber and the end office. *Single frequency* (SF) is used between telephone company offices. Following is an example of how conventional signaling is used to process a call.

DC signaling relies on a DC current to signal the distant end. The simplest example of DC signaling is used in POTS between the subscriber and the local end office. When a subscriber goes off-hook, a DC current from the central office is allowed to flow through the telephone (the switch-hook provides the contact closure between the two-wire interface) and back to the central office. The central office switch uses a DC current detector to determine when a connection is being requested.

In the case of special circuits (such as tie lines), two separate circuits are used for the purpose of signaling (referred to as the E and M leads). For these circuits, the M lead is used to send a −48 V DC current or ground to the distant switch (implementation-dependent). The M lead of one switch must be connected to the E lead of the distant switch. When the distant switch detects a current on its E lead, it closes a relay contact and enables the current to flow back to the sending switch through its M lead. When the sending switch detects the current flow on its E lead, the connection is considered established and transmission can begin on the separate voice pairs.

The central office acknowledges receipt of the loop current by sending a dial tone. A dial tone signals the subscriber to begin dialing the telephone number. This can be done using a rotary dial or a DTMF dial. Rotary dials use a relay to interrupt the current, creating pulses (10 pulses per second). The central office switch counts each series of pulse "bursts" to determine the number dialed.

When DTMF is used, the dial creates a frequency tone generated by mixing two frequencies together (hence the name dual tone). The central office switch "hears" these tones and translates them into dialed digits.

After the telephone number has been dialed, the central office switch must determine how to connect to the destination. This may involve more than two central offices. A facility (or circuit) must be connected between every telephone company office involved in the call. This circuit must remain connected until either party hangs up. The originating office determines which circuits to use by searching its routing tables to see which office it must route the call through to reach the final destination. That office, in turn, will search its routing tables to determine the next office to be added to the call.

Once the circuits are all connected, the distant party can be alerted by sending voltage (80 V AC at 20 Hz) out to the telephone. This activates a ringer inside the telephone. At the same time, the distant telephone company switch sends a ringback tone to the originator to alert the caller that the called party's phone is being rung. When the distant party answers, the ringback tone is interrupted and the circuits now carry the voice of both callers.

If the called party is busy, the same facilities are used so that the far end office can send a busy tone back to the originator. This means those facilities cannot be used for any other calls and are being tied up to send the busy tone.

The limitations of DC signaling are somewhat obvious. For example, the telephone number of the originator cannot be sent to the called party (at least not without long delays in setup). Signaling is limited to seizing circuits, call supervision, and disconnect. Because DC signaling uses the voice trunk, the trunks are kept busy even when the two parties are never connected.

In-Band Signaling

In-band signaling is used when DC signaling is not possible, such as in tandem offices. In-band signaling uses tones in place of a DC current. These tones may be *single frequency* (SF) tones, *multifrequency* (MF) tones, or DTMF. The tones are transmitted with the voice. Because these tones must be transmitted over the same facility as the voice, they must be within the voice band (0 to 4 kHz). There is the possibility of false signaling when voice frequencies duplicate signaling tones. The tones are designed for minimal occurrence of this, but this is not 100 percent fault tolerant. Signal delays and other mechanisms are used to prevent the possibility of voice frequencies from imitating SF signals.

SF signaling is used for interoffice trunks. Two possible states exist: on-hook (idle line) or off-hook (busy line). To maintain a connection, no tone is sent while the circuit is up. When either party hangs up, a disconnect is signaled to all interconnecting offices by sending a tone (2.6 kHz) over the circuit. Detectors at each end of the circuits detect the tone and drop the circuit.

SF signaling has become the most popular of all the in-band methods, and the most widely used of all signaling methods. SF is still in use today in some parts of the telephone network. However, as deployment of the SS7 network spreads, SF is no longer needed.

MF is much like DTMF and is used to send dialed digits through the telephone network to the destination end office. Because voice transmission is blocked until a connection to the called party is established, there is no need for mechanisms that prevent the possibility of voice imitating signaling tones. MF is also an interoffice signaling method used to send the dialed digits from the originating office to the destination office.

Out-of-Band Signaling

Out-of-band signaling has not shared the popularity and widespread usage of SF signaling. Out-of-band signaling was designed for analog carrier systems, which do not use the full 4 kHz bandwidth of the voice circuit. These carriers use up to 3.5 kHz and can send tones in the 3.7 kHz band without worrying about false signaling. In other words, some frequencies within the 4 kHz bandwidth are left as "buffers" and are not used for anything. This is where out-of-band signaling takes place, using audible tones. It is called out-of-band signaling because the signaling takes place outside of the voice frequency bands; however, it is still sent on the same facility as the voice. Out-of-band signaling is an analog technology and is of no advantage today.

SS7 is often referred to as out-of-band signaling. This is not an accurate label for SS7 because the messages sent in SS7 are not sent in the frequency band outside of the voice band. From a purist perspective, SS7 is best described as CCS because it is sent on a completely separate facility from the voice.

Digital Signaling

As digital trunks became more popular, signaling methods evolved that greatly enhanced the reliability of the network. One technique used in digital trunks (such as DS1) is the use of signaling bits. A signaling bit can be inserted into the digital voice bitstream without sacrificing voice quality. One bit is "robbed" out of designated frames and dedicated to signaling (robbed-bit signaling). The digitized voice does not suffer from this technique since the loss of one bit does not alter the voice signal enough to be detectable by the human ear.

Because of its digital nature, digital signaling is much more cost effective than SF. SF requires expensive tone equipment both for sending and detection, whereas digital signaling can be detected by any digital device loaded in the switching equipment and can create any kind of signaling information. This has fueled the efforts to make carriers digital rather than analog.

Digital signaling has another fundamental difference. It does not use messages as SS7 and other message-based protocols do. Rather, it uses a limited number of bits to represent simple, connection-specific data. This limits the type of signaling data it can provide. This form of signaling is used to maintain the connection of a digital facility and does not contain any information about the call itself (therefore, it can be classified as transmission-level signaling).

Common Channel Signaling (CCS)

CCS uses a digital facility, but places the signaling information in a time slot or channel separate from the voice and data it is related to. This enables signaling information to be consolidated and sent through its own network apart from the voice network. It is this method that is used in SS7 today.

In addition, this method of signaling is capable of sending and receiving large amounts of data within packetized messages, supporting an unlimited number of signaling values. This is what adds value to SS7 today. By sending data messages rich in detail, the network can operate much more intelligently than with any other method of signaling. Even information retrieved from a remote database can be transferred from one entity to another using CCS.

It was mentioned earlier that SS7 is sometimes referred to as out-of-band signaling. When one compares the methods used to transmit SS7 with conventional out-of-band signaling, it becomes clear why SS7 is best defined as CCS. Of course, out-of-band signaling is also an analog method, whereas CCS is digital.

To understand how signaling is used, you must understand the various hierarchies of telephone networks and the basics of telephony. The next section provides am overview of telephony.

Basic Telephony

Telephony is the science of transmitting voice over wires electronically (or close anyway). Telecommunications is often used to refer to both the voice and data industry. This section will deal primarily with voice telephony, outlining the various aspects of the telephone network and the carriers responsible for those networks.

Bell System Hierarchy

The Bell System network can be divided into several distinct functions: signaling, data, video, audio, and voice switching. The signaling network is the SS7 network. Data transmission is routed away from the telephone switches onto separate data networks, either owned and maintained by the telephone company or leased through a data provider. The video network is used for transmitting programs from television studios out to their transmitters. This is usually over DS4/5 facilities. An audio transmission is done through radio broadcasts and is also transmitted from the studio out to the radio tower transmitters. The switching network is the portion used for the transfer of voice from one subscriber to another through the telephone service providers' network.

It is important to understand this separation of networks because what is happening today with the convergence of networks has a big impact on how telephone companies manage these different transmissions.

Telephone networks were designed for voice transmission. The typical engineering rule used when building a voice network is to provide enough capacity for the typical number of phone calls within a given period of time. There is a limited duration of time applied to each phone call as a principle and a specific number of calls per hour per circuit. Without getting into all the equations for these engineering rules, the point is this: telephone networks were not designed for long sessions dialed into the Internet!

Internet traffic is now "offloaded" from the switch network right at the local telephone office and is immediately routed to an *Internet service provider* (ISP) through dedicated circuits leased by the local telephone company from the ISP. This enables the telephone company to remove any and all Internet-related traffic from the backbone switching network, saving them millions in equipment and facility costs. This also opens new opportunities for data network providers.

Television and radio transmissions are sent over dedicated cables to their respective destinations, rather than through the switched network. This limits where the signals can be sent, but given the nature of the signals, this limitation is acceptable.

A hierarchy exists within the switching network to ensure the efficient use of trunk facilities and to provide alternate circuits in the event of failures or congestion. Before the divestiture of the Bell System in 1984, the hierarchy was much different than today. Remember that the Bell System provided both local and long-distance services, so they owned the entire network. After divestiture, this was changed. The Bell companies were no longer allowed to provide both local and long-distance services, so the network had to be split.

The old hierarchy provided five levels of switching offices, with class five offices designated as the end office or local office (see Figure 1.1). The class five office was capable of connecting to other end offices within its calling area, but relied on the class four office to connect to offices outside of its calling area. The calling area was not defined by area codes, but was a geographical service area drawn by the Bell System. Service access areas have since been reallocated as *Local Access Transport Areas* (LATAs) by the Justice Department as a result of the Bell System divestiture.

The class four office allowed the Bell System to aggregate its facilities and used high-capacity trunks to interconnect to other class four offices. In this way, the class five office did not require high-capacity facilities and handed off the bulk of its calls to the class four office. This also prevented the necessity for the class five office to have trunks to every other class five office in the service area. The class four offices were also known as toll centers.

As shown in Figure 1.1, the class four office provided two paths for a telephone connection. The interconnection of these various offices depended mostly on distance. There were many occurrences of a class five office connecting directly to a class one office.

The toll office searched for an available trunk as low in the hierarchy as possible. If one was not available, it would search for a trunk to a primary center in the destination calling area. If there were no available trunks to the primary center, then the last choice would be an overflow trunk to its own primary center.

The class three office, or primary center, was part of the toll network. This office connected to class two offices, or sectional offices, but also provided a path to other class

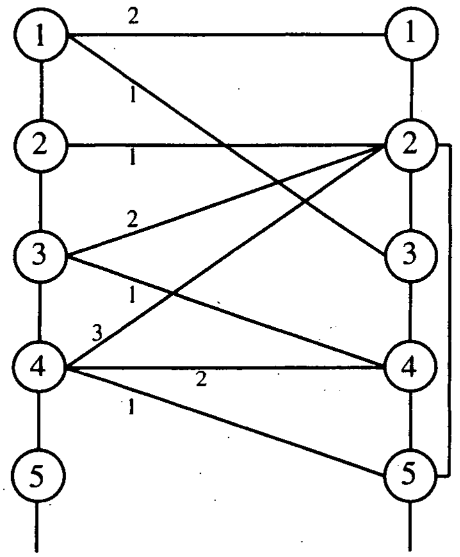


Figure 1.1 This figure illustrates the Bell System switching hierarchy prior to the divestiture of the Bell System. The priority of the routes is indicated by the numbers.

three offices and class four offices. This office served as an overflow-switching center in the event that other routes lower in the hierarchy were not available.

The class two office was also known as the sectional center and provided access to the regional center. Only two routes were available at this level, one to its peer in the destination calling area and one to the regional switching center, or class one office.

The class one office was known as the regional center and was used for toll calls. The regional center also provided access to the long-distance network. A typical toll call required an average of three trunks. The maximum number of trunks allowed in a connection was nine.

Postdivestiture Switching Hierarchy. In the mid-1980s, technology enabled many of the functions just described to be combined. As switching equipment was improved, systems were given the capability to function as local switches, tandems, and even toll switches. In addition to better routing functionality, these switches were also given the capability to record billing records and perform alternate routing in the event of congestion or failures.

The new hierarchy consists of fewer levels, consolidating many of the functions of the previous hierarchy into two or three layers (see Figure 1.2). Long-distance access is accomplished through a *point-of-presence* (POP) office. The long-distance carrier has a similar multilevel hierarchy, which may be several layers as well.

Packet-switched networks are much different. A hierarchy still exists in the packet-switched world, but there are fewer layers to deal with and, of course, no switching

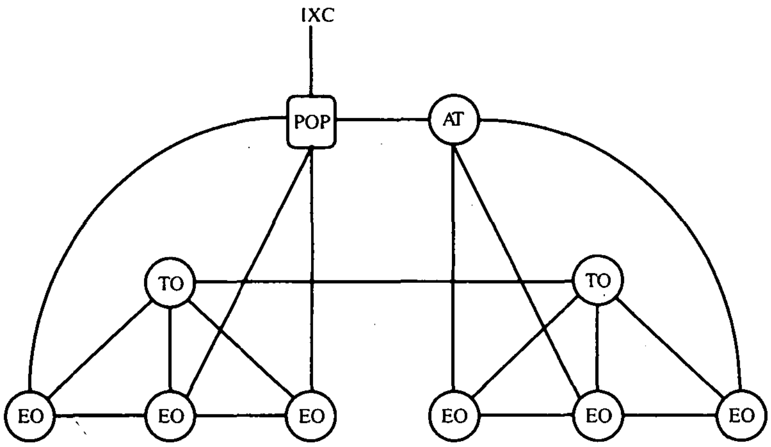


Figure 1.2 This figure illustrates a much flatter switching hierarchy, used after the divestiture of the Bell System.

within the network. Everything is transmitted over the same facilities. We will talk about packet-switched networks later on.

Local Access Transport Areas (LATAs)

After divestiture, the telephone companies' service areas (or exchanges, as they were sometimes called) were redrawn by the Justice Department so that telephone companies would have evenly divided service areas with equal revenue potential. These areas, called LATAs, were divided according to census information as well as the amount of capital already invested in each area by existing carriers in the form of network equipment. Each *Regional Bell Operating Company* (RBOC) and each independent telephone company received a service area or combination of areas (more than one LATA could be assigned to a carrier) that would provide it a fair and equal market. LATAs are smaller in most cases than the service areas carved out by the RBOCs, but still permit telephone companies to maintain their original network investments. There were originally 146 LATAs, but as changes take place in the networks, the number of LATAs is growing.

To maintain fairness between local and long-distance telephone companies, all traffic that crosses the boundaries of a LATA must be connected through a long-distance carrier. The local operating companies and independents are not allowed to carry traffic from one LATA to another. This ensures that the local telephone companies do not interfere with long-distance competition and provide the long-distance companies a fair and profitable boundary for which they can compete with other carriers.

The Telecommunications Act of 1996 enables long-distance companies to provide local telephone service in their markets only after they pass specific criteria. This reverses the legislation put into place by Judge Green when divestiture of the Bell System reshaped the nation's telephone industry and limited long-distance companies from offering local service.

At the same time, the Telecommunications Act of 1996 enables local telephone companies to offer long-distance (interLATA) service in their market areas. They must demonstrate that they have allowed competition in their market areas and pass criteria set in the Telecommunications Act before offering such service. The result of this sweeping new legislation is still unfolding, but many anticipate local telephone companies will partner and merge with long-distance companies, taking advantage of one another's markets.

To promote competition in long-distance service, the telephone companies were forced in the 1980s to provide long-distance carriers equal access into the telephone network. This was accomplished in the current switching hierarchy by establishing a POP, which serves as an interface to all interexchange carriers into the LATA. Every LATA must have one POP. The telephone companies collect access fees from the long-distance carriers for this interface into their network to offset the cost of equipment and ensure a revenue stream from long-distance traffic. They will also bill the long-distance carrier for all traffic that comes from the long-distance network and terminates in the local network. This has become an important source of revenue for local telephone companies.

The local telephone companies further divided each LATA into a local market and a toll market. The toll market is within the LATA (intraLATA), but is considered by the telephone company as a long-distance call because of the distance from one city to another or the distance between central offices handling the call. These toll calls are currently very expensive and have recently been open for competition. These are the only long-distance calls local telephone companies are allowed to provide. Many states have allowed long-distance carriers to compete in this market, opening up the LATA to competition.

Over 300 LATAs are presently defined throughout the United States. Throughout these LATAs are hundreds of telephone companies competing for revenues. Many of the smaller independent companies are investing capital to get connected to the SS7 network so that they can offer the same types of advanced calling features the larger service providers offer. These independent companies have joined forces, forming telephone associations across the nation. This allows them to represent their industry in standards committees and voice their concerns to the government as a collective body, rather than a lone voice. These associations also pool their resources and build their own networks using monies from the member companies to pay for the cost of the network. This is how many small independent companies are getting involved in SS7.

Who Are the Players?

A number of companies offer telephone services of some type today. To fully understand their business, we need to understand their roles in the network. Let's define what these companies do so we can understand the services they provide. Some of these companies, by the way, offer their services to other telephone companies and do not have any subscribers connecting to their network.

Regional Bell Operating Company (RBOC). These are the companies resulting from the divestiture of the BOC. Only a couple of these companies are left now due to mergers and acquisitions. Southwestern Bell (now known as SBC) has acquired *Southern New England Telephone* (SNET), Ameritech, and Pacific Bell, and continues to work on others. Bell Atlantic merged with NYNEX and acquired GTE to become Verizon. The RBOCs provide local telephone service, but as they open up their territories for competition (meeting the requirements of the Telecommunications Act of 1996), some are starting to provide long-distance service.

Local Exchange Carrier (LEC). Typically, these are the independent telephone companies offering local service where the RBOCs did not provide service. An example of a LEC is *General Telephone* (GTE). Many other LECs throughout the United States are considerably smaller in size. They continue to operate today as small, independent telephone companies, typically in the smaller, rural areas of America.

Incumbent Local Exchange Carrier (ILEC). The *Incumbent Local Exchange Carrier* (ILEC) is a local telephone company, usually one of the independents, not affiliated with any of the RBOCs. They fall under the same guidelines as the RBOCs and have been given territories like the RBOCs.

Competitive Local Exchange Carrier (CLEC). The *Competitive Local Exchange Carrier* (CLEC) is a new entrant into a service area. These are usually small startups offering local telephone service in a few LATAs. They typically start as resellers of long-distance service, and then begin investing in their own telephone network as they build their customer base. They target specific LATAs to offer service in and then expand out. Hundreds of CLECs exist in the United States today, and many more are starting. With new packet-switching networks rolling out, the number of CLECs is increasing sharply, due to the decrease in network deployment costs associated with packet switching.

Interexchange Carrier (IXC). These are the long-distance carriers providing long-distance service between LATAs. They started out as long-distance service providers in specific areas, but now many of them have expanded their markets by offering local telephone service. AT&T, Global Crossing, and Sprint are all examples of *interexchange carriers* (IXCs). It is also important to note that these IXCs also provide services to other carriers and may also be classified as a CLEC (if they are providing local telephone service).

Data Competitive Local Exchange Carrier (CLEC). You don't hear much about data CLECs, but this is big business. These companies take the Internet and data traffic from the local telephone companies and route it through their own packet-switched networks. The RBOCs are big customers of data CLECs because they do not want to manage this data traffic in their own networks. ISPs are also big users of these networks because they save them from having to build huge, nationwide networks to provide service in every city. They can use a data CLEC in every city, route the traffic through these networks to one national backbone network, and save the equipment and facility costs associated with nationwide network buildouts.

Hub Providers. Another emerging business is the hub provider business. These are carriers who provide facilities and interconnections to other telephone companies. For small telephone companies entering into a new market, this is usually the most economical way to get started because there is little capital investment for them to make. They simply lease the services they need from the hub provider.

A hub provider provides switching and circuits to other networks for an access fee. They will also provide database services such as *calling name* (CNAM) and 800 applications. They collect fees from the networks they connect to based on the amount of traffic they send into other networks as well as the amount of traffic they terminate in their own networks. Several hub providers provide SS7 services in the United States today, but the concept is still new in Europe and the rest of the world.

Many of the RBOCs have entered into the hub provider business themselves, recognizing this as a new source of revenue. The market has become very competitive for hub providers, forcing a reduction in access fees and other fees collected.

Hierarchy of the Synchronization Network

All digital networks rely on timing mechanisms to maintain integrity of the data transmission. Because all digital transmissions are multiplexed and based on time division, accurate timing is critical. This is especially true when DS0A links are used in SS7. DS0A links must have accurate timing sources in order for them to synchronize and carry signaling traffic.

Digital facilities must have reliable, accurate clock sources to determine proper bit timing. These clocks must be synchronized with the same source and are deployed throughout the telephone network. To maintain timing in the telephone network, a separate synchronization network has been defined (see Figure 1.3).

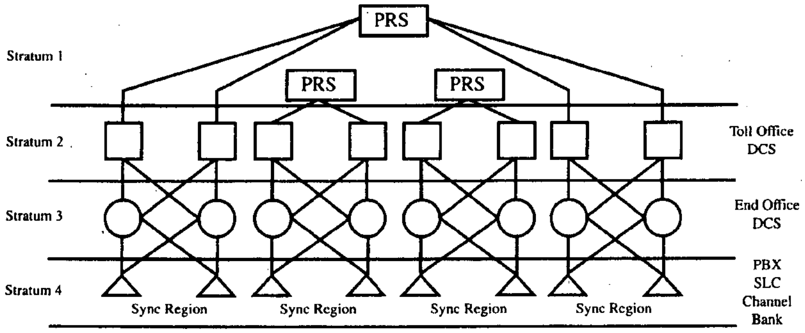


Figure 1.3 To maintain synchronized timing throughout the digital network, the BOCs use this timing network.

The source for a clock signal is referred to as the *primary reference source* (PRS). These clock sources reside in the various regions of the telephone network. They are highly accurate clocks, usually cesium beam or rubidium-based clocks. These clocks must be resynchronized and continuously verified using a universal time source. Many companies currently use Loran-C and the *Global Positioning System* (GPS) to check the accuracy of their clocks. The distribution of clock signals is implemented at different levels, referred to as *strata*. The highest stratum obtainable is stratum 1, which is the primary clock source (primarily meaning it is referenced directly from Loran or GPS sources).

Clock signals are distributed in a primary/secondary relationship to all other levels. This means that central office switching equipment at stratum 2 distributes its clock signal to equipment at stratum 3. Equipment that sits downstream of the clock signal will not receive as accurate a clock signal as those connected directly to the source (stratum 1).

The PRS distributes clocking signals to toll offices. The toll office is considered stratum 2 and must redistribute the clock to end offices within its LATA. Each LATA must have at least one stratum 2 clock. Whenever a clock is redistributed, it loses some accuracy. Yet the clock signal is accurate enough to operate throughout the network reliably, despite the loss of accuracy. End offices are considered stratum 3.

The end office will distribute clock signals to other users of digital transmission facilities, such as *private branch exchanges* (PBXs) and channel banks. These are considered to be at stratum 4. In some cases, devices at this level can redistribute the clock signal to other adjunct equipment. These devices are considered to be at the lowest level of the hierarchy, stratum 5.

Within a central office, clocks are distributed through a *building integrated timing system* (BITS). BITS is a distribution system for clock signals and is distributed throughout the office to switching equipment in that office. BITS is critical to the proper operation of DS0A links in the SS7 network. Failure of this clock signal will result in the failure of the signaling links.

Digital Signaling Hierarchy

The telephone network also has a digital hierarchy for all digital transmission facilities. This digital hierarchy is a means for expressing the capacity of these various facilities. Typically, the highest level of the hierarchy is an aggregate of the levels below it. Thus, the lowest digital signal in the hierarchy is multiplied to establish the next level, which is multiplied to obtain the next level, and so forth. Table 1.1 illustrates the correlations between levels in the hierarchy.

The SS7 network uses DS0As for signaling links. This is a 56 or 64 kbps data link, capable of sending voice or data. The DS0A in the United States is always 56 kbps because the telephone company uses 8 kbps for control information. This control information is used by the transmission equipment to maintain the integrity of the data link. Some studies have been underway regarding the use of 64 kbps, but the multiplexers used throughout the telephone network today do not support 64 kbps links.

TABLE 1.1 North American Hierarchy

Digital signal destination	Bandwidth	Channels (DSOs)	Carrier designation
DS0	64 kbps	1 channel	None
DS1	1.544 Mbps	24 channels	T1
DS1C	3.152 Mbps	48 channels	T1c
DS2	6.312 Mbps	96 channels	T2
DS3	44.736 Mbps	672 channels	T3
DS4	274.176 Mbps	4,032 channels	T4

In order for equipment to use the DS0A, special digital interfaces that are capable of sending and receiving signals at this level must be installed. If a DS1 is used, a device called a channel bank must be used as the interface. The channel bank divides the 24 time slots of the DS1 into 24 separate DS0As, which can then be distributed to their proper destinations. Other types of multiplexers/demultiplexers exist that perform the same task, depending on the location in the network.

Currently, the DS0A level is the most commonly used in the SS7 network. As *asynchronous transfer mode* (ATM) is deployed in the public networks, signaling traffic is migrating to these new facilities. The current digital facilities are being replaced, and the signaling network is becoming integrated with the new broadband network. The first step is to support DS1 speeds over an ATM interface. This is currently deployed in some larger networks and is gaining acceptance in some smaller networks as well.

Originally, it was thought that ATM would be the technology of choice for telephone companies migrating to broadband networks. However, TCP/IP has most recently entered into the picture as a more efficient and cost-effective technology for signaling and, in some cases, voice and data. Of course, TCP/IP can be packetized and sent over ATM backbone networks, which seems to be the choice for many network operators today.

As shown in Table 1.1, the DS1 facility provides a total bandwidth of 1.544 Mbps. New TCP/IP facilities provide 100 Mbps of bandwidth and are being deployed around the world as the next-generation network choice. The existing network will require new hardware as well as software upgrades to support these new technologies. The SS7 protocol is being modified to adapt to the TCP/IP network as well, with new transport protocols replacing the current *Message Transfer Part* (MTP) used today.

Fiber optics is also playing a significant role in the evolution of the telecommunications network. Fiber optics has the capability to transmit at much higher data rates than copper and is critical to the success of technologies such as broadband and ATM.

Synchronous Optical NETwork (SONET) is currently found in telephone company networks worldwide and has become the transmission medium of choice. SONET provides data rates of up to 2.4 Gbps and will support broadband ISDN, TCP/IP, and ATM. SONET is also being used to link *local area networks* (LANs) through the PSTN.

As seen in Table 1.2, SONET is also divided into different levels of service, each level being an aggregate of the levels below it. Two designations are used for these levels: the electrical signal itself and the optical signal. They are terms used for different reasons.

Electrical signals are directly related to the optical signals and therefore can almost be used synonymously in most discussions. In this book, we will always refer to the optical signal.

When compared to the digital signal hierarchy, there is a stark difference (see Table 1.3). Even at the lowest level of the optical hierarchy, 28 DS1s can be supported on one facility. This represents a significant cost savings to telephone companies. At OC-1, 672 time slots are supported for voice, data, or even signaling.

A single SONET facility is not dedicated entirely to one application. One channel may be used for signaling, while the remainder carry voice, data, and video. This practice enables telephone companies to use existing transmission facilities between offices, rather than deploying a special link just for SS7.

Current Trends in Telecommunications Technology

Today's telecommunications industry has changed dramatically. The Internet has provided a new communications vehicle and spawned new services. To support these many new services, the signaling network must realize changes as well.

The signaling network has become more than just a services enabler. Because of the rich information provided in the signaling network, SS7 has become a rich source of data used by all departments within a carrier. Subscriber profiles, usage measurements

TABLE 1.2 SONET Digital Hierarchy

Electrical signal	Optical signal	Data rate (Mbps)	ITU designation
STS-1	OC-1	51.84	
STS-3	OC-3	155.52	STM-1
STS-9	OC-9	466.56	STM-3
STS-12	OC-12	622.08	STM-4
STS-18	OC-18	933.12	STM-6
STS-24	OC-24	1244.16	STM-8
STS-36	OC-36	1866.24	STM-12
STS-48	OC-48	2488.32	STM-16

TABLE 1.3 Optical and Digital Compared

Digital signal	Optical signal
DS0 (64 Kbps)	OC-1 (51.84 Mbps)
DS1 (1.544 Kbps)	OC-3 (155.52 Mbps)
DS1C (3.152 Kbps)	OC-9 (466.56 Mbps)
DS2 (6.312 Kbps)	OC-12 (622.08 Mbps)
DS3 (44.736 Kbps)	OC-18 (933.12 Mbps)
DS4 (274.176 Kbps)	OC-24 (1244.16 Mbps)
	OC-36 (1866.24 Mbps)
	OC-48 (2488.32 Mbps)

used to verify billing, and traffic analysis are a few examples of how signaling information is being used.

The architecture of an *Intelligent Network* (IN) has changed somewhat, although the concept remains the same. The concept of an IN is to move the service logic out of the switching platforms and onto platform-agnostic servers. The switches can then determine how individual calls are to be handled by accessing these servers and receiving instructions. Convergent networks that depend on TCP/IP for the transport of voice and data follow this model.

The ultimate goal is to provide one network capable of transferring all kinds of information regardless of the bandwidth necessary and sending it through the network just as if it were placing a telephone call. To support this level of service, the network must be changed.

Again, TCP/IP has quickly become the technology of choice for these networks. There are several reasons for this development. First, TCP/IP is widely available. Virtually every major corporation using e-mail of some type is connected to a TCP/IP network, usually their own.

Second, TCP/IP is a proven technology. Originally developed for use by the Defense Department for their data networks, TCP/IP has quickly migrated into the private sector and gained widespread popularity because of the Internet. However, TCP/IP is not well suited for real-time applications such as voice transmission and must undergo changes to support voice transmission without noticeable delays in the transmission.

This change is underway today through many of the standards organizations. In the meantime, many telephone companies have relied on ATM to provide *quality of service* (QoS) for voice transmission. A number of new companies are emerging who use TCP/IP as their primary network for all traffic. These companies are driving a new and fast-growing market.

As a result, a number of new equipment vendors have found themselves in the telephony business. They are building a new generation of switches that are based on computer platforms rather than the conventional switching platforms used in older legacy networks. This new breed of switches enables small companies to enter the telephone business and deploy their own networks quickly and a lot cheaper than conventional telephone companies.

The IN is also evolving to support TCP/IP. The same vendors who developed the legacy computer systems used in the IN are quickly adding TCP/IP interfaces to support access to their products through both the Internet and intranets. SS7 itself is being adapted to meet the demands of this fast-growing market.

Introduction to the Intelligent Network

The 1990s will be known as the years of the Clinton administration's Information Highway. The fact is that the Information Highway was already underway before this time. The IN has been under development for many years, with the goal of allowing all types of information to pass through the telephone network without special circuits or long installation cycles.

The problem facing telephone companies today is being able to provide these features and services quickly and efficiently. Ordering an 800 line for two weeks' usage is now easy and can be implemented within hours instead of days. The IN makes it easier, because now, when subscribers order new services, technicians do not have to be dispatched to add programming to the switching equipment and to cross-connect the circuits.

In the IN, everything is controlled or configured by workstations with user-friendly software interfaces. Telephone service representatives can create new services and tailor a subscriber's service from the terminal while talking with the customer. The changes are immediately implemented in the switches. Circuits are cross-connected using digital cross-connect systems, which are also controlled by the workstation. Customers today can order high-speed communications, video, audio, and digital voice facilities on an as-needed basis.

Networks were not always equipped to handle such demand. But switch manufacturers have added new features to switching equipment that enable services to be added to subscriber lines by simple commands at a terminal. Some new products even enable customers to order services and features by dialing a sequence of codes on their telephones. Soon ordering an 800 line will be as simple and as fast as ordering a pizza.

Welcome to the age of the IN. The IN is just what its name implies: intelligent. Services and features can be changed or deployed using simple procedures through a terminal, rather than through expensive programming changes made by certified technicians. All the customer needs is the facility (trunks) to utilize the new services. With TCP/IP, the customer needs only enough bandwidth to handle the traffic.

Imagine a small business with fewer than 100 employees. By building a TCP/IP network and using computers equipped for Internet telephony, this company can interconnect all of its employees over the company's computer network. Now imagine if that same company were to extend its computer network out to the local telephone company. Suddenly, everyone connected to the local telephone company could access this small business through the same computer network.

If the local telephone company were to extend this same network to other telephone companies, the small business would suddenly find itself connected to a huge computer network, with access to thousands of computers equipped to send not only data, but voice as well. This is an oversimplified explanation of computer telephony, but the basic concept has changed the telephone industry overnight.

Still, computer networks do not provide the telephone services we depend on daily. This is where the IN comes into play. The IN can be deployed into this same computer network and used to deliver services and features to computer telephony devices.

As more and more customers line up to deploy ATM and TCP/IP, the IN will become as commonplace as Touch Tone™ dials. Yet little is understood about the IN. To understand what the IN is about, let us first examine the architecture of such a network. The IN consists of a series of intelligent nodes, each capable of processing at various levels, and each capable of communicating with one another over data links.

The IN relies on the SS7 network, which forms its backbone. SS7 provides the basic infrastructure needed for the *service switching point* (SSP), which provides the local access as well as an ISDN interface; for the *signaling transfer point* (STP), which provides packet switching of message-based signaling protocols for use in the IN; and for the *service control point* (SCP), which provides the service logic to be used in determining how to handle calls. The SCP is connected to a *Service Management System* (SMS), which provides a human interface to these databases. The SMS uses a command-line interface or a *graphic user interface* (GUI) interface and a man-to-machine language to build services and manage the network. The SMS can also be used in some applications as a central control point for updating multiple databases and controlling the updates to those databases from a central authority.

One additional node used in the IN that is not seen in the SS7 architecture is the *Intelligent Peripheral* (IP). The IP provides resource management of devices such as voice response units, voice announcers, and DTMF sensors for caller-activated services. The SCP accesses the IP when services demand its interaction. IPs provide the IN with the functionality to enable customers to define their network needs themselves without the use of telephone company personnel.

When a call is placed in the IN, a request for call-handling instructions is sent to the SCP using the *Transaction Capabilities Application Part* (TCAP) protocol. The database provides the instructions for handling the call based upon the customized service instructions the subscriber has programmed and sends them to the end-office switch. The end-office switch then communicates to the IP using the ISDN protocol to attain the use of resources such as recordings and other devices. The call setup and teardown is handled using conventional SS7 protocols.

Advanced Intelligent Networks (AINs) in Figure 1.4 provide many components not found in the earlier versions of INs. One of the key components is the *service creation environment* (SCE). In the AIN standard, SCE defines the look and feel of the software used to program end-office switches to provide a new service. This look and feel defined in the AIN standard provides a GUI, which uses icons, for building customized services. AIN administrators can then tailor services to meet the customers' specific needs by clicking network capability icons rather than programming via commands on a command line.

Eventually, this terminal and the SCE will be extended to the customer premises, enabling large and small companies with special network needs to tailor their services on an as-needed basis, without telephone company assistance. This concept has already intrigued many large companies with large volumes of inbound and outbound calls, especially those in the telemarketing industry whose network needs vary from week to week.

The first uses of an IN were in the early 1980s when AT&T deployed a centralized database for all 800 numbers. End offices wishing to handle 800 calls would have to access this database through the SS7 network.

In addition to 800 services, AT&T launched a calling card application using centralized databases to qualify calling cards as valid or invalid. Control of the network was provided by *operations support systems* (OSSs), which used intelligent workstations to provide maintenance and administration for the network databases and other network nodes.

In 1988, AT&T asked vendors to participate in defining the standards for the evolving IN. The end result was the release of an evolutionary path defined in a series of re-

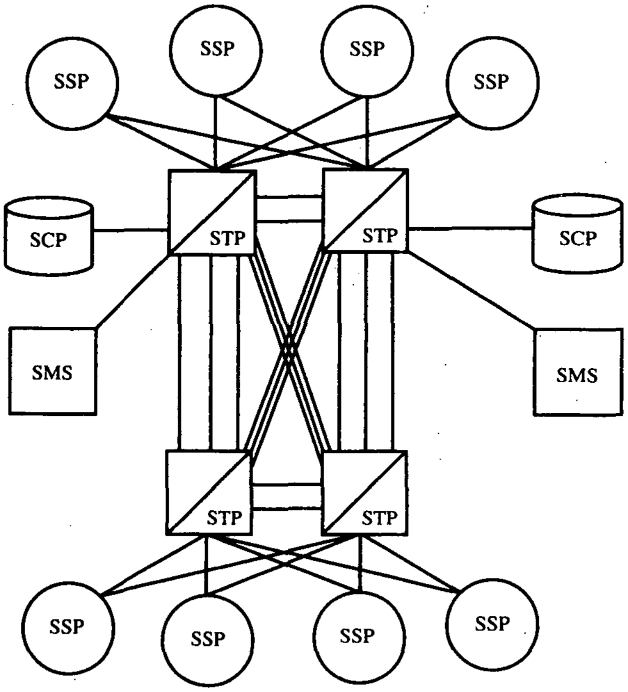


Figure 1.4 The AIN relies on SS7 for the interconnection of network switches.

leases, now known as the AIN. The AIN defines how the telephone company or the subscriber will invoke features.

AIN offers a variety of services to the consumer, such as redirecting the destination of calls on a per-line basis. This can be altered by dialing a code through a phone and entering in the destination telephone number. This feature would be of huge value to marketing firms with large inbound call volumes. Eventually, the subscriber, through a terminal located at the customer premises and connected to the telephone company's signaling network, will deploy the feature.

Other features currently available include call screening (Do Not Disturb), selective call acceptance, calling name delivery, and spoken caller identification. The many services and features offered to customers are based on databases linked to the SS7 network through SCPs. Local end offices and other networks access these databases by sending TCAP database query messages through the SS7 network to the SCP. The SCP replies to the query after accessing the information from the appropriate database and sending the requested information in an SS7 message format through the SS7 network to the requesting end office. Based on the information received, the end office (or service node) is then able to create the requested services.

The SCP provides the call-handling instructions and service instructions to the end office so that it knows how to handle calls for a specific subscriber. The following features are just a sampling of the features of an IN. As the IN evolves, new services will become available:

- Find Me service
- Follow Me service
- Computer security service
- Call pickup service
- Store locator service
- Call routing service
- Multilocation extension dialing
- Name delivery
- Outgoing call restriction

Find Me Service. This service enables calls to be forwarded to another location. The difference between this feature and today's call-forwarding feature is the capability to screen unwanted calls from forwarding. Only authorized callers are forwarded to the new location.

Follow Me Service. Similar to call forwarding, this enables a number to be forwarded on a time schedule. The subscriber determines the time forwarding is to take place when the feature is invoked. Destinations can include cellular telephones or *Personal Communications Services* (PCS) handsets.

Computer Security Service. This feature prevents unauthorized callers from accessing a computer via modem. Only callers with the authorized access code or calling from an authorized number can access the computer. The SS7 network delivers the calling party number to the destination end office. This number is then checked in a database located with a SCP and, if authorized, is allowed to connect with the modem.

Call Pickup Service. When a call is placed to a number and is unanswered, the called party can be paged via radio pager. The called party can then dial a code from any telephone at any location and immediately be connected with the waiting caller. Some manufacturers have already developed two-way pagers that connect the caller with the party being paged. The pager is a two-way transceiver capable of receiving calls (pages) and connecting the caller with the paged party (similar to the voice pagers used a few years back, but this pager enables two-way conversation).

Store Locator Service. Businesses can advertise one number, and callers are automatically transferred to the nearest location based on their own telephone number. The telephone company provides the routing service based on the prefix of the calling party number. This enables businesses to advertise nationwide for all locations without special ads based on geography. The calling party number is matched in a routing database located at an SCP. The SCP provides the end office with the routing instructions based on the calling party number.

Call Routing Service. This enables businesses to reroute calls when congestion occurs or after business hours. It is an excellent feature for telemarketing and reservation centers with multiple locations. *Automatic call distribution* (ACD) switches can be interfaced to the end office by SS7 data links. This enables the ACD to send network management messages via SS7 protocols to a database, located by an SCP. Calls are then rerouted around the call center based on the routing instructions in the database.

Multilocation Extension Dialing. This enables the usage of abbreviated extension numbers to reach personnel regardless of their location and without PBX equipment. Subscribers receive personal numbers that can be used to reach them no matter where they go.

Name Delivery. As a call rings the telephone, the caller's name is displayed on a digital display. This is offered to residential and business customers alike. The digital display is built into a digital phone or is an adjunct to any standard telephone. This is somewhat different than the controversial *Automatic Number Identification* (ANI) feature, which displays the caller's telephone number. ANI enables telemarketing companies to store calling party numbers in their databases, which are later sold to other telemarketing companies. Name delivery delivers the name only, retrieved from a line subscriber database.

Outgoing Call Restriction. This feature enables the restriction of specific numbers or prefixes and area codes, allowing customers to restrict long-distance calls and service numbers such as 900 and 976 numbers from being dialed on their phones. The subscriber can enter telephone numbers to be blocked from their telephone keypad.

It is important to note that AIN does not define the features and services, but how those features and services will be deployed. The features and services are defined by the service providers themselves, and may or may not be consistent from one telephone company to another. They are limited, however, to the capabilities of the equipment used in their networks.

Many telephone companies are reluctant to deploy AIN because of rumors of a newer network called *Information Network Architecture* (INA). This new technology is still in development, but many feel it will succeed AIN. Yet others view INA as a subset of AIN. At any rate, INA provides better utilities for managing the new broadband services being deployed by major telephone companies. It is very likely that there will be two architectures: AIN for the voice network and INA for managing the broadband network.

The AIN is certainly not a new concept. As we have already seen, this concept evolved from an earlier introduction to make 800 services easier to customize and maintain.

The capability to create new services quickly and efficiently was one of the early driving forces behind the IN. In fact, 800 services today can be customized by the customer according to their immediate needs with a simple phone call. AIN has not been deployed for local telephone service, however, and that is the undertaking today.

Wireless and Centrex also benefit from the IN. Wireless is beginning to support services such as SMS, which relies on SS7 for delivery. Centrex customers can tailor their specific service requirements within hours instead of days. With features such as networked voice mail and ACD, Centrex will become a powerful competitor to the PBX.

Presently, Centrex is difficult to market against the PBX because of the limited feature set offered with Centrex. With PBX systems, proprietary phones can offer a multitude of high-tech features that greatly enhance the systems' capabilities.

When Centrex is offered with IN support, subscribers can take advantage of the telephone network to provide them with seamless end-to-end call-handling capabilities. Callers can be routed to noncongested calling centers dependent on traffic and/or time of day. Voice mail can be linked so that users can reach their voice mail from any phone at any company location, without dialing into the system from an outside line. Even *station detailed message recording* (SMDR), the feature that provides the calling records of every extension in the system, can be bridged to include all extensions in the corporation, rather than just those within a specific office handled by one carrier. The IN will enable Centrex to provide many of the same features previously seen only in the PBX environment.

The Integrated Services Digital Network (ISDN)

The term ISDN was originally used to refer to the entire IN, including SS7. Originally, the creators of SS7 thought of extending the SS7 network all the way to the subscriber. This was abandoned, however, over concern for security and network fraud, as discussed later. The solution was to create an intelligent interface, compatible with SS7, which could offer the same services and intelligence as the SS7 network to subscribers (see Figure 1.5). It was this that spurred the creation of the ISDN protocol.

Perhaps the most important application for ISDN is the concept of connecting PBXs within a private network. When SS6 was first deployed, there was the thought that the network signaling could be extended to the local PBX. This would enable the PBX to send its signaling information directly to a central office switch using the same message packet-switching protocol used by SS7 today.

This concept was quickly dropped, however, due to security issues. Instead, a separate access protocol was developed. With a specialized access protocol, signaling could be extended through the PSTN to distant PBXs, without sacrificing security of the PSTN. ISDN was created as the access protocol to deliver PBX signaling through the network to distant PBXs, enabling large companies with multiple PBXs to bridge their switches together transparently. The United Kingdom already uses the *Digital Private Network Signaling System* (DPNSS), an ISDN protocol designed to extend PBX signaling through the SS7 network to distant PBXs.

ISDN offers many services to the subscriber. The basic levels of service are defined as the following:

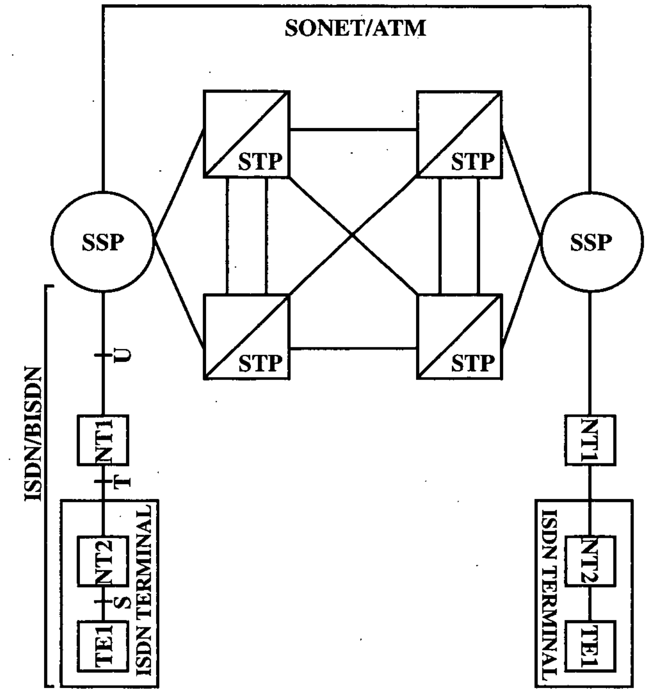


Figure 1.5 ISDN and BISDN extend the services of the IN out to the subscriber premises. They both act as interfaces to the signaling network (SS7) without providing direct links.

- *Transport elements.* Enable information to be transported through the telephone service provider's network and its switches, routers, multiplexers, and other network equipment transparently, without alteration to the original data.
- *Control elements.* Support real-time operations of transport capabilities (connection establishment and database queries).
- *Network management elements.* Provide procedures and capabilities to administer, maintain, and operate the communications infrastructure. Includes the provisioning of transmission facilities, fault management, congestion control, and administration of databases and routing tables.
- *Communications applications environment.* Provides a development environment for programmers from which applications can be developed, using the other three elements.
- *Transport.* Provides the lower three layers of the *Open Systems Interconnection* (OSI), providing the allocation of bandwidth, routing, relaying, and error detection/correction.

To understand how ISDN can be a significant benefit to PBX networks, consider this example. Many large corporations own several PBXs at different locations. Tie lines are often used to tie the PBXs together. This enables users to access extensions in any other company location by dialing an access number (to access the tie line) and dialing the extension. Many digital PBXs enable callers to dial extension numbers of remote extensions without dialing an access code. Automatic routing features provide software to determine which trunk the call must be routed to.

Another advantage of tying PBXs together is that long-distance calls can be routed over tie lines to a PBX in the calling area of the dialed number. The call is then routed to a trunk in the remote PBX as a local call. This can save corporations thousands of dollars in long-distance charges. The problem is that the remote PBX does not know what class of service the calling phone has been assigned. The class of service is a software feature that determines what numbers a phone is allowed to dial. This programming takes place in the PBX terminating the telephone.

ISDN enables this information to be passed along to the remote PBX. In addition, other information regarding the privileges of a telephone and even features can be passed from one PBX to another. This enables corporations to create their own proprietary network, without expensive facilities between PBX locations.

ISDN never got a good start and has not shared the popularity of other service offerings, mostly because it was somewhat premature in its release. Many telephone service providers tried to market this service to residence subscribers, but failed because of the prohibitive cost and lack of intrinsic value. To truly take advantage of ISDN capabilities, the signaling information must be able to travel from the originating end to the distant end. Without SS7, this is not possible. When ISDN was first introduced, SS7 was not yet fully deployed, leaving "islands" of ISDN service that could not be connected with other ISDN networks.

Now with SS7 deployed in all the BOCs' networks, as well as those of many of the independent telephone companies, ISDN can finally be used to its full potential. Unfortunately, the telephone companies still do not understand how to market ISDN and often cite the many features ISDN provides, rather than real applications.

One feature often used in marketing ISDN is ANI. For a short period, there was a lot of interest in ANI, as telemarketing companies scrambled for databases that could provide names and addresses of callers triggered by the calling party information supplied by the ISDN network. All interest in ANI was quickly lost as the states began legislation blocking ANI, or at least forcing telephone companies to provide ANI blocking at no charge. Without the guarantee that at least a majority of the calls would provide ANI information, and without any other substantial advantages in the way of features, ISDN quickly faded into the background.

The problem was that most areas could not provide the calling party information. This was due to the fact that many central offices had not yet been upgraded to digital switches and were not capable of passing this information through the SS7 network. By the time networks caught up with ISDN and the SS7 network bridged all the "islands," digital switch vendors began offering features in their central office switches that would support not only the delivery of the calling parties' telephone number, but their name as well, without having an ISDN interface. *Calling Name* (CNAM) continues to be a popular feature today, possibly because of the SS7 network.

Another plague of ISDN has been the failure of manufacturers to follow a standard. Unfortunately, many manufacturers created ISDN terminals and devices that were proprietary and did not share the concept of open interconnection, making interoperability difficult, if not impossible.

The *North American ISDN User's Forum* (NIUF) later addressed this in June 1988. The NIUF comprises industry vendors and service providers. This forum agreed on a set of features and how they would be deployed in the ISDN network, and ratified a new ISDN standard, now known as National ISDN-1. This standard is an agreement among ISDN vendors here in the United States to standardize the equipment interfaces, interoperability, and features, guaranteeing ISDN's success in North America.

Still, ISDN continues to be marketed for its features rather than for true applications. What ISDN really offers to any business with a PBX is the ability to consolidate its trunking requirements to one or more digital spans, or T-1s. Using the ISDN protocols, they can take advantage of end-to-end digital communications for both voice and data. This allows owners of PBX equipment to rid themselves of costly dedicated lines for data communications, facsimile, and video conferencing. ISDN can provide all these services and more, with a common facility, eliminating the need for special circuits. In addition, these same facilities can be assigned dynamically on an as-needed basis.

Digital Subscriber Line (DSL) has had a big impact on ISDN, providing bandwidth (hence higher speeds) to residential customers looking for high-speed Internet access. DSL is cheaper and simpler to deploy for small companies and residential customers. Larger corporations with higher bandwidth requirements will find *broadband ISDN* (BISDN) more attractive, with enough bandwidth to support Ethernet, Token Ring, and even *Fiber Distributed Data Interface* (FDDI). BISDN will provide bandwidth up to 155 Mbps and higher.

There are two classes of ISDN service: *basic rate* (BRI) and *primary rate* (PRI). BRI service provides two 64 kbps bearer channels (B channels) and one 16 kbps signaling channel (D channel). This service is designed for residential and small business usage. PRI offers 23 64 kbps B channels and one 64 kbps D channel. PRI is designed for larger businesses with large call volumes. Many PBX manufacturers already provide ISDN-compatible trunking interfaces for their equipment, making ISDN a good choice for companies who need end-to-end voice and data communications.

ISDN cannot be successful by itself. As we have already seen, without SS7, ISDN remains a local digital service providing a limited number of features and applications. With the addition of SS7, ISDN can become an extension of the telephone network to the customer premises, offering true end-to-end voice and data communications with no boundaries.

The ISDN standards can be found in the *International Telecommunications Union—Telecommunication Standardization Sector* (ITU-TS) "I" series. The signaling standards are defined in publication Q.921 (which defines the link access procedure—D channel) and the Q.931 publication (which defines the ISDN call control procedures at layer 3).

The Wireless Network and SS7

Wireless networks have evolved into several different network technologies. The differences between the various wireless networks lie primarily in the frequency allocation as well as the services supported. The trend is to support data and voice using wireless phones. There have been previous attempts to use technologies such as *cellular digital packet data* (CDPD), but the industry today is moving towards support of TCP/IP.

There are many obstacles to supporting Internet access on wireless phones. The bandwidth required to support Web access is not yet available in North America (although Japan and many other countries are already enjoying this level of access). Carriers throughout North America and Europe are phasing in new technology that will support this type of access. You will hear talk of *second-generation* (2G), 2.5G, and *third-generation* (3G) wireless. These all refer to the network changes required to provide the extra bandwidth and features, as well as the protocols needed to support these networks.

In this book, we will focus on the role SS7 plays in these networks, rather than the various network technologies themselves. Many North American wireless networks began deploying SS7 in the late 1980s and early 1990s, replacing their X.25 networks. The networks use SS7 somewhat differently than their wireline counterparts because of the way calls are originated and terminated in the wireless network. IS-41 is used for the origination and termination of calls involving wireless subscribers. This protocol uses the SS7 TCAP for transport. The ISUP is used to connect calls to the wireline network.

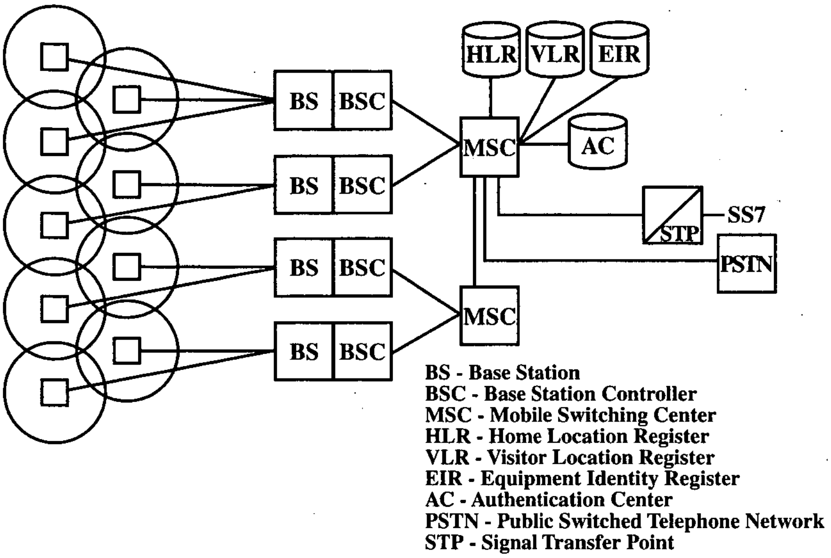
In Europe (as well as several other countries), *Groupe System Mobile* (GSM) is used instead of IS-41. The GSM set of protocols also relies on SS7 TCAP for service delivery as well as ISUP for connecting calls into the wireline network. One other common use for SS7 in both network types is SMS, which uses TCAP and the SS7 network to deliver short text messages to both wireless phones and pagers.

The GSM Network. European wireless networks have always relied on SS7 for their signaling requirements and have enjoyed a much more robust and feature-rich network because of SS7. GSM is deployed in several networks in North America as well, but has not shared the same popularity as in Europe. It is possible now to purchase a GSM wireless telephone in the United States. Provided you have a service agreement that supports usage in Europe, you can use the same telephone when you travel (see Figure 1.6).

The GSM wireless network comprises two different segments: the radio segment and the switching segment. The radio segment consists of the wireless telephone itself, or transceiver, and the antenna system used to receive and aggregate signals within a geographical location. This antenna is referred to as the cell site.

Cell sites use different types of antennas, depending on the coverage required. Omnidirectional antennas provide coverage in a circular pattern, radiating in all directions from the cell site. Directional antennas can be used to cover a specific sector and normally cover an area of 120 degrees.

All cell sites consist of the antenna and the radio transceiver, as well as interface equipment. This equipment is referred to as the *base transceiver station* (BTS) and the



BS - Base Station
BSC - Base Station Controller
MSC - Mobile Switching Center
HLR - Home Location Register
VLR - Visitor Location Register
EIR - Equipment Identity Register
AC - Authentication Center
PSTN - Public Switched Telephone Network
STP - Signal Transfer Point

Figure 1.6 The GSM network is the standard for cellular networks outside of the United States. Many U.S. cellular providers are now looking at GSM for new PCS networks.

base station controller (BSC). The combination of both the BTS and the BSC is called the *base station subsystem* (BSS).

The BTS is actually the radio transceiver used to communicate with the wireless telephone. When a caller is connected with a cell site, the BTS measures the strength of a *supervisory audio tone* (SAT), which is sent at a much higher frequency than the actual voice transmission. The SAT is sent at regular intervals by wireless telephones and is received by multiple cell sites. Each cell site reports the strength of the signal it receives to the *mobile switching center* (MSC), which then determines the cell site that will take possession of the call. If a call is in progress and the wireless phone moves into a new cell site coverage area, the call is handed off by signaling messages through the MSC to the new cell site.

The hand-off is controlled by the MSC, which communicates with all cell sites within its geographic area. The MSC does not communicate directly with the BTS, but communicates with the BSC, which serves as an interface between the radio segment and the switching segment. The BSC uses digital facilities to communicate with the MSC.

The interface between the MSC and the BSC is called the A-bis interface. The A-bis interface is a 64 kbps digital link and uses four protocols to transport signaling information to the MSC:

- *Link Access Procedure on the D Channel* (LAPD)
- *Base Transceiver Station Management* (BTSM)

- *A-bis Operations and Maintenance* (ABOM)
- *Direct Transfer Application Part* (DTAP)

The LAPD protocol is used as the layer 2 transport protocol and provides the node-to-node communications necessary to send packets through the network. The BTSM protocol is used for managing the radio equipment resident at the base station as well as the interface between the base station and the MSC. Data and other signaling information are sent from the base station equipment via the DTAP.

In addition to the PSTN, the MSC must also interface to other entities within the wireless network. These entities include the following:

- HLR
- *Visitor Location Register* (VLR)
- *Operation and Maintenance Center* (OMC)

The HLR is a database used to store the subscriber information for all subscribers within the home service area of the service provider. In European GSM networks, the HLR is linked to other service areas so that subscriber information may be shared between networks. Networks in North America can now offer the same seamless roaming through interconnection agreements with other wireless providers. It has only been the last four years or so that this has been possible in North America.

The VLR is used to store information about visiting subscribers who are not in their home service area. This is where the roaming number information gets stored, so that subscribers may use their wireless phones while in another city. This is also linked to other networks, so that this information may be shared. While subscribers are in this network, the information regarding their service will remain in the VLR. This information is retrieved from the home HLR by the network.

SS7 protocols are used throughout the wireless network to provide the signaling information required to establish and disconnect circuit connections, as well as share database information from one entity to another. In addition to the MTP and the *Signaling Connection Control Part* (SCCP), the following protocols are used from the MSC to other entities within the network:

- Mobile Application Part (MAP)
- Base Station Subsystem Mobile Application Part (BSSMAP)
- DTAP
- TCAP

Wireless Entities. The HLR is a database used to store information regarding the users of the wireless network. When a wireless telephone is purchased, the phone must be activated before it can be used on the network. The purpose of the activation is to program its serial number into the HLR database.

Whenever the wireless telephone is activated (powered on), the serial number is transmitted to the closest cell site. This information is used to identify the location of a wireless station, so that incoming calls can be routed to the appropriate cell site for reception by the destination wireless phone. The wireless phone continues to transmit its identification at regular intervals, so the network always knows the whereabouts of all active wireless phones. This is in addition to the SAT, used by the MSC to measure the signal strength of wireless telephones as they move from one cell area to another.

The VLR is a database used to store information regarding wireless telephones being used in the coverage area that are not normally registered to operate in this home area.

With the advent of IS-41, roaming can now be seamless. No advance provisions are necessary; the user just carries the wireless telephone from one area to the next. This technology is widely deployed in many wireless networks today. IS-41 provides the signaling protocols necessary for wireless providers to share database information.

The Operation and Maintenance Center is used to access the *Equipment Identification Register* (EIR) and the *Authentication Center* (AC). The EIR stores the identification serial number of all wireless telephones activated within the coverage area. The AC stores a security key embedded into all wireless phones. This code is transmitted along with the serial number when the phone is activated and prevents unauthorized phones from being used on the network.

So far, we have discussed only the wireless network, which must connect to the PSTN at some point. The signaling information used to request service and connect calls within the public network is sent through the SS7 network. The MSC connects to the SS7 network via an STP and the ISUP, TCAP, and MTP protocols. The SS7 network is instrumental in connecting all the wireless providers together and allowing their various databases to be shared with one another.

The IS-41 Network. In the United States, moving from one calling area to another no longer requires prior negotiations with the service providers. When the calling area belongs to another service provider, access to subscriber records is achieved using IS-41, and subscribers are able to use their wireless phone as if they were in their own home network. To support seamless roaming, wireless providers have negotiated interconnect agreements, providing access to each other's databases.

To provide for seamless roaming between calling areas, the *Electronic Industries Association / Telecommunication Industry Association* (EIA/TIA) developed the IS-41 protocol (endorsed by the *Cellular Telecommunication Industry Association* [CTIA]). IS-41 is really an application entity, which relies on the TCAP and the SCCP protocols to travel through the network. The standard is divided into a series of recommendations presented by the *TIA 45 subcommittee* (TR45). The same subcommittee is responsible for PCS standards.

The principal differences between IS-41 and GSM lie in the protocols used to communicate between the various entities and the frequencies of the telephone units themselves. The network topology is virtually the same (see Figure 1.7).

IS-41 aligns with the *American National Standards Institute* (ANSI) version of SS7, using the TCAP protocol from the SS7 protocol stack to communicate with databases

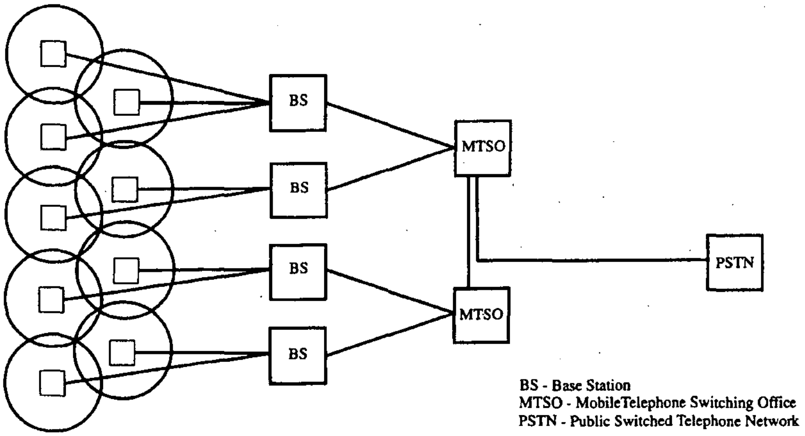

BS
BS
BS
BS
MTSO
MTSO
PSTN
BS - Base Station
MTSO - MobileTelephone Switching Office
PSTN - Public Switched Telephone Network

Figure 1.7 A typical cellular network in the United States.

and other network entities. These are the same databases found in the GSM network, the HLR, VLR, and EIR. The ISUP and MTP are also used in the IS-41 network to connect wireless calls to the public-switched telephone network, and to connect wireless circuits from the MSC to the base stations.

The IS-41 network uses an MSC, as does the GSM network, for connecting to other networks and databases. Little has changed functionally within the MSC other than the fact that the data stored in the HLR can now be shared with other MSCs across the SS7 network. This requires a transport protocol (TCAP) to move the data through the SS7 network. The HLR and VLR can be collocated with the MSC.

The biggest advantage to IS-41 is the passing of wireless telephone information needed when wireless subscribers wander from one service provider's calling area to another. Previously, users had to call ahead and arrange for a special roaming number. Callers then had to call the roaming number when the user was in a different area. With IS-41, the information stored in each service provider's network is shared by passing signaling information between wireless networks.

In essence, the wireless networks were much like ISDN was several years ago. Every service provider could provide service within its own network, but was not connected with other service providers. This created a bunch of individual "islands" of service. IS-41 is the bridge that enables service providers to share database information with other networks and eliminate the need for setting up roaming in advance. Subscribers can now move from calling area to calling area without worrying about coverage.

The PCS Network. PCS is a new type of wireless communication based on the same philosophy as wireless, but with significant differences (see Figure 1.8). The most apparent difference is in the distance between the base station and the handset.

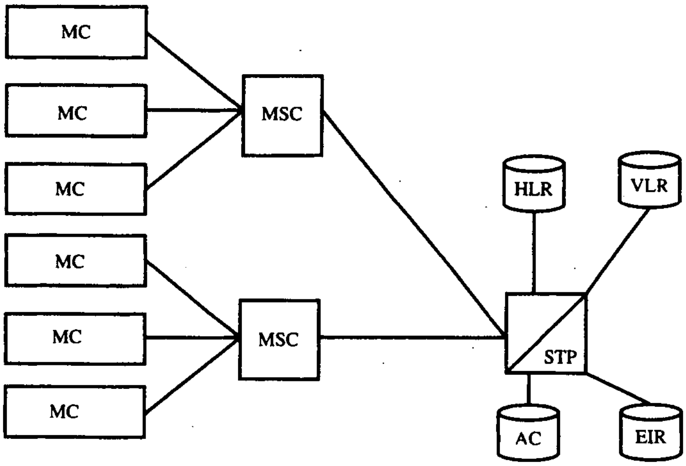


Figure 1.8 The PCS network being proposed by many cellular providers today.

PCS is really a combination of the IN and wireless networks, with a different topology. The PCS network relies heavily on the IN for the delivery of custom features on a demand basis. The PCS network offers many new features not currently available in the wireless network. These features include

- Available mode
- Screen
- Private
- Unavailable

Available mode. This enables all calls, except for a minimal number of telephone numbers, which can be blocked from reaching the PCS subscriber. Available mode relies on the delivery of the calling party number, which is checked against a database. The subscriber updates the database through the dialpad of a PCS handset.

Screen. The name of the caller appears on the display of the PCS handset, enabling PCS subscribers to screen their own calls. Unanswered calls are forwarded to voice mail or another number. The subscriber determines the forwarding destination.

Private. All calls can be forwarded, except for a limited list of numbers that are allowed to ring through. The list is maintained by the subscriber and can be changed at any time through a special set of codes, which are input by the subscriber through the dialpad of the handset.

Unavailable. Here no calls are allowed to get through to the subscriber. In the wireless network, a cell site is deployed to cover a wide area (typically four to eight miles). This distance between the subscribers' handsets (which uses radio technology) does not support the same level of quality as terrestrial lines. In PCS networks, the area of coverage is much smaller (only a radius of about a quarter of a mile). This allows calls to be of a much higher quality than in the traditional wireless network. In addition, PCS handsets use digital technology, which is quieter than analog, and supports advanced services such as SMS.

The disadvantage of this type of deployment is in the cost of the network. To cover such a small area in a metropolitan area requires many more antennas and transceivers than in a wireless network. The positioning of these antennas is also critical. Large towers, such as those currently used for wireless, are not acceptable in most neighborhoods, which is where many of these PCS antennas may have to be located.

The base station communicates to other networks (whether wireless or the PSTN) through the PCS Switching Center (PSC). The PSC connects directly to the SS7 network with a signaling data link to an STP. PCS networks rely heavily on the SS7 network for internetworking as well as for database access.

The most important function within the network is its many databases. These databases support the revenue-generating services used by the subscribers. The problem in wireless networks is that the database access takes place through the mobile switching center, and the databases are a noncompatible mixture of mainframes and minicomputers deployed in proprietary networks. In the case of PCS, the database is located in the SS7 network itself. The SS7 SCP acts as the interface point to these databases. These databases also store data for the PSTN, such as calling card information and subscriber data, and also are capable of providing information to all networks regarding PCS telephones.

This is an improvement over wireless, in which each network has its own database. In theory, the database in PCS networks can be centrally located, so all service providers can share it equally, but, in practice, each service provider has deployed its own database services and charges other service providers an access fee for sending queries to their database.

The purpose of having quick access to the telephone network databases is to allow subscribers of PCS services to define their own call-handling instructions and tailor their services to meet their immediate needs. For example, a subscriber may want to be left undisturbed during an important meeting, yet is expecting an important phone call from the president of a new client company and wants to receive that call during the meeting. The telephone number of the company president could be entered into the database with a service that would normally route all calls to voice mail, except that the call from the president would be routed to the PCS subscriber immediately.

The PCS radio spectrum is much higher than the 900 MHz wireless network. In 1993, the FCC allocated 160 MHz of bandwidth in the 1850 to 2200 MHz range. The FCC divided this range into licensed frequencies and unlicensed frequencies. The licensed frequencies have been further divided into seven separate bands.

The service areas have been drawn according to Rand McNally's boundaries for *major trading areas* (MTAs) and *basic trading areas* (BTAs). There are 51 MTAs and 492 BTAs in the United States. A debate is taking place over the service areas and is related to the fairness of trading areas versus the LATAs created by the Justice Department during the divestiture of the Bell System. Many service providers argue that it is unfair to use different boundaries for PCS than those issued for local telephone companies. Others argue that the LATAs are a better approach because they represent a more fair and equitable market.

To add to the subscribers' confusion in this market, consumers are now faced with the decision of whether to buy 49 MHz cordless telephones, 900 MHz wireless telephones, or PCS communications devices. The manufacturing community has resolved this problem by developing handsets that are dualmode, working in both analog wireless networks and digital PCS networks. Several carriers have used the tried and true GSM technology for their networks, allowing subscribers to use their phones in Europe and Latin America, which are predominantly GSM networks. Although there has not been widespread acceptance and deployment, GSM networks are growing, and coverage with GSM is now available in all the major markets.

Video and the Telephone Network

During the early to mid-1990s, there was a lot of activity between cable television companies and telephone companies. The cable companies own lots of coaxial cable that is already installed in nearly every American home. In fact, statistics show cable companies already pass 90 percent of the homes in the United States.

Along with that, coaxial cable is a healthy fiber optic backbone, capable of handling lots of high-speed data. Although the cable network may be inadequate for broadband data services, it certainly can handle the demands of most telecommuters today. In fact, with a bandwidth of at least 10 Mbps, cable telephony is the answer for many of those who must reach LANs from their homes. Cable modems are now readily available from all of the major cable companies and have proven to be more popular than ISDN or DSL.

The cable companies today have already successfully deployed voice services using their fiber optics network, supporting voice, data, and cable television to the subscriber's premises. Rather than deploy expensive switching equipment as used by the BOCs today, cable companies have been deploying cheaper, TCP/IP-based packet networks to deliver voice and data.

Another application is multimedia. Multimedia enables video to be combined with graphics and a user interface that enables the user to make selections by choosing icons on the television screen. When an icon is selected, a database is accessed to retrieve specific information. The medical profession is a big supporter of this type of service. Distance learning and teleconferencing are also likely candidates. However, the medical profession has already begun using this technology for patient diagnosis.

For example, let's say a doctor is sitting in his office in Fargo, North Dakota, diagnosing a patient with possible heart disease. For a second opinion, the doctor dials a specialist in New York City. By using a video camera connected to the telephone and a computer connected to the same interface, the doctor can video the entire office visit live to the specialist in New York City. The specialist can ask pertinent questions regarding the patient and view the patient through video as if the specialist were in the office with the patient.

The patient's medical records can be transmitted in seconds to the specialist, while the office examination is underway using the computer. The specialist can even add comments to the file and send it back to the patient's doctor. Sound futuristic? This is already in use today not only in the medical field, but in many other professions as well. In fact, Web conferencing has grown in popularity over the last year or so, allowing meeting participants in diverse geographical locations to gather together, sharing documents and presentations through a Web-based server (accessed from their desktop computers) while discussing the information over a voice conference call.

SS7 is in the background during all of this activity, quietly setting up connections and accessing databases. Information needed by the telephone companies to complete the video conference is provided by SS7 by accessing a SCP where all the information is stored and sending it to the requesting end office. SS7 will provide the connection control and billing services needed for such services.

Broadband Data Communications

Broadband (see Figure 1.9) is formally defined as any data communications with a data rate from 45 up to 600 Mbps. While SS7 does not carry the actual user data, it does provide the services necessary to connect the end-to-end telephone company facilities required for data transfer between two end points. The specific requirements of SS7 have been defined (see Chapter 9, "Overview of ISUP") and are being used in networks where switched ATM is deployed.

ATM is a transport protocol that relies on upper-layer protocols for functions above and beyond layer 3. The ATM technology was created to support broadband ISDN, but has found favor with a number of other networking protocols as well. Frame Relay, with speeds up to 155 Mbps, has been used as a solution for those requiring a PVC service. These work much like a dedicated special circuit and are not typically switched, but can be.

TCP/IP has also found new popularity as new telephone companies build entire networks based on this protocol for both voice and data. TCP/IP is not a good protocol for voice transmission today because the protocols used within TCP/IP networks do not support real-time transmission (such as video and voice) and introduce a lot of delay in the transmission. For this reason, many of the companies deploying TCP/IP for real-time applications are using ATM as the transport, which supports real-time applications.

Many service providers suggest narrowband ISDN (1.554 Mbps) as cheaper access to broadband services, while broadband ISDN (155 Mbps) is being targeted at the mega-networks that need much more bandwidth.

BISDN provides fast data transmission for services such as video dialtone. With the bandwidth of BISDN, video and audio can be transmitted simultaneously on the same

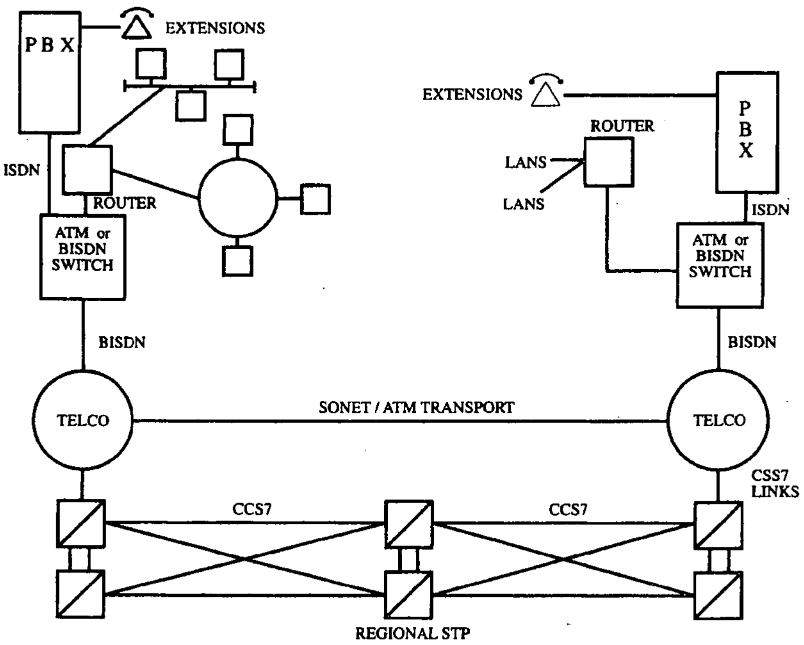


Figure 1.9 This drawing depicts a typical broadband network. The subscriber side of the network may consist of these components or others. The SS7 network is used to control the connection of circuits between exchanges (end-to-end).

facility. A point that often gets lost among all the marketing hype over ATM is the purpose of this technology. Talk of a new telephone network is nothing new. Telephone company officials have been planning for many years to upgrade their old analog networks to support the services and applications under much demand. Subscribers are no longer satisfied with slow data transmission and expensive special circuits for sending data and other information through the network.

ATM was developed to eliminate the need for time division multiplexed (TDM) circuits used in telephone networks between switching centers. The concept is to support one common network, rather than several distinct networks serving specialized applications such as television broadcast and voice transmission. Many vendors have placed ATM switches in the hands of the consumer (philosophically), thinking that this will meet the needs of every business and become a necessity in office buildings everywhere. Some have even been so bold as to claim that ATM will replace the LAN.

In reality, ATM is found at customer premises where nothing else will deliver the bandwidth required for video and data services. ATM is too expensive for the average business to deploy in place of a LAN, and certainly constitutes overkill for most daily business activities. But for the university needing the 600 Mbps of bandwidth and the hospital using the telephone network to send high-resolution medical images, ATM and broadband ISDN can be an integral part of their networks.

Narrowband ISDN is still used as the customer interface for many subscribers, whereas ATM has become the transport mechanism within the telephone network to the destination. DSL technology is now being deployed as a better (and more affordable) alternative to narrowband ISDN. One of the attractions of DSL is that existing twisted copper wire can be used. Likewise, with the popularity of the Internet, TCP/IP transmission over DSL or modem cables has become equally popular.

ATM requires a transmission facility capable of carrying the bandwidth (600 Mbps and higher). SONET is being deployed as the physical medium for ATM.

The Information Highway

No book on telecommunications networking would be complete without some discussion on the Information Highway. The Information Highway is a phrase coined from the Clinton administration describing the technologies required to provide information services to all citizens. The requirements for the Information Highway are unclear. The only criterion with which politicians seem to agree is the ability for all citizens, regardless of status or wealth, to have access to a wealth of information using a variety of media, all accessible through the PSTN.

This Information Highway in itself requires a major investment in the telephone network infrastructure. Certainly, this task could be achieved today if private networks were used and if information was somewhat centralized. This is not the philosophy of the Internet, and this has driven the transformation of the telephone network into an all-purpose information network, capable of providing access to telephones, databases, and video sources anywhere in the world.

The challenge is not how to access information from a variety of locations. This can be done today through packet switching. The challenge is how to provide this access through the PSTN to anyone anytime, and to support all forms of media, whether it be audio, video, high-resolution graphics, or data.

The telephone network was not designed to handle much more than voice and some data transmission. The telephone companies have embarked on a major upgrade of the existing infrastructure. This upgrade has been implemented in phases.

The first phase is to upgrade the carrier facilities. Telephone switching offices typically use DS1 or DS3 circuits for interoffice trunks. These facilities carry voice, data, and signaling traffic between exchanges. Video and high-speed data require more bandwidth than any of these facilities provide and, for that reason, fiber optic facilities are now replacing the previous DS1s and DS3s.

This fiber optic technology, called SONET, provides the bandwidth necessary to meet almost all applications. Yet another technology will be necessary for the switching and routing functions to carry the information from originator to destination across the backbone network. ATM has been developed for this reason.

ATM provides the mechanism for transporting information in all forms of media from one exchange to another, and eventually to the subscriber. ATM development has been slow, as telephone companies' battle with the cost of the existing infrastructure and the cost of replacing that infrastructure with new technology.

There was much talk about the Information Highway, ATM, *video on demand* (VOD), and high-speed data being available to every household by the end of the Clinton administration. But this development did not happen as fast as most would have liked.

An existing technology has suddenly gained popularity for voice transmission. VoIP has been pushed into the limelight by what began as a means for bypassing the existing telephone network (and associated long-distance costs) by making telephone calls over the public Internet. Many technical issues are associated with using the public Internet, delay being the most prominent. Nonetheless, the industry has recognized the many benefits of TCP/IP as a means of transmitting voice as well as data.

If there is any technology that will enable us to realize the Information Highway, it is TCP/IP. Already being deployed nationwide by a number of carriers, these networks will enable subscribers to send and receive packetized voice, data, audio, and video on computer terminals operating with software that supports telephone applications as well as audio and video. The following section describes packet telephony in more detail.

Convergent Networks and Packet Telephony

The Internet has spawned a completely new telephone network. VoIP has become a reality, with several companies currently providing VoIP services as an alternative to local telephone companies. Some of these companies are providing their network to other telephone companies, routing packetized voice over the public Internet. (This is not the trend for the majority by the way, as the public Internet lacks the robustness and reliability needed for telephone services.)

TCP/IP brings many benefits to telephone service providers today. For new companies deploying new networks, packet-switched technology is much more affordable (and more efficient) than the older legacy switching systems. The issue of VoIP and the incapability of IP to support real-time applications such as voice and video remain to be addressed, but there are other benefits to this approach of networking.

A surprising number of new telephone companies are popping up all over the world, building their entire voice and data networks with TCP/IP. Rather than use expensive TDM-based switches, they are deploying next-generation computer platforms designed specifically for voice and data transmission in carrier networks. This new generation of switches is fully programmable, much more scalable, and, of course, much cheaper, enabling new startups to deploy many switches in markets they would not otherwise be able to afford.

Many existing carriers are looking at TCP/IP for a different reason. They have realized that in many parts of their network, TCP/IP will save them money. Because many networks rely on databases to deliver services to their customers, and because these databases are deployed on computer platforms that support TCP/IP, carriers are finding that connecting SS7 networks to these databases via TCP/IP makes good sense.

Even STPs can be connected via TCP/IP today, reducing the cost of facilities within the SS7 network as well as increasing the bandwidth of these switches to 100 Mbps per link (rather than 56/64 kbps supported by DS0A links).

Using TCP/IP as a transport for SS7 provides huge benefits to carriers. SS7 is nothing more than packet data, and TCP/IP is a packet network protocol. Sending SS7 traffic over TCP/IP can be much cheaper because the facilities are less expensive than conventional channelized facilities. Another advantage to TCP/IP is the fact that every network supports it for data transmission (and Internet access). TCP/IP is everywhere, so finding a network to connect to is much easier than finding an existing ATM network.

Some issues still exist, however. The IP protocol does not support many of the protocol functions we will discuss later in this book. Specifically, network management (a layer 3 protocol function) is lost when using IP. The industry is actively working to solve this problem through a variety of solutions. Changes are being made to the IP protocol to support this type of function, and vendors are developing overlay protocols that emulate the functions of SS7 layer 3 in an IP network (such as Tekelec's *Transport Adaptation Layer Interface* [TALI]). The *Internet Engineering Task Force* (IETF) is defining a new set of TCP/IP-based protocols specifically for use in transporting SS7 (level 4 protocols) over TCP/IP networks.

Carriers are already deploying TCP/IP into their networks today as a transport for SS7 as well as data. By moving data traffic away from the existing interoffice facilities onto TCP/IP networks, carriers save money they would normally have to spend to expand their switching equipment in order to support the growing number of callers making long-duration data calls.

A number of companies are working on standards to support this new networking trend. This has impacted the telephone industry more than any other technology and warrants discussion in this book. The impact on SS7 will be significant, but by no means will packet telephony make SS7 obsolete (although it will generate a new generation of SS7 protocols and procedures). If anything, packet telephony will enable telephone carriers to use the full potential of SS7.

New Architecture

There are numerous proposals for the new converged networks. The data world tends to follow the architecture of data networks, whereas the telephony world leans towards architectures more similar to the existing circuit-switched networks. There are pros and cons for both. To understand the basic concepts of next-generation networks and the factors that will cause this architecture to evolve, one must first understand why the industry is moving in this direction.

Traditional circuit-switching equipment used in today's legacy networks provides three core functions: transmission, call control, and switching. The call control is the most expensive component, requiring powerful processors duplicated throughout the switch.

If this function could be removed from the switch and placed in a more central location, the function of transmission becomes more manageable, scaleable, and less expensive. In the next-generation network, the gateway supports transmission and switching functions, while the *media gateway controller* (MGC) provides call logic for the media gateways. One media gateway controller can then interface to many media gateways.

There are many other benefits to this type of design, all related to centralizing your intelligence source. When one thinks of all the support systems needed to maintain the network and provide back-office functions such as billing and service creation, centralizing call control makes a lot of sense, rather than distributing it all over the network.

Routing SS7 through TCP/IP networks presents an entirely different set of problems. As we examine the SS7 MTP protocol, you will find that the SS7 network sends continuous messages to every node in the network. The purpose of these messages is to determine whether the node is still capable of handling traffic. If a node fails, the network knows about the failure immediately and notification of adjacent nodes is already taking place. The SS7 MTP protocol is much more proactive than TCP/IP routing protocols, which is why there is still much work to be done to support these functions in TCP/IP networks.

The best approach for converging networks is to centralize the intelligence of the network, as shown in Figure 1.10. By centralizing the SS7 to IP conversion, you minimize the administration within the network. You also provide a central point of access for applications such as network monitoring, billing, and data mining.

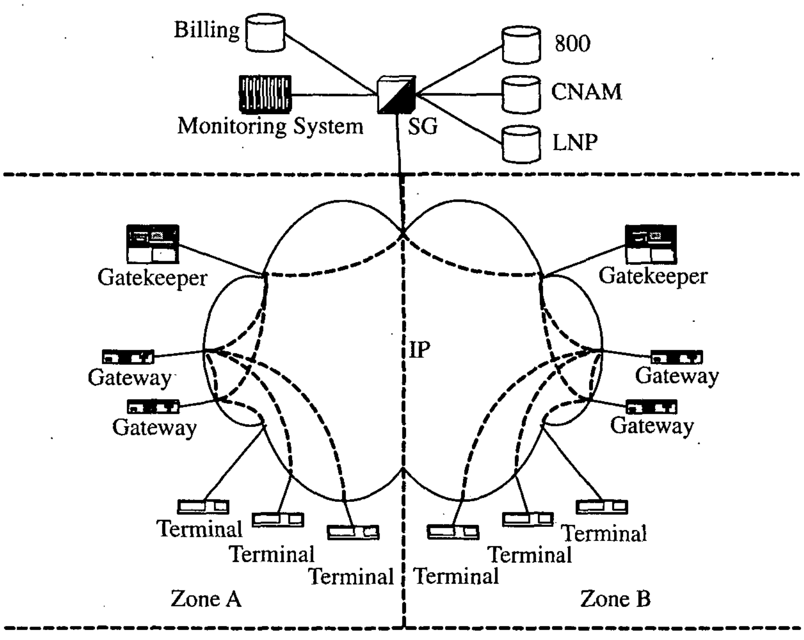


Figure 1.10 A centralized approach for network intelligence, supporting cheaper, nonintelligent devices at the network edge.

This architecture also lowers the cost of network equipment. By moving the intelligence up in the network, you maintain simplicity in the edge equipment. Because there are more of these devices than databases and STPs, this is where the cost should be minimized.

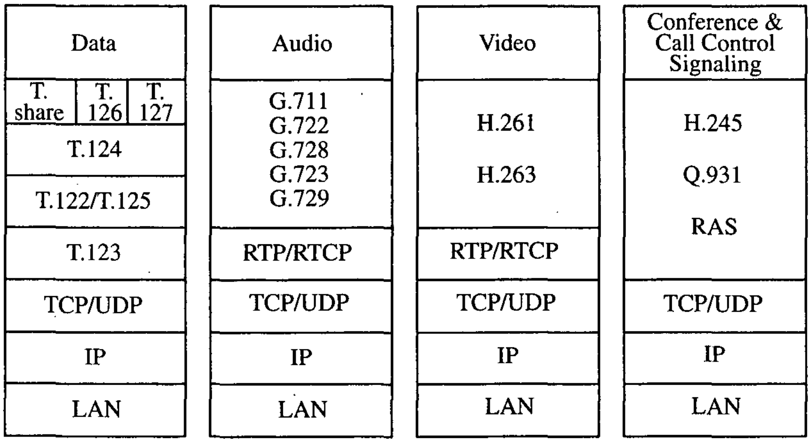
If legacy equipment must be interfaced, some vendors offer front ends. These devices are more like scaled-down signaling gateways, providing a conversion between SS7 and TCP/IP. This is much more economical than developing modules to incorporate into vendors' products and provides a faster path to market than if development was required.

There are really two different efforts underway to define the various standards to be used for convergent networks. The ITU is defining the standards for protocols to be used in convergent networks on an international basis. The IETF is addressing standards for use in North America. Although there are many similarities to both sets of standards, they are very different from one another.

ITU-TS H.323. H.323 is really a suite of standards defining a variety of media types through packet networks. These standards cover everything from voice to data, facsimile, and even video. They define how the signals are to be converted from analog to digital and what signaling is to be used. It is this set of standards that is being adopted for VoIP networks throughout the world, with the exception of North America, which follows the ANSI standards.

In Figure 1.11, we see the various recommendations from the ITU for the applications to be supported. For the most part, these recommendations come from the H.323 suite of recommendations.

The ITU has defined several entities for the converged packet telephony network. The four principal components are the terminal, gateway, gatekeeper, and signaling gateway. Each of these components has distinct functions, and various protocols are be-



Data			Audio	Video	Conference & Call Control Signaling
T. share	T. 126	T. 127	G.711 G.722 G.728 G.723 G.729	H.261 H.263	H.245 Q.931 RAS
T.124					
T.122/T.125					
T.123			RTP/RTCP	RTP/RTCP	
TCP/UDP			TCP/UDP	TCP/UDP	TCP/UDP
IP			IP	IP	IP
LAN			LAN	LAN	LAN

Figure 1.11 The protocol stack for H.323 applications.

ing defined to operate between the various components. Although similar functions are defined in the IETF standards, they are not identical to those defined by H.323.

Terminals. These are the actual telephone devices themselves. They can be part of a computer terminal or some other form of device with the responsibility of supporting voice transmission over the packet-switched network. The voice is digitized and transmitted over the network using the *Real-Time Protocol* (RTP), which is defined in TCP/IP. Think of terminals as computers on a LAN, even though in actuality the standards are referring to a logical function rather than a physical entity.

Gateways. The gateway is used to connect to other networks. It provides protocol conversion (say, from an ANSI protocol to an ITU version of the same protocol) and uses Q.931 and Q.2931 (both defined in the ISDN recommendations) or a version of the *Media Gateway Controller Protocol* (MGCP) for sending call control and signaling between terminals and gateways.

Gatekeepers. Gatekeepers are deployed and assigned to zones, much like a tandem telephone switch. The gatekeeper is the central point for all calls within its own zone and interconnects with gatekeepers in other zones using Q.931/Q.2931 or some other similar protocols such as MGCP, Megaco, or the *Session Initiation Protocol* (SIP). Gatekeepers communicate to the PSTN network through a signaling gateway (see Figure 1.12).

Signaling Gateway. The signaling gateway acts as a bridge between the PSTN and the packet telephony network. Standard SS7 messages are received from the PSTN and are converted to TCP/IP-based messages. The upper levels of SS7 can remain the same as they are in the PSTN (ISUP, SCCP/TCAP), but the MTP is not usable in packet telephony and must be eliminated. Unfortunately, this means the many functions of MTP must be emulated (or replaced by something else) in the packet network. A great deal of work is being done today within the standards groups to define the methods to be used for SS7-type signaling within packet networks.

Internet Engineering Task Force (IETF) The IETF has defined components similar to the ITU, but they use entirely different terminology.

Media Gateway (MG). The MG is synonymous with the terminal. It supports the digitization and transmission of voice over the packet network. Many vendors are using Q.2931, MGCP, Megaco, or some other derivative for call control between MGs and MGCs. RTP is being used for the voice transmission over IP.

Media Gateway Controller (MGC). The MGC is much like the gateways and gatekeepers defined by the ITU. They are used to connect to other networks and provide interworking between dissimilar networks. The MGC interfaces with the signaling network, receiving information about the call from the media gateway (using Q.2931 or SIP).

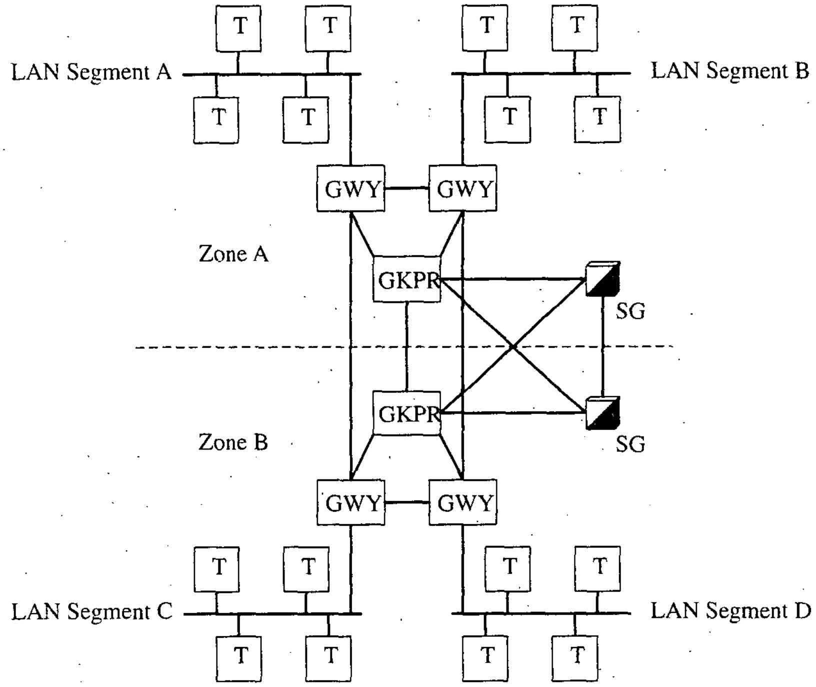


Figure 1.12 The role of the gatekeeper and the gateway, and how they tie networks together.

Several competing protocols are being introduced as possible standards for these components to use for call control between one another, but none have been actually adopted as yet. It is important to understand that these protocols really do not replace SS7 because they will not work in the PSTN world. Rather they interface to SS7, through the *Signaling Gateway* (SG).

Signaling Gateway (SG). This component is the same as in the ITU world and is used to bridge the PSTN networks with the packet networks. The signaling gateway uses mapping tables to map SS7 point codes to the appropriate IP address. The gateway must be able to maintain current network status information as part of these routing tables to ensure the proper routing of messages during failure modes. In the future, as new protocols such as MGCP and SIP are ratified by the IETF as standards to be used for signaling within the network, the SG may also be responsible for converting these protocols to SS7, and vice versa.

Some vendors have built SS7 interfaces into their MGCs. However, they may still rely on TCP/IP as the transport for upper-level SS7 protocols. This means something must replace the lower levels of SS7 used as a transport (the MTP).

Tekelec has submitted TALI for this purpose and has gained the support of numerous other vendors who have incorporated this interface into their own products. TALI uses TCP and adds an overlay layer to emulate SS7 MTP functions. TALI also provides additional features not yet supported in M2UA and M3UA, such as automatic *circuit identification code* (CIC) registration and MGC registration. These two features enable MGCs to be added to the network without operators having to perform translations in the network switches.

TALI will not be replaced by the protocols being developed by the IETF as a replacement to MTP. These protocols are M2UA or M3UA, SUA, and SCTP. TALI will continue to be supported as an augment to these protocols for those vendors wishing to continue supporting the interface on their platforms.

M2UA, M3UA, SUA, and SCTP are presently being defined as replacements for MTP by the IETF, providing the same services as MTP but in a packet network. SCTP is a peer protocol to the *User Datagram Protocol* (UDP) and TCP, providing connection-oriented services to guarantee the delivery of messages. The M2UA and M3UA protocols provide replacements to SS7 MTP, providing the same services as MTP in packet networks, including message delivery, network management, routing management, and routing. SUA provides delivery services for the TCAP, like SCCP does today.

Standards Organizations

The ITU commissioned the then *International Telegraph and Telephone Consultative Committee* (CCITT) to study the possibility of an all-digital IN. The result was a series of standards known now in the United States as SS7. These standards have paved the way for the IN and, with it, a variety of services, many yet to be unveiled.

The ITU-TS (once known as the CCITT) developed a digital signaling standard in the mid-1960s called *Common Channel Interoffice Signaling System 6* (CCIS6) that would revolutionize the telephone industry. Based upon a proprietary, high-speed data communications network, CCIS6 later evolved into C7 (known as SS7 in the United States), which has now become the signaling standard for the entire world.

The secret to its success lies in the message structure of the protocol and the network topology. The protocol uses messages, much like X.25 and other message-based protocols, to request services from other entities. These messages travel from one network entity to another, independent of the actual voice and data they pertain to, in an envelope called a packet, using a different facility than the voice transmission itself.

CCIS6 was implemented in the United States in the 1960s using 2.4 kbps data links, and later changing to 4.8 kbps data links. The first implementation of CCIS6 was for establishing trunk connections for voice calls. The CCIS6 protocol used packets (or signal units) arranged in groups of 12 and consisting of 28 bits each. The group of 12 signal units was arranged as one data block.

SS7 was derived from the earlier CCIS6, which explains the similarities. SS7 provides much more capability than CCIS6. Where CCIS6 used fixed-length signal units, SS7 uses variable-length signal units (with a maximum-sized length), providing more versatility and flexibility. SS7 also uses higher-speed data links (56 kbps). This makes the signaling network much faster than CCIS6. In international networks, the data links operate at 64 kbps. Recently, *high-speed links* (HSL) operating at 1.544 Mbps have been deployed in the United States, conforming to a Telcordia (formerly Bellcore) standard. TCP/IP has also been introduced as a transport for SS7 over 100 Mbps facilities.

As of 1983, CCIS6 was still being deployed throughout the U.S. telephone network, even though SS7 was being introduced. As SS7 began deployment in the mid-1980s, CCIS6 was phased out of the network. SS7 was used in the interoffice network and was not immediately deployed in the local offices until many years later.

Standards such as SS7 and TCP/IP do not happen quickly. They are the result of years of research and development conducted by standards committees. These organizations are usually composed of government agencies or industry representatives from manufacturers and service providers.

To understand the various standards available today, one must first understand the purpose of new standards and how they are developed. There are two different types of standards: de jure and de facto.

The de jure standard is formed by committee. These standards take many years to develop because the processes used in committees are long and bureaucratic. Nonetheless, many of the standards used today are the result of standards committees.

A de facto standard is the result of a manufacturer or service provider monopolizing a market. A good example of a de facto standard is the personal computer. Microsoft was instrumental in saturating the market with their operating system and applications, but, more importantly, with encouraging third-party vendors to use their operating system in their PC platforms. The result of this marketing strategy is still felt by its competitors today. There are so many Windows-based computers in the market that introducing a new platform requiring a different operating system is a high risk. Yet there are no standards in existence that define the use of Windows in all personal computers.

De jure standards and de facto standards can be voluntary or regulatory standards. Voluntary standards are adopted by companies on a voluntary basis. There are no rules that say all manufacturers and service providers must comply with a voluntary standard. However, the advantages are many. Voluntary standards help ensure that everyone developing networking products builds their products for interconnectivity. Without this interconnectivity and interoperability, only a few equipment manufacturers would win the market—those with the largest install base.

Interoperability is another issue in data communications. Interoperability is the capability for equipment to communicate with equipment from different vendors in a network environment. Often, vendors will implement varying protocol versions in their equipment that are noncompliant with the standards. When this occurs, other equipment cannot communicate with the noncompliant system because it uses a proprietary interface, forcing carriers to purchase all their equipment from the same manufacturer.

Voluntary standards help ensure the interoperability of all networking equipment. With voluntary standards, all those participating in the technology can have a voice in the final "product." The organizations responsible for creating these voluntary standards are usually made up of industry representatives. These representatives work for the same companies who build the equipment. As the technologies evolve, companies participating in the development of the standards can also get a sneak preview of what the final standards will consist of and can be the first-to-market with a product that is compliant with the standards.

Regulatory standards are created by government agencies and must be conformed to by the industry. These standards do not hold any major advantage to the service provider or the manufacturer, but are in place in most cases to protect the consumer.

Regulatory standards are monitored by government agencies such as the FCC. These agencies ensure the protection of the public and other network users by enforcing standards covering safety, interconnectivity, and, in some cases, health (that is, radiation emission from computer terminals and wireless phones).

SS7 networks use standards from a variety of organizations and standards committees. Some of the standards used in SS7 networks were developed for other applications as well, not specifically for SS7. The following organizations have written standards directly related to SS7:

- ITU
- ITU-TS
- ANSI
- Telcordia (formerly Bellcore)
- IETF

In addition to these organizations, many other standards have been written that affect SS7. The following standards organizations have contributed to the SS7 network with standards not written specifically for SS7 but used by equipment in the network. These organizations are responsible for the standards that govern the quality of cables, quality standards for manufacturing practices, electrical specifications, and interfaces used to interconnect telecommunications equipment:

- Electronic Industries Association (EIA)
- ATM Forum
- FCC
- Underwriters Laboratories (UL)
- Canadian Standards Association (CSA)
- International Organization for Standardization (ISO)

In addition to these standards organizations, many new forums have evolved. As the industry begins to understand the importance of standards bodies and compliance to these standards, new industry forums evolve composed of vendors and service providers in the industry with a vested interest in the technology.

These forums are often commissioned by the ITU-TS and ANSI to develop new standards on their behalf (such as ATM) or work on issues with existing standards (such as

ISDN). In many cases, these forums can develop standards much faster than the standards committees themselves, saving the committees years in development time.

The rest of this section will look at these organizations in greater detail. The purpose of describing these organizations is to provide a better understanding of who the players are and what significance they carry. The major organizations are described in greater detail than some of the less significant ones.

International Telecommunications Union—Telecommunications Standardization Sector (ITU-TS)

Formerly known as the CCITT, this organization is a part of the ITU, which is a *United Nations* (UN) Treaty organization. The purpose of the ITU-TS is to provide standards that will allow end-to-end compatibility between international networks, regardless of the countries of origin. The standards are voluntary standards, but many countries require full compliance to connect to their networks.

The members of the ITU-TS are government representatives from the various nations. The representative for the United States is the Department of State. In addition to government agencies, manufacturers and service providers carry some influence as well. Membership is limited to four categories:

- Administrations of a country's public telephone and telegraph companies
- Recognized private operating agencies
- Scientific and industrial organizations
- Standards organizations

The ITU was reorganized into three sectors: the *Radiocommunication Sector* (ITU-RS), the *Telecommunication Development Sector* (ITU-D), and the *Telecommunication Standardization Sector* (ITU-TS). The ITU-TS is the sector responsible for defining SS7 (or C7, as it is known internationally) standards and other related standards.

The ITU-TS standards for SS7 have been embraced by every country that is deploying SS7, yet not every country's network is the same. One would think that, with international standards in place, interconnectivity would not be an issue. Yet every country creates its own standards to meet the requirements within its own networks.

Because of this independence within individual countries, the SS7 network hierarchy consists of an international network and many national networks. The national networks are based on the ITU-TS standards, but are modified for usage within individual countries. The United States uses the ANSI standards as its national standards. The ANSI standards are based on the ITU standards, but with several differences. The differences between the ANSI standards and the ITU-TS standards are mainly in the addressing (point codes) and in network management procedures. Telcordia has also published a set of standards, based on and endorsed by ANSI. The ANSI standards and the Telcordia standards are virtually the same except for the additions and modifications added by Telcordia to ensure network reliability and diversity.

The ITU-TS C7 standards were first defined in 1980. These are referred to as the Yellow Book. ITU-TS standards were once published every four years in a set of documents, which are color-coded to indicate the year in which they were published. The color code is as follows:

- 1980 Yellow Book
- 1984 Red Book
- 1988 Blue Book
- 1992 White Book

The ITU has changed the way it releases updated standards today. Rather than wait four years, they have adopted the model used by ANSI and release new updates and standards as soon as contributions have been adopted and agreed upon by the members.

The C7 standards can be found in the ITU-TS documents numbered Q.701 through Q.741. The following list identifies all those documents that are SS7 standards or are related to SS7:

- Q.700–Q.709 Message Transfer Part (MTP)
- Q.710 PBX Application
- Q.711–Q.716 Signaling Connection Control Part (SCCP)
- Q.721–Q.725 Telephone User Part (TUP)
- Q.730 ISDN Supplementary Services
- Q.741 Data User Part (DUP)
- Q.761–Q.766 ISDN User Part (ISUP)
- Q.771–Q.775 Transaction Capabilities Application Part (TCAP)
- Q.791–Q.795 Monitoring, Operations, and Maintenance
- Q.780–Q.783 Test Specifications

The ITU-TS has changed the documentation structure from previous years. There should be no effect, however, on existing older SS7 standard publications.

It should also be noted here that the ITU sponsors the world's largest telecommunications trade show every four years in Geneva. This trade show (as well as the associated conferences) have become the highlight of the industry, with companies from all over the world assembling to discuss the latest technologies and products available.

American National Standards Institute (ANSI)

The ANSI (see Figure 1.13) is responsible for approving standards from other standards organizations for use in the United States. Many organizations are considered accredited standards bodies by ANSI, including Telcordia and the EIA. ANSI is divided into

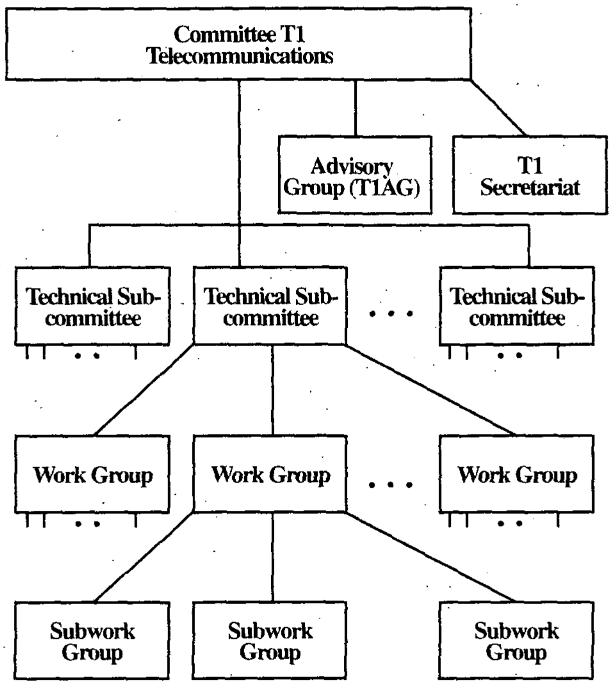


Figure 1.13 The ANSI organization chart.

committees. The ANSI Accredited Standards Committee T1 is responsible for standards associated within the telecommunications industry.

The T1 committee's responsibilities include developing standards for the interconnection and interoperability of telecommunications networks. The T1 committee is divided into seven technical subcommittees that receive their direction from the T1 Advisory Group (T1AG).

The T1AG meets on a bimonthly basis and is responsible for the establishment and administration of procedures for the committee's activities. As illustrated in Figure 1.13, the Advisory Group reports to the T1 committee.

The secretariat provides support functions to the subcommittees and interacts with ANSI to get approval and publication of the standards created by the subcommittees. Duties of the secretariat include scheduling committee meetings and approving memberships.

The following is a description of each of the seven subcommittees along with their missions, according to the T1 Committee Procedures Manual.

Technical Subcommittee T1E1. This subcommittee defines standards for network interfaces, concentrating on physical layer interfaces. This includes the electrical, optical, and magnetic specifications of data communications interfaces as well as telephony interfaces.

It consists of four working groups, all which work closely with each other and with other groups within the T1 Committee:

- T1E1.1 Analog Access
- T1E1.2 Wideband Access
- T1E1.3 Connectors and Wiring Arrangements
- T1E1.4 DSL Access

Technical Subcommittee T1M1. This subcommittee's primary focus is on the processes and procedures used in the operations, maintenance, and administration of telecommunications networks. This includes the testing of facilities, measurements, routine maintenance, and traffic routing plans. Their standards focus on the engineering and planning functions, network resources, and support systems. There are four working groups in this subcommittee:

- T1M1.1 Internetwork Planning and Engineering
- T1M1.2 Internetwork Operations
- T1M1.3 Testing and Operations Support Systems and Equipment
- T1M1.5 OAM&P Architecture, Interfaces, and Protocols

Technical Subcommittee T1P1. This subcommittee provides support services and program management for the rest of the T1 subcommittees. It provides high-level descriptions, high-level overviews and architectures, scheduling for interactive sessions between subcommittees, publishing support of standards, and reference models. It also determines T1 endorsement of programs.

Technical Subcommittee T1Q1. This subcommittee focuses on performance issues of network traffic, switching, transmission, maintenance, availability, reliability, and restoration. Its performance specifications are standards used from carrier to carrier and from carrier to customer interfaces. There are four working groups:

- T1Q1.1 4 kHz Voice and Voiceband Data
- T1Q1.2 Survivability
- T1Q1.3 Digital Packet and ISDN
- T1Q1.5 Wideband Program

Technical Subcommittee T1S1. The T1S1 subcommittee is directly involved with SS7 signaling standards as well as ISDN and other related services, architectures, and signaling. It reviews international standards and makes decisions on how those standards will be implemented in the United States. In addition, it works closely with the ITU-TS in developing standards for the international community. There are four working groups in this subcommittee:

- T1S1.1 Architecture and Services
- T1S1.2 Switching and Signaling Protocols
- T1S1.3 Common Channel Signaling
- T1S1.5 Broadband ISDN

Technical Subcommittee T1X1. The members of this subcommittee define the standards used to define the hierarchy of digital networks and synchronization networks. This subcommittee defines the standards used for internetworking, focusing on the functions necessary to interconnect at the network transport level. There are four working groups:

- T1X1.1 Synchronization Interfaces
- T1X1.4 Metallic Hierarchical Interfaces
- T1X1.5 Optical Hierarchical Interfaces
- T1X1.6 Tributary Analysis

Technical Subcommittee T1Y1. This subcommittee defines standards not covered by any of the other subcommittees and serves as a sort of miscellaneous standards committee. Specialized video and audio services, including broadcast services, teleconferencing, and graphics, as well as specialized voice and data processing, are within this group's jurisdiction. There are three working groups:

- T1Y1.1 Specialized Video and Audio Services
- T1Y1.2 Specialized Voice and Data Processing
- T1Y1.4 Environmental Standards for Exchange and Interexchange Carrier Networks

The ANSI publications regarding SS7 define the function of the protocols. Telcordia has published numerous other documents detailing the specific requirements of all SS7 entities and management procedures. Telcordia protocol chapters and ANSI publications for SS7 are numbered as follows:

- T1.110 Signaling System 7 (SS7), General
- T1.111 Message Transfer Part (MTP)
- T1.112 Signaling Connection Control Part (SCCP)
- T1.113 ISDN User Part (ISUP)

- T1.114 Transaction Capabilities Application Part (TCAP)
- T1.115 Monitoring and Measurements
- T1.116 Operations, Maintenance, and Administration Part (OMAP)

An ANSI catalog is available for all ANSI publications by contacting the ANSI organization in New York City.

Telcordia (Formerly Bellcore)

Telcordia began as the research and development arm for the seven BOCs (known then as Bell Laboratories). These seven companies were divided from the Bell System in 1984 as part of the divestiture. AT&T was separated from the local exchanges, taking with it Bell Laboratories. The new BOCs then founded what was then named Bellcore to replace the services once provided them by Bell Laboratories. Bellcore was later sold and separated from the BOCs and named Telcordia. The seven operating companies were

- Ameritech
- Bell Atlantic
- BellSouth Telecommunications
- NYNEX
- Pacific Telesis
- Southwestern Bell
- US West

Through the many mergers and acquisitions over the last few years, this list has dwindled down to just a few companies. Today the RBOCs are known as

- Verizon
- BellSouth
- Southwestern Bell (SBC)

In addition to these regional operating companies, Telcordia also provided services to the following "nonowners:"

- Cincinnati Bell, Inc.
- Southern New England Telephone Co.
- Centel Corporation
- General Telephone Company and its local telephone companies

- Sprint and its local telephone companies
- Canadian local telephone companies

Telcordia publishes standards and recommendations for all kinds of telecommunications services, maintenance and operations procedures, and network architecture. These documents are available for a fee to any individual through the Telcordia organization.

Telcordia changed their documentation numbering system a few years ago, so you will find some older documents may not match their newer counterparts. What is described next is the former identity used to communicate document phases. The three levels described have since been collapsed into one identity, known as the *Generic Requirement* (GR). All new Telcordia documents will use this new convention. The former identity is defined because there are still many documents in circulation using the older identification.

Bellcore documents were developed and published in three phases. The first phase of publication was the *Framework Advisory* (FA). Any document beginning with FA is a Framework Advisory and is still in draft format. These documents were published and submitted to the industry for comments and suggestions.

The second phase was known as the *Technical Advisory* (TA). These documents were preliminary publications and were still subject to minor changes. They were submitted to the industry for final comment and suggestions before reaching final publication.

The third phase of publication was the *Technical Reference* (TR), which was the final phase of the document. TRs represent final released versions of a document. All of these publications were numbered with the following convention: TR-NWT-000082

The preceding document number is just one example of the document numbering of Telcordia publications. The TR indicates that this is a technical reference, the NWT indicates this is a network-related publication, and the six-digit number identifies the unique publication. As previously mentioned, Telcordia has recently implemented a new document numbering scheme, which alleviates the three levels of numbers and identifies all new documents as GRs.

In addition to these types, Telcordia also publishes documents on technology findings and services offered through their regional companies. These are referred to as *Science and Technology* (ST) publications and *Special Reports* (SRs).

SRs are issued for a variety of reasons today. Vendors who deal with the RBOCs are sometimes required to have their systems tested by a third party (such as Telcordia) and provide the results of the testing through a publication. Telcordia publishes their test results through serialized SRs, which are tightly controlled documents (in this case).

They also issue SRs to the general industry when there is an event that the industry needs to know about. For example, a major network outage is usually investigated, with the results published in an SR for the industry. This is one means of educating the entire industry of potential network problems so the same event will not repeat itself.

A publication regarding Telcordia requirements for interconnectivity within networks is called an FR and specifies design requirements as well as the functionality of specific network elements. These are classified as

- LATA Switching Systems Generic Requirements (LSSGR) (FR-NWT-000064)
- Operator Services Systems Generic Requirements (OSSGR) (FR-NWT-000271)
- Operations Technology Generic Requirements (OTGR) (FR-NWT-000439)
- Reliability and Quality Generic Requirements (RQGR) (FR-NWT-000796)
- Transport Systems Generic Requirements (TSGR) (FR-NWT-000440)

The Telcordia publication GR-246-CORE is a multivolume series that closely matches the ANSI publications and defines the SS7 protocol. Telcordia has added many modifications and procedures enhancing the security and reliability of SS7 networks. Originally, the Telcordia version identified the Bell System implementation of SS7 and identified specific procedures and functions required in these networks.

The chapter numbering is the same as in ANSI publications, allowing cross-referencing between publications. The Telcordia version identifies the suggested 1requirements for implementation of SS7 in the United States as adopted by many carriers (including the BOCs) and identifies specific procedures and functions required in these networks. The protocol descriptions are identical to the ANSI publications, with the addition of Telcordia implementations. A catalog of Telcordia documents is available through Telcordia at www.telcordia.com.

Electronic Industries Association (EIA)

The EIA develops standards focused on the physical interfaces used in the telecommunications industry. Probably the best example of an EIA standard is the RS-232C. This standard was originally created for interfacing modems to computer systems, but it proved so simple and versatile that it quickly became the de facto standard for any application requiring a serial interface.

The EIA has developed many other standards besides the RS-232C, including some faster interfaces designed to replace the RS-232C. The EIA is also responsible for many wireless standards, including the wireless standard called IS-41. This standard defines the interfaces between wireless network entities as well as the communications protocols used at these interfaces. The EIA/TIA Subcommittee TR-45.2, Wireless System Operation, developed the IS-41 standard.

ATM Forum

The ATM Forum is a voluntary organization consisting of industry and public sector representatives. Their mission is to assist the ITU-TS in developing a standard for international use in ATM networks. The ITU-TS has been actively working on this standard, but was not expected to complete the standards until the year 2000. The ATM Forum was formed to help expedite the process of development, in hopes of finishing this standard much sooner.

Much of the work accomplished by the ATM Forum has been submitted and accepted (with modification) by the ITU for inclusion in the final ATM standards. It is important to understand, however, that the ATM Forum does not write standards. They write implementation agreements, which allow vendors to agree on certain aspects of the technology and begin development on products prior to the final standards being completed.

Formed in November 1991, the ATM Forum consists of representatives from a wide spectrum of interests. Data communications and telephone companies, service providers, and private networking types have joined together in this consortium to ensure a standard that meets the needs of all interested industries.

Federal Communications Commission (FCC)

The FCC was created as part of the Communications Act of 1934 and is responsible for the regulation of the airwaves as well as for the regulation of the telecommunications industry. There are five commissioners in the FCC, all appointed by the President of the United States. Each commissioner serves a term of five years. One commissioner is appointed by the President to serve as chairperson of the FCC.

Four operating bureaus exist within the FCC organization, with a total staff of 1,700 personnel. Each operating bureau serves a specific function: mass media, common carrier, field operations, and private radio. The common carrier bureau regulates all aspects of the telecommunications industry, including paging, electronic message services, point-to-point microwave, wireless radio, and satellite communications.

The FCC regulates the access of interconnect companies and determines the type of interfaces to be used. They do not necessarily create standards, but they enforce regulations regarding the interconnection of networks and network devices. They also issue licenses for radiotelephone circuits and assign frequencies for their operation. The FCC has allocated new frequencies for the PCSs being deployed in the United States by many major wireless service providers. Rather than create new frequencies, the FCC has reallocated frequencies previously used by microwave.

The FCC also supervises charges and practices of the common carriers, approves applications for mergers, and determines how the carriers will maintain accounting for their operations. The FCC also governs the type of services that a service provider can provide.

A good example of how the FCC regulates the telecommunications industry is the registered jack program. All manufacturers of telephone equipment, or any equipment that connects to the telephone network, must have their equipment registered with the FCC. The FCC then determines which type of interface (*registered jack* [RJ]) the equipment must use to connect to the network.

The former Bell System sometimes gets credit for these interfaces, since they are usually the ones who install and maintain them. But the FCC determines how these interfaces will be used and who must use them. The RJ-11, used to connect single-line telephones to the central office line, is found in every home across the United States today. Most of us refer to them as modular jacks.

The *Network Reliability Council* (NRC) was formed to monitor network outages and seek resolutions through the industry vendors and service providers. This council works closely with the key corporations in resolving key issues that can be related to service outages in the nation's SS7 networks. This information is published and shared with all service providers to prevent future outages from occurring due to common failures. The FCC formed the NRC after several network outages in the SS7 network shut down telephone service in several major cities in 1991.

Today all outages must be reported by the service providers to the NRC. This information is then disseminated among service providers as an information-sharing mechanism. The FCC hopes that sharing information regarding software deficiencies from the various vendors can help deter many network outages.

Underwriters Laboratories (UL)

The UL is a not-for-profit organization that began in 1984. The UL uses a suite of tests and gives approval to any equipment that passes the minimal requirements set by the test suite.

UL approval is not required here in the United States, but is sought after by most manufacturers of electrical or electronic equipment. The UL label certainly influences buying decisions, and in most building codes it is a requirement for electrical equipment.

Canadian Standards Association (CSA)

The CSA is Canada's equivalent of the UL. The CSA approves all electrical and electronic products for sale in Canada. Manufacturers in the United States who plan to sell their equipment in Canada usually have the equipment approved by the CSA. If the equipment is sold in the United States as well, the equipment must be UL and CSA approved.

As with the UL, CSA approval is not a requirement. However, in the telecommunications industry, most buyers will require CSA approval before purchasing equipment. This is their guarantee that the equipment they are buying went through some level of testing for electrical compliance.

International Standards Organization (ISO)

The ISO is an international standards organization responsible for many data communications standards, including the OSI model. The OSI model was developed after SS7 and was not adopted as a standard until 1984. However, layering was well understood and practiced during the early to late 1970s, which is when the work was being done on SS7 protocols. The OSI model is still used today to define the functions of the various levels within a protocol stack.

The ISO is composed of other standards bodies from various countries, mostly government agencies responsible for setting communications standards within their own governments. The U.S. representative is ANSI.

The ISO does not limit itself to just data communications standards. They have created many other types of standards as well. One of their most recent contributions to the industry is the ISO 9000 quality standards. The ISO 9000 defines processes to be used in manufacturing to ensure quality production. Again, these are not mandatory standards, but are essential for companies selling products in Europe because most European buyers now require ISO 9000 compliance.

Internet Engineering Task Force (IETF)

This organization has the responsibility of defining all standards related to the Internet. The standards are published in the form of *Recommendations for Comments* (RFCs). Anyone can submit RFCs; however, they do not become adopted standards without approval from the IETF working committees. Thousands of RFCs cover hundreds of protocols used in TCP/IP networks. The IETF also validates through testing that the proposed RFCs will actually work within the TCP/IP networks.

IETF has three different areas of concern: applications, transport, and security. Signaling is being addressed through the transport working groups. Several groups are involved in defining transport protocols: Sigtran, SIP, and Megaco.

Sigtran is defining the protocols to be used with SS7. This includes M2UA, M3UA, SCTP, and SUA. At the time of writing, Sigtran had ratified SCTP and M3UA RFCs. It should be noted that these protocols are being defined for use as a transport for SS7 protocols ISUP and TCAP, eliminating the TDM-based protocol MTP.

The SIP working group is responsible for SIP to be used between MGCs within a network. This protocol will enable call setups from gateway to gateway when MGC domains have to be traversed.

The Megaco working group is responsible for defining the protocol to be used between the MG and the MGC. This started as the MGCP, but has since been adapted and renamed as Megaco.

Other Agencies

In addition to the standards organizations, other agencies have made a significant impact on the telecommunications network. These agencies are responsible for ensuring reliability in our telecommunications network. The following are the most prominent agencies.

Network Reliability Council (NRC). First commissioned by the FCC in 1992, this council was chartered to investigate network outages and report them to all network providers as well as to vendors. They were put into existence only after numerous network outages caused telephone service to be out for extended periods of time, costing many corporations millions of dollars (including Wall Street in New York) and even necessitating the closing of an airport.

Their charter was to have expired in 1994, but they received a new charter by the FCC and are still tasked with the investigation of network outages. The commission is made up of CEOs from leading carriers and manufacturers, and provides reports regarding the reliability of the nation's network as well as explanations for outages and how they can be prevented in the future.

Network Operations Forum (NOF). The *Network Operations Forum* (NOF) was originally formed in 1984, over concerns as to who would track and clear trouble reports that crossed network boundaries. They have since expanded their operations to include definitions for interoperability testing and interworking issues. They also meet with manufacturing companies to resolve issues regarding the reliability of network equipment.

Chapter

2

The SS7 Network

The *Signaling System 7* (SS7) network is a packet-switched network (separate from the voice network) that is used solely for the purpose of connecting telephone calls. It provides two types of services: circuit-related and noncircuit-related services. Circuit-related signaling is used for the setup and teardown of voice connections in both *time-division multiplexing* (TDM) networks and packet-switched networks (*voice over IP* [VoIP]). Noncircuit-related services are all of the other services provided by the network, such as database access for translations and subscriber information, and network management.

The network is deployed as two distinct levels, or planes: the international plane, which uses the *International Telecommunications Union—Telecommunications Standardization Sector* (ITU-TS) standard of the SS7 protocol, and the national plane. The national plane uses whatever standard exists within the country in which it is deployed. For example, in the United States, the *American National Standards Institute* (ANSI) is the standard for the national plane. Telcordia standards are an extension of the ANSI protocol and ensure the reliability required to interwork with networks of the *Bell Operating Companies* (BOCs).

Other nations may have one or several different versions of national protocols for SS7. In situations where these national variants must interwork with one another, a protocol converter is sometimes used. The protocol converter maps parameters from one country variant to another. All countries are capable of communicating with one another through gateways that convert the national version of the SS7 protocol to the international version. This ensures that all nations can interwork with one another while still addressing the requirements of their own distinct networks.

All nodes in the SS7 network are called *signaling points*. A signaling point has the capability to perform message discrimination (read the address and determine if the message is for that node) and route SS7 messages to another signaling point. Every signaling point has a unique address called a *point code*. In SS7 messaging, both the origination point code and destination point code are provided. There are three different types of signaling points:

- *Service switching point* (SSP)
- *Signal transfer point* (STP)
- *Service control point* (SCP)

Service Switching Point (SSP)

The SSP is the local exchange in the telephone network. An SSP can be the combination of a voice switch and SS7 switch or an adjunct computer connected to a voice switch. Using adjuncts enables telephone companies to upgrade their SS7 signaling points without replacing expensive switches, supporting a modular approach. Upgrades are typically limited to software loads because these devices require very little hardware. With the advent of packet telephony, this has become less of an issue because new-generation switches are server based and a fraction of the cost of legacy circuit-switched equipment. The SSP function is now found in multiple devices throughout the packet-switched network.

The SSP communicates with the voice switch using primitives. The SSP must convert signaling from the voice switch into SS7 signaling messages, which then can be sent to other switches in the network (see Figure 2.1). The switch will typically send messages related to its voice circuits to the switches with a direct voice trunk connection to it.

In the case of database access, the SSP will send database queries through the SS7 network to computer systems located centrally to the network (or regionally).

The SSP function uses the information provided by the calling party (such as dialed digits) to determine how to connect a call. A routing table in the switch itself will identify which trunk circuit or *Transmission Control Protocol* (TCP) socket to use to connect the call and at which switch this trunk terminates. An SS7 message must be sent to this adjacent switch requesting a circuit connection on the specified trunk or socket.

The circuit identification (referred to as the *circuit identification code* [CIC]), the calling and called telephone numbers, and information about the voice transmission method used are in this SS7 message. There is also information about the type of call, the type of decoding used in the voice transmission, and possibly any switch features needed during the call.

The adjacent switch grants permission to connect this trunk or socket by sending back an acknowledgment to the originating switch. Using the called party information in the setup message, the adjacent switch can then determine how to connect the call to its final destination. The same process is followed using a setup message to any adjacent switches and circuits connecting those switches. The entire call may require several connections between several switches. The SSP function in each switch manages these connections, but really has no knowledge of the status of remote connections (nonadjacent connections). The SSP only has visibility of its own connections and does not maintain the status of all the connections needed to connect and maintain a call.

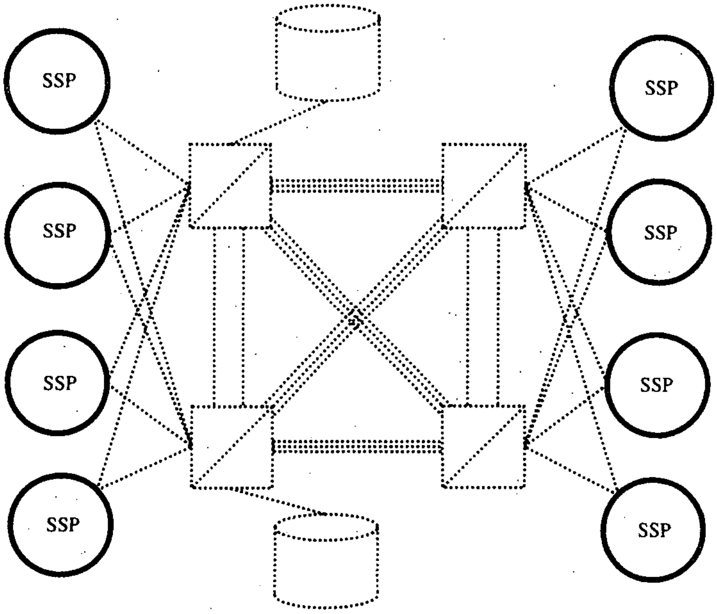


Figure 2.1 This figure shows the relationship of the SSP to the SS7 network.

Very few SS7 features are required of an SSP. The capability to send messages using the *ISDN User Part* (ISUP) protocol and the *Transaction Capabilities Application Part* (TCAP) protocol is the only requirement besides the network management functions. SSPs are responsible for the management of ISUP messages specifically, which may include different variants of ISUP standards.

Specific Telcordia requirements for an SSP can be found in Telcordia publication GR-024-CORE, "Service Switching Points (SSPs) Generic Requirements."

Signal Transfer Point (STP)

All SS7 packets travel from one SSP to another through the services of an STP. The STP serves as a router in the SS7 network. To maintain redundancy and diversity in the network, STPs are always deployed in pairs. Should one STP node fail, the other node assumes all the traffic. Both STPs in a pair process traffic using load sharing.

An STP does not usually originate messages. In fact, the STP does not process any upper-layer protocols (ISUP or TCAP, for instance). The STP only processes the transport layers (*Message Transfer Part* [MTP] or TCP/IP-based protocols). The STP routes SS7 messages as received from the various SSPs through the network to their appropriate destinations (see Figure 2.2).

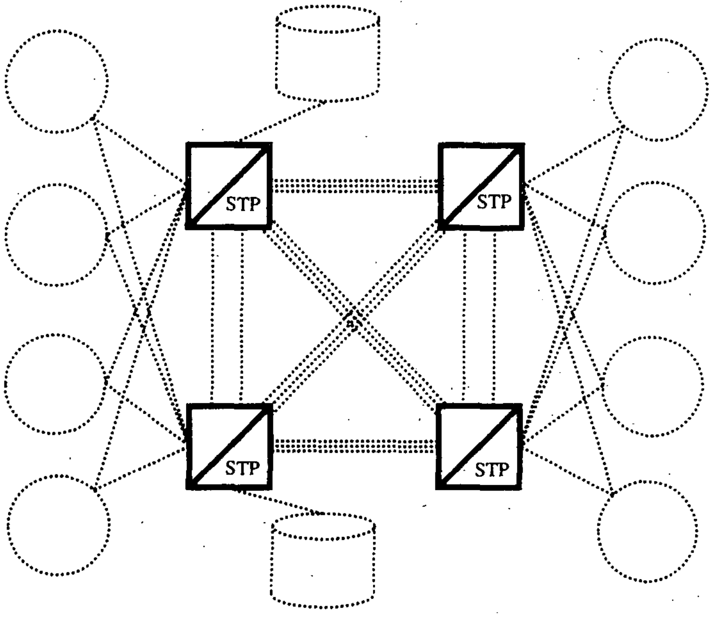


Figure 2.2 This figure shows the relationship of the STP to the SS7 network.

The STP can be an adjunct to a voice switch or a standalone packet switch. Many tandem switches provide both switching functions and STP functions (through the use of an adjunct computer). Although several manufacturers provide STP equipment, very few provide a standalone STP. A standalone solution offers many benefits.

By using a standalone STP, companies can centralize many important revenue-generating functions on their network. This includes services provided by SCPs and the monitoring of the entire SS7 network. Updates to SS7 are much simpler and more cost-effective when standalone STPs are used because the entire switch must be upgraded when this function is integrated into a tandem switch. Many carriers have changed their network architecture, replacing switched STPs with standalone STPs for this very reason.

In large networks, multiple STPs are sometimes deployed in a hierarchical fashion. In these cases, three levels of STPs can be used:

- National STP
- International STP
- Gateway STP

The national STP exists within a national network and is capable of transferring messages using the same national standard of protocol. Messages may be passed to another level of STP, the international STP, but the national STP has no capability to convert messages into another version or format. The conversion is typically MTP only, as STPs do not process the upper layers of the SS7 protocol. At least one manufacturer provides the protocol conversion on the STP itself, eliminating the need for two STPs (one national and one international).

The international STP functions the same as the national STP, but is used in the international network. The international network provides the interconnection of all countries using the ITU-TS standards for MTP. This ensures interconnectivity between worldwide networks despite the use of different point code structures and network management. All nodes connecting to the international STP must use the ITU-TS protocol standard.

The gateway STP serves as the interface into another carrier's network. For example, long-distance service providers may have access into a local telephone company's database for subscriber information, or the local service provider may need access into the long-distance service provider's database. In any event, this access is accomplished through a gateway STP.

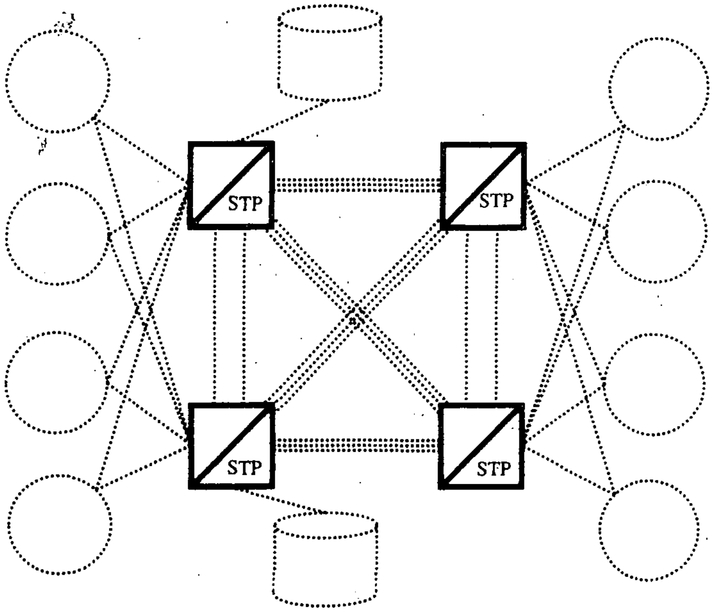
The gateway STP may also provide protocol conversion from a national MTP standard to the ITU-TS MTP standard or some other national standard. A gateway STP is often used to access the international network, providing access and the conversion of messages to the ITU-TS protocol standard. This eliminates the need for adjunct protocol converters on the network. Gateway STPs must be able to work using both the international standard of protocol and the national standard, depending on the location of the STP.

Gateway STPs are also used in wireless networks. Many wireless networks use X.25 as a transport protocol between their *mobile switching centers* (MSCs) and databases. The X.25 networks were private networks and did not provide the support for accessing other wireless providers' networks.

X.25 networks have other limitations that do not lend themselves well to the applications of the wireless network. All information transfers in X.25 networks require connection-oriented services. Trying to connect to multiple networks and entities in various locations can become somewhat cumbersome. Many messages that originated in wireless or SS7 networks do not require connection-oriented services.

For this reason, many wireless providers have eliminated the old X.25 networks and replaced them with SS7. The MSCs now exchange information about the location of mobile phones and update their databases using the TCAP protocol, which is much better suited to this task. More importantly, SS7 has enabled wireless providers to share their location databases with other wireless carriers, supporting roaming arrangements that would not be possible otherwise.

The purpose of the gateway STP is to provide a single interface into a network for network security. The gateway STP uses screening features to determine which carriers are allowed access to the network. *Screening* is the capability to examine all incoming and outgoing packets and allow only those SS7 messages that are authorized. This is determined through a series of gateway screening tables that must be configured by the service provider. Gateway screening also prevents messages from unstable networks



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In the international network, gateway STPs may provide an additional function. International SS7 is based on ITU-TS standards, yet every country uses a national version that is not 100 percent compliant with the ITU-TS standard, For example, in the United States, we are ITU compliant, yet the ITU-TS standards have been modified for usage here. The major difference between the two standards is in addressing and the network management functions of the protocol.

The gateway STPs that connect us to other countries do not deviate from the ITU-TS standards. They must be 100 percent compliant. They can perform a protocol conversion, enabling ITU-TS messages into the network by converting the messages into the national format before transferring them into the network. This is true gateway functionality, yet not all gateway STPs must perform protocol conversion.

Routing to databases is another important STP function. Although a carrier may only have one SCP, this SCP may provide many services (see Figure 2.3). For example, the SCP may support calling cards, 800 routing, and calling name display. Each of these services in the SCP must have a unique address. The STP uses a function called *global title translation* (GTT) to determine which database will receive the query generated by SSPs.

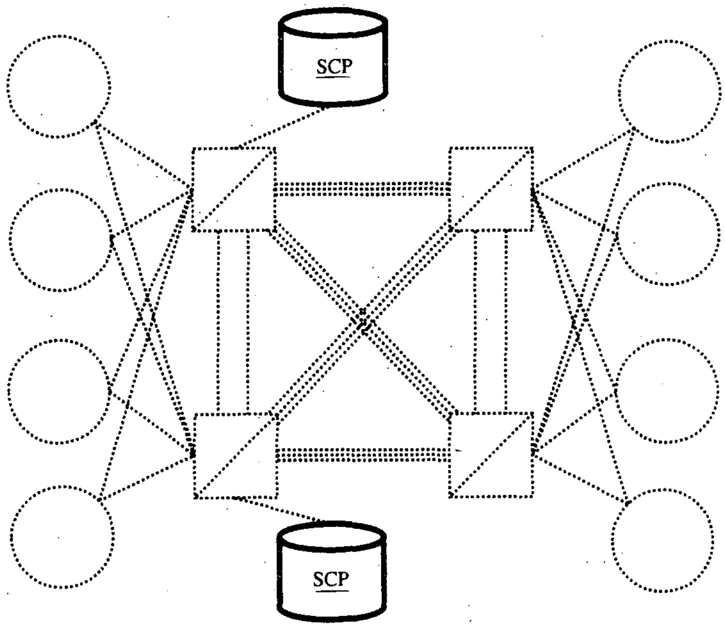


Figure 2.3 This figure shows the relationship of the SCP to the SS7 network.

It may seem easier for the SSP to know the specific address of each and every database in the network, but this increases the amount of administration that must be managed. Every time a database is added or changed, the address information would have to be added to every SSP on the network. By using an STP and GTT, the SSPs only need to know the address of the STP. By using this type of architecture, SCPs can be changed or added, affecting only the STP.

In larger networks, there may be multiple GTTs. STPs that are deployed regionally may provide a partial GTT (routing the query to a central STP) that provides the final GTT. The purpose of using partial and final GTT is to eliminate unnecessary administration on the network.

This is important to hub providers as well. Imagine if you were a hub provider who supported many SCPs and had to notify hundreds of carriers to update the routing information in their SSPs to reflect the new addressing information. By using the global title feature in an STP, the changes to the hub provider network are completely transparent to the carriers.

The SSP sends a database query to the local STP with the destination address of the STP. The STP looks at the dialed digits in the *Signaling Connection Control Part* (SCCP) portion of the *message signal unit* (MSU) (or *global title digits* as they are often called) and determines the address of the database through its own translation tables. The GTT consists of the subsystem number (address) of the database and the point code of the SCP that interfaces with the database.

An STP is the most important investment carriers can make for their networks. Carriers who use hub providers rather than investing in their own STPs are missing out on the opportunity to be more responsive to their subscribers and provide unique services that are not provided by hub providers.

For specific information regarding Telcordia requirements for an STP, refer to the Telcordia publication GR-082-CORE, "Signal Transfer Point (STP) Generic Requirements."

Service Control Point (SCP)

The SCP serves as an interface to telephone company databases. These databases provide the storage of information about subscribers' services, the routing of special service numbers (such as 800 and 900 numbers), calling card validation, and even *Advanced Intelligent Network* (AIN) services.

The SCP is actually a function of the computer used as a front end to the database application itself. The SCP does not necessarily have to be a standalone computer system. Some new SCP database applications are being implemented in STPs, providing an integrated solution. In all cases, the address of the SCP is a point code, whereas the address of the database is a subsystem number. Entities within the network route to SCPs using the SCCP protocol.

The SCP function does not necessarily store all the data, but it is the interface to the mainframe or minicomputer system that is used for the actual database. These computer systems are usually linked to the SCP through X.25 or IP links.

The SCP communicates with the database application through the use of primitives. A primitive is an interface that provides access from one level of the protocol to another level. In the case of the database, the database is considered an application entity, and TCAP is the protocol used to access and interface with this application entity.

The type of database depends on the network. Each service provider has different requirements, and their databases will differ. Telcordia has defined some basic models of databases for achieving the network needs of the *Regional Bell Operating Companies* (RBOCs). In addition to the RBOC networks, wireless providers also use databases for the storage of subscriber information. The databases most commonly used within either of these networks are as follows:

- *Call management services database* (CMSDB)
- *Local Number Portability* (LNP)
- *Line Information Database* (LIDB)
- *Calling name* (CNAM)
- *Business services database* (BSDB)
- *Home location register* (HLR)
- *Visitor location register* (VLR)

Each database is given a unique address called a *subsystem number*. The subsystem number is used to route queries from SSPs through the SS7 network to the actual database entity. The subsystem numbers are defined by the service provider and are fixed.

The following databases are not absolutes; in other words, not every network must have these specific databases. These are the databases used within the networks of the BOCs and they are mentioned here as a model of database types.

Call Management Services Database (CMSDB)

The CMSDB provides information relating to call processing, network management, and call sampling (for traffic studies). The call-processing portion defines the routing instructions for special service numbers such as 800, 976, or 900 numbers. In addition to routing instructions, this database also provides billing information, such as the billing address or third-party billing procedures.

CMSDB also provides certain network management functions used to prevent congestion on the network. When congestion occurs on the SS7 network, this database can provide important routing instructions for rerouting messages around the congested node.

Call sampling is used to create reports that indicate the types of telephone calls being made in the telephone network. These reports are then used in traffic studies to determine if additional facilities are needed to handle the voice traffic. The *service management system* (SMS) schedules reports for automatic printing and enables administration personnel to update the database records through a terminal interface.

Local Number Portability (LNP)

The Communications Act of 1996 mandated LNP. The purpose of LNP is to enable subscribers to change telephone companies without having to change their telephone numbers. This, of course, changes the way telephone calls have been routed for years.

Telephone numbers were assigned to carriers in blocks. These number blocks were then assigned to switches in the network (also in blocks). For example, the NNX of 550 (and all of the telephone numbers 550-0000 through 550-9999) may have been assigned to a rural switch. The switches within the telephone network were then programmed to route all calls to telephone numbers in this range to the specific switch.

With LNP, a subscriber may decide to change to another carrier. The new carrier must then assign the subscriber's telephone number to its own switch. This, of course, means that the old (donor) carrier must remove the one telephone number from its own switch (which is usually flagged as vacant).

When a call is initiated, the originating exchange must first search its routing table to determine if the called number has been flagged as ported. If the NNX has been flagged as ported, then a query must be sent to an LNP database to determine if the actual called number has been ported and, if so, how the call is to be routed. Even if only one telephone number within an NNX has been ported, the entire NNX is considered ported and a query must be generated for each and every call made to that NNX.

If the called number has been ported, then the database will identify the new terminating switch by giving its *local routing number* (LRN). The LRN works the same as the NNX code did, providing a unique identity for each and every exchange in the network. This information is then returned to the originating exchange so the call can be routed. If the telephone number has not been ported, then the call is routed normally.

LNP also impacts how numbers are assigned. A clearinghouse assigns telephone numbers to telephone companies. Numbers are issued in much smaller blocks than before. A database is used for number pooling as well. The concept of number pooling is very similar to LNP, but in this case the database identifies the carrier to which a particular number has been assigned.

The implementation of LNP is now taking place in international networks. Spain has already implemented LNP on their networks, and many other European countries are quickly following. International number portability presents many new challenges to vendors because every country uses a different implementation, so what may work in Spain will most likely not work in Great Britain.

Line Information Database (LIDB)

The LIDB provides information regarding subscribers, such as calling card service, third-party billing instructions, and originating line number screening. Billing is the most important feature of this database. Third-party billing instructions, collect call service, and calling card service all determine how subscribers will be billed for their telephone calls in real time.

In addition to billing instructions, the LIDB also provides calling card validation, preventing the fraudulent use of calling cards. The user's *personal identification number* (PIN) is stored in this database for comparison when a user places a call.

Originating line number screening provides information regarding custom calling features such as call forwarding and speed dialing. These features carry from network to network as each service provider provides its own distinct calling services.

Calling Name (CNAM)

SS7 automatically enables carriers to provide their subscribers with the calling party number because this information is carried in call setup messages. However, the name of that calling party is not included. This database provides the name of the calling party based on information from LIDB databases.

The RBOCs are typically the owners of these databases; however, a number of companies are looking to provide their own databases by using many sources. There are also a number of companies providing the database entries themselves and updates to CNAM databases are given weekly. Some of these information providers have more accurate information than the RBOCs themselves.

CNAM is a very popular feature and a great revenue-generating service for any carrier. The subscriber must possess a specially equipped telephone or external display device to receive the information sent from the end-office switch, but this equipment can be found for a very reasonable cost in most retail stores that sell telephones.

Business Services Database (BSDB)

This database is mentioned only as a model because the last publication of Telcordia recommendations still has not defined the applications for this database. The purpose of this database is to enable subscribers to store call-processing instructions, network management procedures, and other data relevant only to their own private network. The telephone company could offer this database as an extra service, enabling large corporate customers to create their own private networks linking *private branch exchange* (PBX) equipment across the country. With their own proprietary databases, corporations can alter traffic routing by the time of day or congestion modes without altering software in the PBX equipment.

This type of database would be a valuable asset to companies using the flexibility of *service creation environments* (SCEs). The network management function can provide special routing instructions for calls destined to congested PBXs, a feature popular with inbound call centers using *automatic call distributors* (ACDs).

Home Location Register (HLR)

The HLR is found in wireless networks and is used to store information regarding a wireless subscriber. Subscribers are assigned to a home area under the control of the home HLR. Billing and feature information is stored in this database along with location information. The HLR identifies the MSC (the wireless switch) currently providing service to a subscriber. The actual location (cell site) of the subscriber is found in a dynamic database called the VLR.

When a wireless telephone is activated, a cell site receives a signal from the cell phone containing the *mobile identification number* (MIN) and other identification. The MIN is equivalent to a *Plain Old Telephone Service* (POTS) number. This information is sent to an MSC, which then determines the HLR to which the MIN belongs (using an internal routing table).

Every few minutes, the wireless phone resends this signal. As the car moves from one cell site to another, the MSC monitors which MSC is servicing the subscriber, and if the subscriber moves to an area serviced by another MSC, the HLR database is updated.

When a call is received into the network for a mobile telephone, the home HLR must determine which MSC to route the call to. The home HLR informs the MSC of the location, and the call is connected using voice circuits through the appropriate MSC to the cell site servicing the wireless subscriber at that time.

Visitor Location Register (VLR)

The VLR is used for determining the location of a wireless subscriber in real time. As a subscriber roams outside of his or her home network, the MSCs will not have any information regarding the subscriber. A query is sent to the subscriber's home HLR, which authenticates the subscriber and sends pertinent information to the roaming network.

The VLR is then updated with this information. As long as the subscriber is active in this network, the information remains in the VLR. This enables the MSC to keep data about the roaming subscriber without having to make changes to its own HLR. Think of the VLR as temporary database storage for visiting wireless subscribers.

The VLR may or may not be colocated with every MSC. In some networks, there is one VLR and one HLR. The only requirement is that all MSCs need to be able to access all HLR and VLR databases using the SS7 protocol.

For specific Telcordia requirements relating to SCPs, refer to the Telcordia publication TR-NWT-001244, "Supplemental SCP."

Operations Support Systems (OSS)

The entire telephone network relies on back-office equipment for provisioning, maintenance, and other administrative functions. Monitoring the signaling network is crucial to ensuring service without interruption. Problems in the signaling network mean service interruptions for subscribers and, in some cases, complete network failures.

The BOCs have established remote maintenance centers for the monitoring and management of their SS7 and voice networks (see Figure 2.4). The remote maintenance centers give them a view of the entire network from one central location and enable them to quickly identify and troubleshoot network problems without dispatching technicians. This is especially helpful where equipment is located in an unmanned central office or a collocation facility.

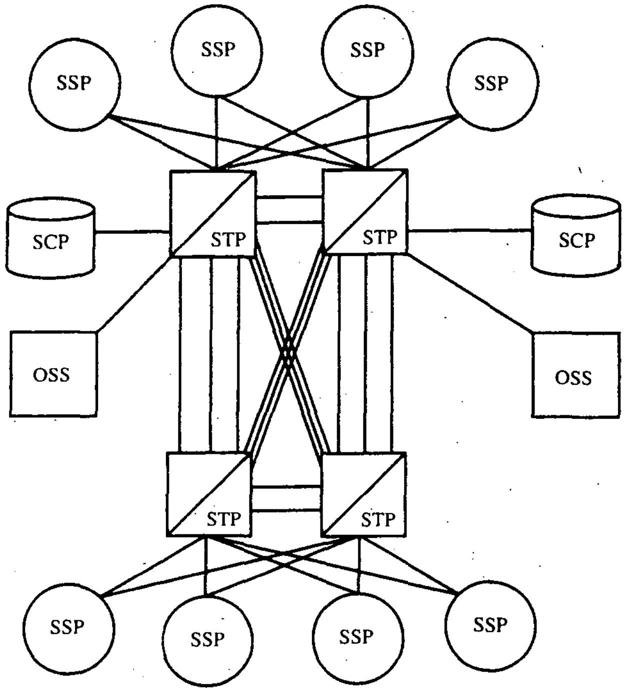


Figure 2.4 The OSS is typically adjacent to an STP and accesses the network via a signaling link to the STP. Using TCAP and SCCP, the OSS is then capable of sending an SS7 message to any entity within its own network.

Telcordia has defined a standard set of commands for use in all of their network equipment, SS7 and otherwise, enabling their maintenance personnel to learn one set of commands for accessing all devices on the network. This eliminates the need for training on specific equipment. This has proven to be an expensive alternative for most carriers and has not been widely accepted, although it is a very sound concept.

Some vendors have developed *graphical user interfaces* (GUIs) for provisioning systems that enable carriers to use diverse mixes of equipment as well as access and provision the equipment through the GUI. This eliminates the need for a common command set, which limits the vendor choice for the carrier.

To update the SCP databases and monitor the performance of the databases, an SMS is used. The SMS is a standard interface consisting of a command set and graphical interface that can be used to administrate the database, monitor the status of the database, and retrieve measurements pertaining to performance from the database. The SMS also provides a central point for making updates to multiple databases. The database changes are made within the SMS and then propagated to all the databases in the network. This eliminates the need to visit each and every database site to incorporate new changes and ensures consistent database updates. It is also an important method to ensure that updates have been tested prior to being sent to all the database elements in the network.

Network Monitoring

Network-monitoring systems have become important tools to all carriers. Even carriers who do not have their own SS7 networks need visibility to what is coming into and leaving their networks.

The monitoring system serves two functions. The first is to provide visibility to the entire network and everything that goes on within the network. The other function is data collection.

Data collection is still somewhat of a new concept for network-monitoring systems. Although they have had this capability for a while, the concept of capturing signaling data for use other than maintenance and troubleshooting is slowly catching on.

The monitoring system serves as a central interface to this data, using probes connecting to every signaling link in the network to access the data. The MSUs are copied and stored on a server for historical traces and analysis.

Monitoring systems can also be used for creating *call detail records* (CDRs). Each time a call is completed, the switch that originated the call creates a CDR. However, switches are not a dependable source for CDRs for a variety of reasons.

Signaling networks, on the other hand, are the best source for CDRs. Every call generates a signaling message, which is captured by the monitoring system. The monitoring system can then be used to interface to a number of business applications.

Fraud detection/management is one of the more popular applications used in this scenario. Revenue assurance groups use this to detect and stop fraud of all types in both wireline and wireless networks.

Traffic analysis tools use CDRs as well as traffic reports generated by the monitoring system to ensure *Quality of Service* (QoS), which may be of significant importance when a carrier has *service level agreements* (SLAs) with customers or other carriers.

The most powerful use of signaling data is data mining. When you examine the contents of a signaling message, you find virtually everything a carrier needs to know about the subscribers using the network. Data mining is used by marketing, sales, engineering, operations, and finance departments. It is by far the most important tool a carrier can have for revenue assurance.

Signaling Data Links

All signaling points are interconnected via signaling data links. These links are 56/64 kbps, 10/100 Mbps, and 1.536 Mbps data facilities (the exception to this rule is Japan, where 4.8 kbps links are used). Links are bidirectional, using transmit and receive pairs for simultaneous transmission in both directions.

Links should always be terrestrial, although satellite links are supported in the standards. Satellite links are unfavorable because of the delay introduced. In the event that satellite links are used, the labeling and functions of the links remain the same.

Network management procedures are the same except for the procedures used at level 2 of the protocol (link alignment and error detection/correction).

Satellite links use a different method of error detection/correction than terrestrial links. Basic error detection/correction is used for all terrestrial links, and *preventive cyclic retransmission* (PCR) is used for satellite links. The difference between the two lies in the retransmission mechanism. In basic error detection/correction, an indicator bit is used to indicate retransmission. In PCR, if an acknowledgment is not received for transmitted signaling units within a specific time, all unacknowledged signaling units are retranslated.

There are three modes of signaling. These three modes depend on the relationship between the link and the entity it services. The simplest mode is referred to as *associated signaling*. In associated signaling, the link is directly parallel with the voice facility for which it is providing signaling. This, of course, is not the ideal because it would require a signaling link from the end office to every other end office in the network. However, some associated modes of signaling do exist (mostly in Europe).

Nonassociated signaling uses a separate logical path from the actual voice, as shown in Figure 2.5. Multiple signaling nodes are usually used to reach the final destination, while the voice may be a direct path to the destination. Nonassociated signaling is commonly used in many SS7 networks.

Quasi-associated signaling (see Figure 2.6) uses a minimal number of nodes to reach the final destination. This is the most favorable method of signaling because each node introduces additional delays in signaling delivery. For this reason, SS7 networks favor quasi-associated signaling.

Signaling data links (see Figure 2.7) are labeled according to their relationship in the network. There is no technical difference between the various links; they are only different in the way the links are utilized during message transfer and how network management interacts with the links (see Figure 2.8).

Links are placed into groups called *linksets*. All the links in a linkset must have the same adjacent node. The switching equipment alternates transmission across all the

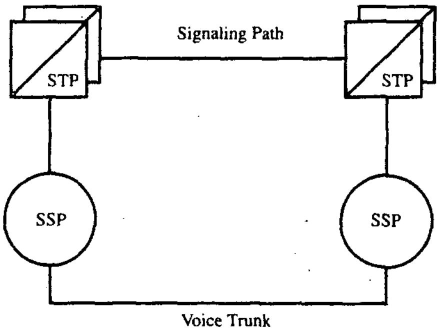


Figure 2.5 Nonassociated signaling involves the use of STPs to reach the remote exchange. As depicted in this figure, to establish a trunk connection between the two exchanges, signaling messages would be sent via SS7 and STPs to the adjacent exchange.

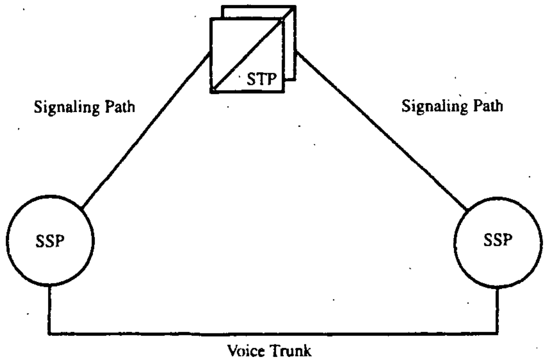


Figure 2.6 In quasi-associated signaling, both SSPs connect to the same STP. The signaling path is still through the STP to the adjacent SSP.

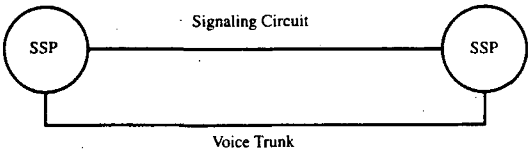


Figure 2.7 In some cases, it may be better to directly connect two SSPs via a signaling link. All SS7 messages related to the circuits connecting the two exchanges are sent through this link. A connection is still provided to the home STP using other links to support all other SS7 traffic.

links in a linkset to ensure the equal use of all facilities. Up to 16 links can be assigned to one linkset.

In addition to linksets, a signaling point must define routes. A route is a collection of linksets used to reach a particular destination. A linkset can belong to more than one route. A collection of routes is known as a *routeset.*

A routeset is assigned to a destination. Routesets are necessary because if only a single route existed and that route became unavailable, an alternate route would not be defined and no signaling could be sent to that destination. A routeset provides alternate routes to the same destination in the event that any one route becomes unavailable.

A destination is an address entered into the routing table of a signaling point. The destination does not have to be directly adjacent to the signaling point, but it must be a point code that can be reached by the signaling point. A signaling point does not have to know all point codes in between itself and its destinations; it only has to know which link or linkset to use to reach its destination. A signaling point can have multiple addresses if it is necessary to partition a signaling point into multiple functions. For example, a gateway STP used to enter the international network may have multiple point codes: one for the gateway function and another for GTT services.

It may sound as if an STP would have to know millions of point codes, but in actuality this is not the case. If a carrier connects to another carrier's network, the STP only

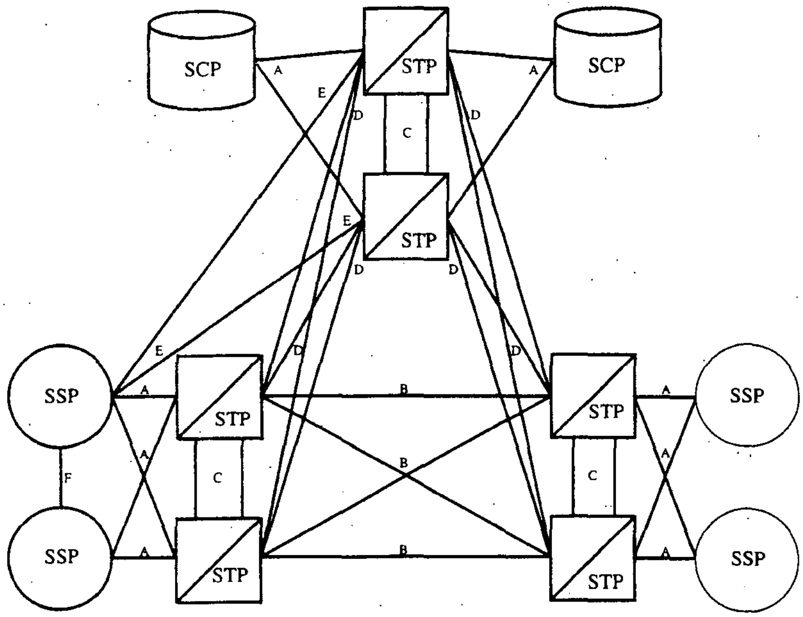


Figure 2.8 All signaling links are labeled according to their location on the network. There is no real physical difference between different links other than network management treatment.

needs to know the point codes on that network. It becomes the other carrier's responsibility to make sure that connections are made on other networks.

Link Implementation

The secret to the SS7 network and to making sure the network is always operational is to provide alternate paths in the event of failures. These alternate paths provide the reliability needed in a network of this nature and ensure that SS7 messages always reach their destinations (see Figure 2.9).

When a node has links to a mated STP pair, the links are assigned to two linksets, one linkset per node. Both linksets can then be configured as a combined linkset. A combined linkset contains links to both STP pairs, which means their adjacent signaling point address (point code) will be different. Combined linksets are used for load sharing, where the sending signaling point can send messages to both pairs, spreading the traffic load evenly across the links (see Figure 2.10).

Alternate linksets are used to provide alternate paths for messages. An alternate linkset or link is defined in the signaling points' routing tables and used when congestion conditions occur over the primary links. Figure 2.11 illustrates a typical configuration using alternate links to other nodes in the network, providing diversity in the event of node congestion.

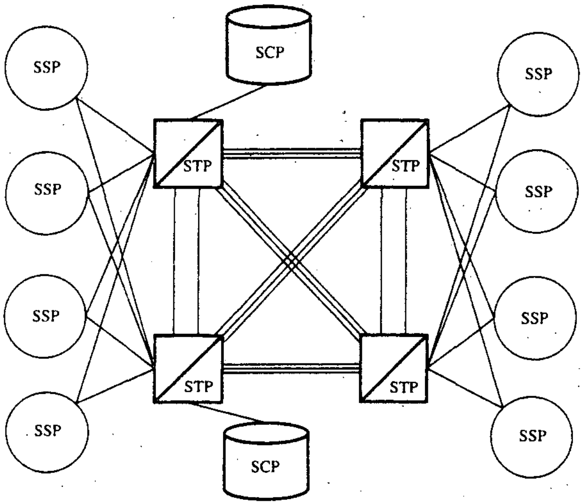


Figure 2.9 This figure depicts a typical SS7 network with multiple *bridge links* (B-links) and *access links* (A-links).

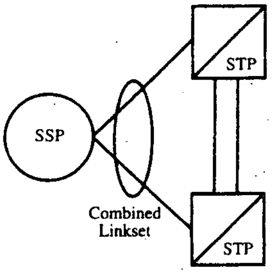


Figure 2.10 A combined linkset connects an adjacent pair. Each link in this figure connects to a different signaling point, but has the same destination.

Links are labeled according to their relationship on the network. Although there is no technical difference between these different links, there are differences in how the links are engineered. There are six different types of links used in SS7:

- A-links
- B-links
- *Cross links* (C-links)
- *Diagonal links* (D-links)
- E-links
- F-links

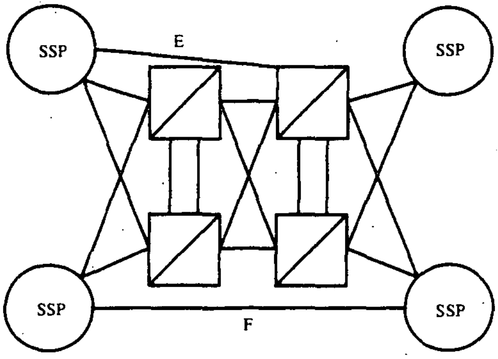


Figure 2.11 In this figure, the *extended links* (E-links) and *fully associated links* (F-links) are alternate links and would be used when the primary links become unavailable or congested.

Access Links (A-Links)

A-links (see Figure 2.12) are used between the SSP and the STP, or the SCP and STP. These links provide access into the network and to databases through the STP. There are always at least two A-links, one to each of the home STP pairs. In the event that STPs are not deployed in pairs, there can be one A-link; however, this is highly unusual.

The maximum number of A-links to any one STP is 16. A-links can be configured in a combined linkset that has 16 links to each STP, providing 32 links to the mated pair. When connecting switches in a network to hub providers, A-links are used.

When trying to determine how many A-links are required, the easiest formula is to calculate the number of access lines supported by the switch. One simple formula is to calculate one signaling link for every 9,600 access lines. Many other formulas are used, but this simple formula will provide close enough results for general usage.

Bridge Links (B-Links)

B-links are used to connect mated STPs to other mated STPs at the same hierarchical level. B-links are deployed in a quad fashion, as shown in Figure 2.13, which is why these are often referred to as quad links. A maximum of eight B-links can be deployed between mated STPs. Although this practice is closely followed in North America, European networks do not use B-links as depicted. Mated STPs are connected to another mated pair via one set of links, but each STP does not have a connection to each of the other mated STPs.

Cross Links (C-Links)

C-links (see Figure 2.14) connect an STP to its mate STP. They are always deployed in pairs to maintain redundancy on the network. Normal SS7 traffic is not routed over these links, except in congestion conditions. The only messages to travel between mated

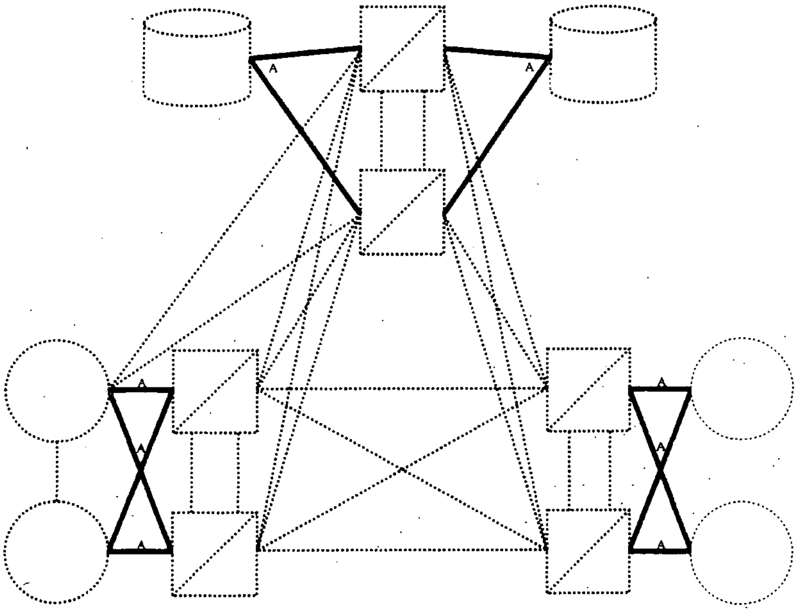


Figure 2.12 A-links connect end-signaling points to the SS7 network.

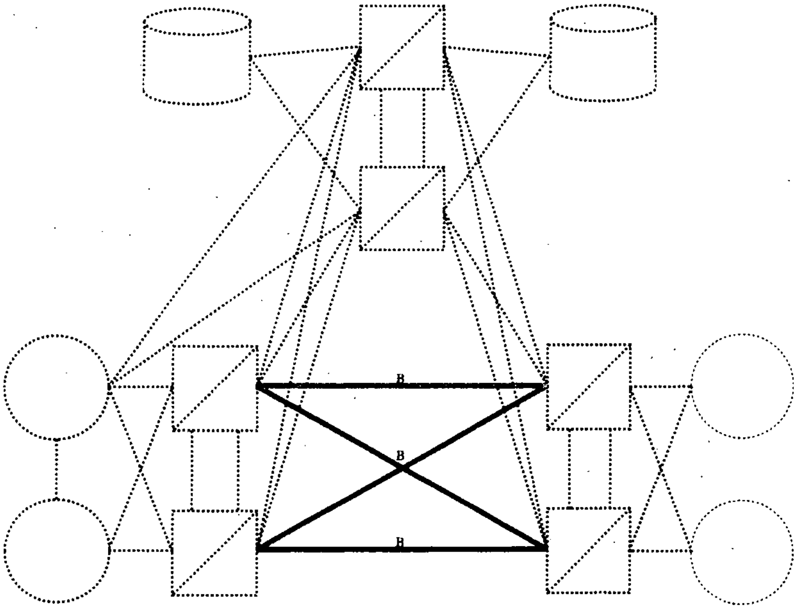


Figure 2.13 B-links connect a mated pair of STPs to another mated pair of STPs.

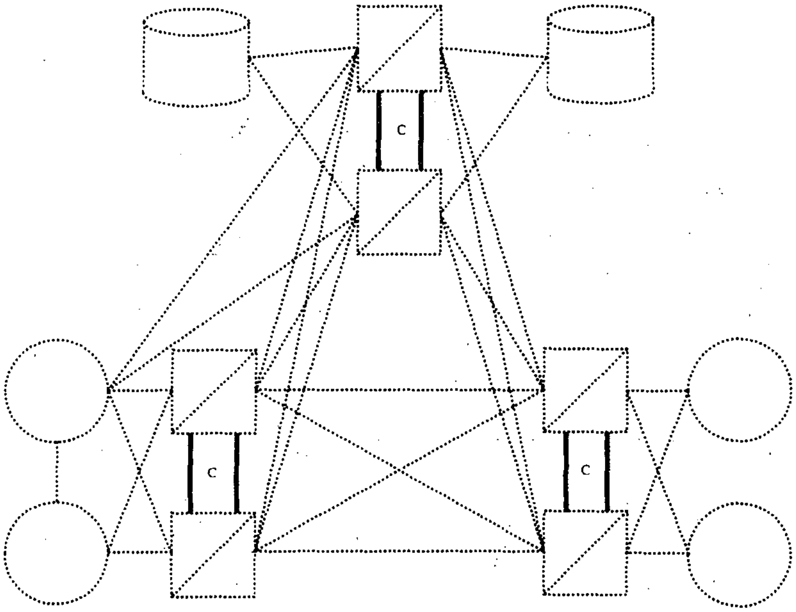


Figure 2.14 C-links connect an STP to its mate STP, creating a mated pair. Mated pairs are identical in function and configuration, and have the capability to assume the traffic of their mate in the event that the mate fails. C-links are used to share network management messages and, when no other route is available, to signal traffic.

STPs during normal conditions are network management messages. If a node becomes isolated and the only available path is over the C-links, then normal SS7 messages can be routed over these links. A maximum of eight C-links can be deployed between STP pairs.

Diagonal Links (D-Links)

D-links (see Figure 2.15) are used to connect mated STP pairs at a primary hierarchical level to another STP mated pair at a secondary hierarchical level. For example, a carrier may have STPs deployed in every *Local Access Transport Area* (LATA). They could then deploy STPs in regions, acting as concentrators. This would prevent the need to interconnect every STP to every other STP. The LATAs within a defined region would all connect to one STP, which would provide connections to the other regional STPs. This hierarchical approach would only be found in very large SS7 networks.

Not all networks deploy D-links because not all networks use a hierarchical network architecture. D-links are deployed in a quad arrangement like B-links. A maximum of eight D-links can be used between two mated STP pairs.

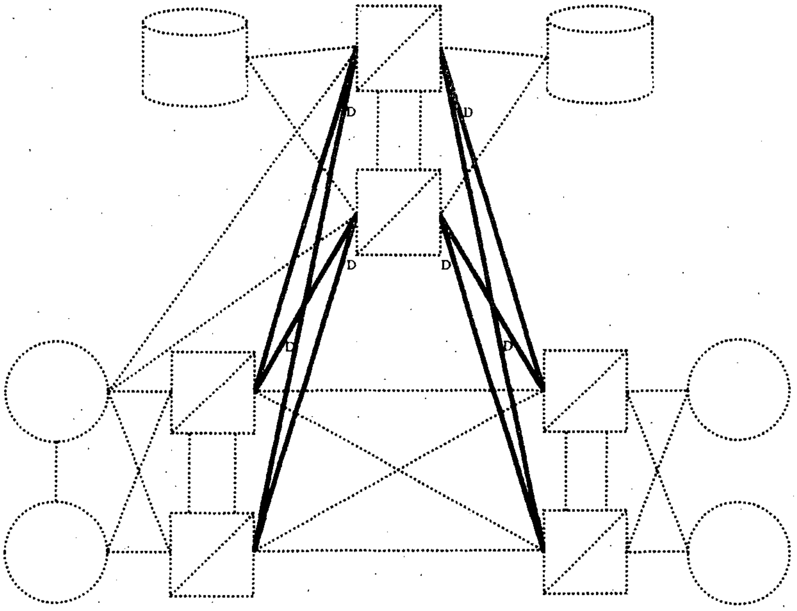


Figure 2.15 D-links connect mated STPs to another mated pair of STPs that are deployed on a higher level in the network hierarchy. In this figure, the D-links are used to connect to a pair of regional STPs, which provide access to a regionally located database.

Extended Links (E-Links)

E-links (see Figure 2.16) are used to connect to remote STP pairs from an SSP. The SSP connects to its home STP pair, but, for diversity, may also be connected to a remote STP pair using E-links. E-links then become the alternate route for SS7 messages in the event that congestion occurs within the home STP pairs. A maximum of 16 E-links can be used between any remote STP pairs.

Fully Associated Links (F-Links)

F-links (see Figure 2.17) are used when a large amount of traffic exists between two SSPs or when an SSP cannot be connected directly to an STP. F-links enable SSPs to use the SS7 protocol and access SS7 databases even when it is not economical to provide a direct connection to an STP pair.

When traffic is particularly heavy between two end offices, the STP may be bypassed altogether, provided that both SSPs are local to each other. Only call setup and teardown procedures would be sent over this linkset.

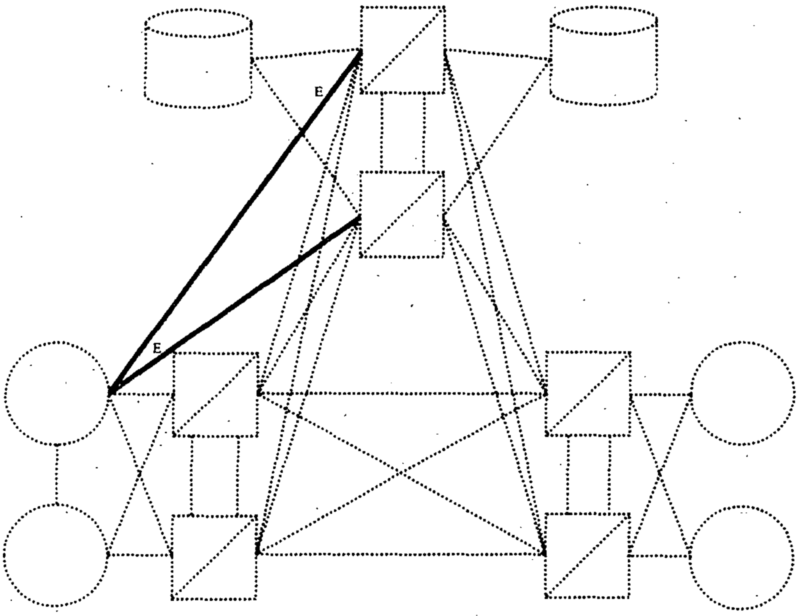


Figure 2.16 E-links connect an SSP with an STP that is not considered its home STP. This is done for diversity and provides an alternate route around its home STP pair. This configuration can also be used when there is a high volume of traffic to a particular destination to prevent the home STP pair from becoming congested.

Link Performance

Links must remain available for SS7 traffic at all times and have minimal downtime. When a link fails, the other links within its linkset must take the traffic. Likewise, if an SS7 entity (such as an STP) fails, its mate must now assume the load. This means that links can suddenly be burdened with more traffic than they can handle. For this reason, SS7 links are engineered for 40 percent bandwidth. The links will obviously handle 100 percent capacity, but when existing links approach 40 percent capacity, additional links should be added.

To determine when additional links are needed, traffic studies should be used. This is no different than with voice circuits. Consistent monitoring and measurements of all links in the network are crucial to understanding the needs of the network.

You can calculate the number of messages that a link will support by first determining the average length (in bytes) of each message. For example, if a DS0 link is being used, the transmission rate of that link is 56 kbps. If you divide 56,000 bits into 1 byte (8 bits), you will see that a link can support 7,000 bytes per second. Given that constant, you must now consider that the link is engineered to carry 40 percent traffic, which comes to 2,800 bytes per second.

If the average message length is 40 bytes (as is the case with most ISUP messages), your link can carry 70 ISUP messages per second (2,800/40×70). You can use this sim-

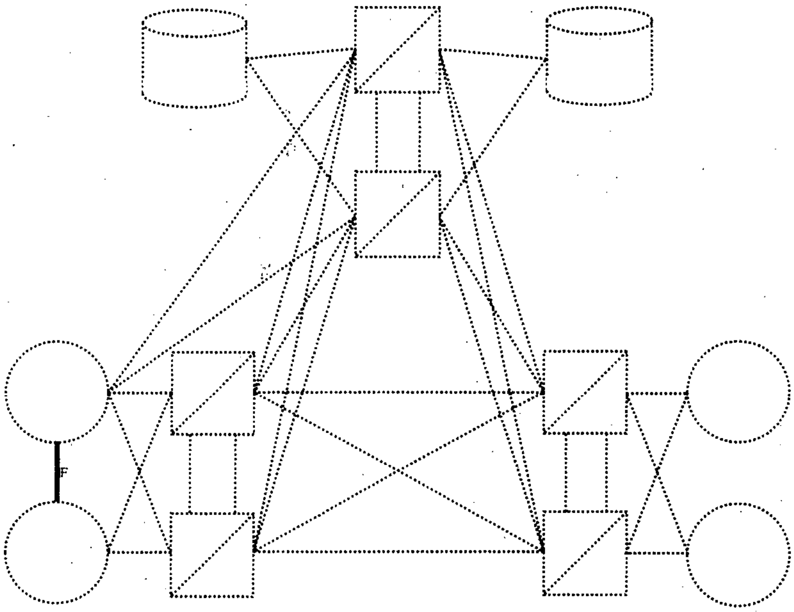


Figure 2.17 F-links connect two STPs directly, enabling signaling traffic to follow the same path (in parallel) as the voice circuits. This is commonly found when there is a large volume of signaling traffic (ISUP) between two exchanges. This configuration can also be used when there is no direct access to the SS7 network.

ple formula to calculate the capacity of any link, which becomes important when you are sizing your network.

Calculating link utilization is just as simple. Given the same rules we already discussed, you can use the following formula. First, you will need to know the number of bytes (or octets) sent or received on a link over a specific time frame. Choose any interval, but for this example, we will use 15 minutes. You must also know the link speed (that is, kbps or 64 kbps).

Let's say that over a 15-minute interval there were 151,694 octets sent and received over a 15-minute interval on a 56 kbps link. The equation would then be the following:

$(x/(y/8 \times 60 \times m)) \times 100$ where

x = the number of octets sent and received

y = the link speed

$8 = 8$ bits per octet or byte

$60 = 60$ seconds in a minute

m = the number of minutes

Therefore, the equation for our example would look like the following:

$$\{(151{,}695)/(56{,}000/8 \times 60 \times 15)\} \times 100 = 2.4 \text{ percent utilization}$$

There are also rules regarding the amount of downtime allowed on a link. A maximum of 10 minutes downtime a year is allowed for any one linkset. This downtime relates to the capability to send SS7 messages to the destination using levels 2 and 3 of the protocol stack. These are stringent rules and are specified in the Telcordia publication GR-246-CORE.

Physical Link Interfaces

The signaling data links are connected to network equipment using electrical interfaces. These interfaces are industry-standard interfaces defined by standards bodies such as ITU-TS and the *Electronic Industries Association* (EIA). The interface type will depend on the type of equipment used with the links. For example, if a *data service unit* (DSU) is used, a V.35 interface will be needed to connect the DSU to the signaling point.

Interfaces operate at level 1 of the SS7 protocol stack and provide the electrical/optical medium for transmission of data packets within the SS7 network. The following are descriptions of the most commonly used interfaces in the SS7 network.

V.35

This interface is commonly used from a data service unit to the SS7 signaling point. The V.35 interface can also be used from a *digital x-connect* (DSX) panel. V.35 provides data rates up to 56 or 64 kbps. Slower data rates are supported as well.

The V.35 interface was originally intended for use with high-speed modems. Interfacing analog modems to a digital line at 48 kbps was the first implementation of this interface. Later, the ITU-TS adopted this interface for use in all digital lines, with data rates of 48, 56, 64, and 72 kbps.

The ITU-TS Blue Book considers this interface obsolete and no longer recommends its use. Instead, the ITU-TS Blue Book recommends the use of V.36 or other similar standards. The V.35 interface is still very common, however, and is found in many equipment types (see Figure 2.18).

When using a V.35 interface, a clock source must be provided. This clock source is typically provided by the switch itself, but it can be provided from an external source. One side of the V.35 connection must be defined as *master* (clock source) and the other as *slave* (uses master clock). If the master side fails, a mechanism should be provided to enable the slave to become master and provide its own clocking.

DS0A (Digital Signal 0)

This is a 56/64 kbps channel located in a DS1 (or higher) facility. The link can be carried with an existing DS1/DS3 circuit between offices as long as one of the DS0As is

Pin 1 - Protective Ground
Pin 2 - Transmitted Data
Pin 3 - Received Data
Pin 4 - Request to Send
Pin 5 - Ready for Sending
Pin 6 - Data Set Ready
Pin 8 - Receive Line Signal Detect
Pin 15 - Tx Signal Element Timing
Pin 17 - Rx Signal Element Timing

Figure 2.18 The ITU V.35 interface may use a 37-pin or a 15-pin connector.

dedicated to SS7 signaling. A DSU/CSU is required to terminate the DS1/DS3 and separate the various DS0As from the circuit.

The DSU/CSU is usually located close to the entrance point of the digital facility into the telephone company building. From there, the various DS0As are crossconnected through a DSX to their final destinations.

This is the most commonly used interface in U.S. SS7 networks. Signaling points on the network usually have the capability to terminate the DS0A circuit without interface adapters. The maximum cable length for a DS0A is 1,500 feet. This limitation is not a transmission limitation, but a consideration due to propagation delay. The Telcordia requirements also specify the nominal impedance as 135 ohms with a balanced transmission path in each direction.

In the future, DS0A with clear channel capability may be used. At the time of this writing, this has not yet been written. However, work is being accomplished in the area of 64 kbps DS0s. The designation for a 64 kbps DS0 is DS0C.

Clear channel means that the data source can transmit a full 64 kbps without any restrictions on ones density or all zeros. Currently, the DS1 level, which carries the DS0 signal, enforces the ones-density rules.

Because DS0A is a digital facility, clocking is critical. For this reason, whenever a DS0A is used on the SS7 network, the DS0A must be synchronized according to the Digital Synchronization Network Plan (TA-NPL-000436, "Digital Synchronization Network Plan," Issue 1: November 1986). Network synchronization is explained in Chapter 1, "Signaling Systems #7."

Synchronization is accomplished at two different levels: bit synchronization and byte synchronization. Bit synchronization ensures that the transmitter and receiver are operating at the same data rate. Byte synchronization ensures that the receiver can properly define the alignment of frames. This is critical in defining the beginning and the end of a received frame.

The DS0A used within the central office uses bipolar encoding. The nominal pulse width of this signal is 15.6 microseconds and has rise and fall times of 0.5 ms. Clocking is provided by the *building integrated timing supply* (BITS).

The DS0A interface provides the reliable digital transfer of data. There are no configurable options to worry about in DS0A circuits, except for encoding schemes and data rates. The encoding method used will vary from network to network depending on the type of multiplexer used.

There has been some discussion about replacing DS0A with a full DS1 interface. This would provide 1.544 Mbps on one link. This is not necessary today, but as the traffic mix changes and becomes more complex, this may become a requirement.

The major obstacle in DS0A links is timing. When a DS3 is used between two exchanges, four multiplexers must be used (end to end) before the DS0A signal can get to the signaling point. Whenever this many multiplexers are present, timing synchronization can become a problem.

When the timing on a link is not synchronized between any two multiplexers, the links cannot carry data properly because the signaling point will not be able to read the data. Remember that timing is used to delineate between bits. If the synchronization between any two devices is not correct, it causes the receiving device to see bits that do not exist.

When using DS0A links, the most common problem encountered is related to losing the clock synchronization, which causes the links to be taken out of service. The fastest correction is to reset all the multiplexers and enable them to resynchronize with one another.

High-Speed Links

There are three interfaces defined to support *asynchronous transfer mode* (ATM) in the SS7 networks: *Synchronous Optical NETwork* (SONET), DS1, and DS3. This provides a migratory path for telephone companies looking to deploy ATM in their signaling networks. The transmission rates vary.

DS1 supports transmission rates of 1.544 Mbps, whereas DS3 supports transmission rates of 44.736 Mbps. SONET supports much higher transmission rates. For SS7 signaling links, transmission rates of 51.840 Mbps, 155.520 Mbps, 622.080 Mbps, and 2.48832 Gbps have been defined using the ATM *signaling ATM adaptation layer* (SAAL) protocol in place of the MTP levels 1 and 2. A subset of MTP level 3 is used to deliver *broadband ISUP* (BISUP) messages over these links. Refer to the Telcordia document GR-1417-CORE for more details on the ATM interface.

Telephone companies can deploy ATM on their SS7 networks in two ways. Currently, only the ATM interface has been implemented because SS7 does not require the amount of bandwidth provided by the other options.

TCP/IP Links

The industry has moved rapidly to deploying TCP/IP facilities for the use of telecommunications rather than traditional TDM-type circuits. TCP/IP facilities are a fraction of the cost of traditional TDM circuits, mainly because there is a significant difference in how these circuits are billed. Cost is only one factor.

Packet networks are far more efficient than channelized facilities because they provide more efficient use of the available bandwidth. In channelized facilities, a 56 kbps link is used for one purpose, whether or not the full bandwidth is being used. In packet networks, everything is sent over the same facilities, enabling the full bandwidth to be utilized.

Using TCP/IP for SS7 does present some problems. The TCP/IP protocols do not support real-time applications such as voice. SS7 also requires a higher QoS, which is mandating changes to the TCP/IP protocols. MTP levels 2 and 3 provide many services not currently supported in TCP/IP. The *Internet Engineering Task Force* (IETF) and the ITU are actively defining new protocols to replace MTP level 2 and 3. *MTP2 User Adaptation Layer* (M2UA) and *MTP3 User Adaptation Layer* (M3UA) replace MTP level 2 and 3, *SCCP User Adaptation* (SUA) replaces SCCP, and *Simple Control Transmission Protocol* (SCTP) is a peer protocol to TCP.

TCP/IP links use an Ethernet connection at level 1, supporting bandwidth of 10 or 100 Mbps. Eventually, all networks will be based on TCP/IP, eliminating legacy channelized equipment. This migration has already begun and is widespread throughout the industry both in North America and internationally.

Miscellaneous Interfaces

The V.35 and DS0A are the most commonly used interfaces for connecting links to network nodes today and TCP/IP is gaining popularity. Other interfaces are used for interconnecting adjunct equipment such as terminals and communications equipment. They are mentioned here so that you can understand their usage.

RS-232/V.24. The RS-232 interface is a serial interface designed originally for connecting modems to computer equipment. Over the years, the RS-232 has found many other uses as a serial interface. Printers, terminals, and other devices requiring a serial interface can use the RS-232. The RS-232 provides a separate transmit and receive path as well as flow control. However, there are limitations. The maximum cable distance for the RS-232 is 50 feet. In many cases, this is not a problem. However, in the central office, this may be unacceptable because terminal equipment and switches are often placed in different areas. In addition to the distance limitation, RS-232 has a maximum data rate of 19.2 kbps.

The signals provided by the RS-232 interface are divided into four categories: data signals, control signals, timing signals, and grounds. Data signals consist of transmit and receive paths for the user data.

The RS-232 provides eight control signals. Not all of these control signals are needed because some were developed for use with modems specifically. *Request to send* (RTS), *clear to send* (CTS), and *data set ready* (DSR) are typically used with printers and terminals for flow control. *Data terminal ready* (DTR), ring indicator, data carrier detector, data modulation detector, and speed selector are all optional control signals used specifically for modems. The mechanical requirements of the RS-232C call for a 25-pin connector (typically the DB-25 type). Most any connector fitting the application and the equipment can be used. The DB-25 is the most common connector used; however, many smaller connectors are being sought because equipment is getting smaller.

The electrical requirements of the RS-232C are specified as follows:

- Binary 1 = voltage more negative than −3 V
- Binary 0 = voltage more positive than +3 V
- Signal rate = <20 kbps
- Distance rate = <15 m

The functional requirements call for an unbalanced transmission path. One ground is provided as a return for both data leads. The other ground is a protective isolation ground. Synchronous transmission is accomplished by sending timing signals over the leads designated as receiver signal element timing.

An unbalanced line means that one lead is used to transmit the data, while the common ground is used as the return path. Interference can cause signals to be altered and, because the signal path is only over one lead, the voltage difference can be damaging to the data. In a balanced circuit, the data is sent over one lead with another lead used as the return for the same circuit. This means that current is carried in one direction (data flow) and returned on another lead, creating a complete circuit. Interference may occur, but it will not affect both leads. Thus, balanced circuits are better for data transmission over long distances. RS-232 is limited to short distances because of its use of unbalanced transmission circuits.

When a device is ready to send data, the RTS lead goes high. The receiving device acknowledges RTS and raises the CTS lead high. These are the minimal signaling requirements of the RS-232C. Many other signaling and control leads can be incorporated, but most manufacturers find that RTS and CTS are all that are needed. Modems require most of the leads indicated for reliable transmission.

In a modem configuration, connections are established in a different fashion. Modems are most often used by maintenance personnel wanting to connect to a remote SS7 signaling point from their computer workstation. They can then perform maintenance and administration tasks from their location. To understand the sequences that take place when a modem is concerned, let's look in more detail at the steps involved.

When the craftsperson is ready to connect to the modem, he or she will choose a communications software application on the workstation. The application will perform the steps necessary to connect to the modem. The interface will go through various stages before transmission actually begins. When the computer is ready to transmit, the DTR lead from the workstation will go high.

The modem has now been alerted that the workstation wants to place a call. The workstation will need to send the telephone number of the signaling point to the modem. This can be accomplished over the *transmitted data* (TD) pins of the interface. The modem then dials the number over the analog telephone line. Most modems will have a speaker incorporated into the modem so the user can actually hear the dial tone as the modem goes off-hook and begins dialing the number. This can be extremely helpful when trouble is encountered because you can hear whether the call actually went through.

When the line begins ringing, the distant modem should detect a ring generator on the line. When it detects the ring generator, the distant modem will raise the ring indicator lead to high to alert the signaling point that there is an incoming call.

The signaling point should then raise the DTR lead high to indicate that it is ready to receive data. This indicates to the modem that it should answer (go off-hook) and establish a connection with the distant modem.

The distant modem then answers the line and places a carrier signal to the calling modem. At the same time, the called modem raises the DSR lead high to indicate to the signaling point that it has answered the incoming call and it is ready to communicate.

The calling modem then sets DSR high to indicate to the workstation that a connection has been established and it is ready to transmit data. The calling modem will return a carrier signal to the called modem so that full-duplex transmission can be established. The called modem will set the *carrier detect* (CD) lead high to indicate receipt of a carrier signal.

The workstation then sets RTS high, indicating that it is ready to transmit data. The modem responds and sets CTS high. This is an acknowledgment to the workstation. The workstation then begins transmitting the data in serial fashion (1 bit at a time) over the TD lead, using the carrier signal to send the data. The carrier signal is modulated (using any means of modulation) to represent the bit stream in an audible tone.

The called modem receives the modulated data, demodulates it, and sends it to the signaling point over the *received data* (RD) lead. When the transmission is complete, either modem can drop the connection. The RTS lead is set low (off), which causes the called modem to set CTS low. The carrier is dropped and the connection is released.

This entire procedure can be monitored at any end of the circuit by using a breakout box or RS-232 monitoring device (a simple device with a series of *light-emitting diodes* [LEDs] for each circuit). Whenever a modem is connected to a signaling point or any other communications equipment on the network, it is strongly suggested that such a tool be kept in the toolkit for troubleshooting the connection. Most modem troubles can be detected and isolated through the use of such a tool.

V.24 is the ITU version that the RS-232 was designed after. The V.24 interface provides the same functions as the RS-232 interface plus additional signals for automatic calling. In addition to the RS-232 signals, the V.24 provides many more signals for timing, control, and data transmission.

RS-449. The RS-449 interface was intended to replace the RS-232, providing increased distance and higher data rates. However, because manufacturers have been using the RS-232 for so long and a large number of these interfaces are already in use today, the RS-449 has not shared widespread acceptance. In the PC market, there really is no incentive for the RS-449 because distance is not a problem. Devices using a serial interface are typically found right next to the computer. However, where distance and speed are issues, the RS-449 is a better interface.

RS-449 supports distances up to 200 feet with a data rate of 2 Mbps. In addition to the enhanced performance, the RS-449 interface provides 37 basic circuits and 10 additional circuits to support loopback testing and other maintenance functions.

The mechanical requirements specified for the RS-449 call for a 37-pin connector for the basic interface and a separate 9-pin connector if the secondary channel is used. The electrical requirements show a significant improvement over RS-232 interfaces, which are limited to an unbalanced line.

The RS-423-A standard specifies an unbalanced mode for the RS-449 interface, whereas the RS-422-A standard specifies the balanced mode. In a balanced mode, the electrical characteristics are as follows:

- 100 kbps at 1,200 m
- 10 Mbps at 12 m

In an unbalanced mode, the performance is not as good, but it is still an improvement over the RS-232 standard. In unbalanced mode, the performance rating is as follows:

- 3 kbps at 1,000 m
- 300 kbps at 10 m

Chapter 3

Overview of a Protocol

Before looking at the *Open Systems Interconnections* (OSI) model, let us examine the functions of a protocol. A *protocol* is a set of rules governing the way data is transmitted and received over data communications networks. Protocols must provide reliable, error-free transmission of user data as well as network management functions. Protocols packetize the user data into data envelopes. Some have fixed lengths and others have variable lengths depending on the protocol used.

Protocols are used whenever a serial bit stream is used. The protocol defines the order in which the bits will be sent and also appends information for use in routing and managing the network. This appended information is only used by the protocol and is transparent to the user.

Some protocols, such as *Signaling System* 7 (SS7), actually send predefined messages to the other nodes in the network. Messages can be used at any layer above layer 1 and are commonly found at layers 2 and 3. A typical example of a protocol message is the *initial address message* (IAM) sent by the SS7 protocol to establish a connection on a voice circuit between two end offices. Other messages exist for SS7 and will be discussed in greater detail in later chapters. Predefined messages are an excellent way to send network management functions and handle data error procedures.

Other functions of a protocol include the segmentation of blocks of data for easier transmission over the network and reassembly at the receiving node. When sending multiple blocks of associated data, procedures must be provided that enable the blocks to be identified in the order they were sent and reassembled. In large networks, these data blocks can be sent in order, but are received out of order.

There are three basic modes of operation for a protocol depending on the type of network. A circuit-switched network protocol establishes a connection on a specific circuit and then sends the data on that circuit. The circuit used depends on the destination of the data. A good example of a typical circuit-switched network is the *Public Switched Telephone Network* (PSTN), which uses various circuits for the transmission of voice from one exchange to another.

Once the transmission has been completed, the circuit is released and is ready to carry another transmission. The protocol must manage the connection and release it when transmission has been completed. It must also maintain the connection during the data transmission.

Another type of network is a *local area network* (LAN). LANs use different types of protocols, but the method of transmission is usually very similar. A LAN usually has a bus topology or a ring topology. In both topologies, the data is transmitted out on the LAN, with an address attached in a protocol header.

When a data terminal recognizes its address, it reads the data. Some mechanism must be used within the protocol to remove the data from the LAN once it has been read. This differs from one protocol to the next. These types of networks only permit one message at a time to be transmitted across the LAN.

Packet-switching networks provide multiple paths to the same destination. Each message has an originating and a destination address. The addresses are used to route the message through the network. Unlike LANs, a packet-switched network enables many messages to be transmitted simultaneously across the network.

The circuits used for this type of network are always connected, and transmission takes place continuously. The direction the message takes from one node to the next depends on the packet address. Each packet provides enough information regarding the data to enable the packet to reach its destination without establishing a connection between the two devices. The X.25 and SS7 networks are both packet-switched networks.

Several layers of addressing are used in any protocol stack. Typically, at least three layers of addressing can be found. Each device on the network must have its own unique physical address. The node address identifies the particular device within its own network. The layer 2 protocols use this address because they are responsible for routing one device to the next adjacent device.

The next address layer is that of the network itself. This address is used when sending messages between two networks. This address can usually be found in layer 3 of most protocols. The network address is used by those devices that interconnect two or more networks (such as a router).

Once a message reaches its final destination, the logical address within the destination node must be provided to identify which operation or application entity within the node should receive the data. An application entity is a function within a network node, such as file transfer or electronic mail. "Application" does not imply something like word processing (in the network sense).

In the SS7 network, application entities are objects such as IS-41, which enables *mobile switching centers* (MSCs) in the wireless network to exchange data from one to the other using the services of SS7 protocols.

As the information is handed from one layer to the next, the protocol appends control information. This control information is used to ensure that the data is received in the same order it was sent and enables the protocol to monitor the status of every connection and automatically correct problems that may occur.

Control information includes sequence numbering and flow control. This function is usually found at layers 2 through 4, but it can also be found at higher layers. In the SS7 protocols, levels 2 through 4 provide varying levels of control.

As mentioned earlier, segmentation and reassembly are also tasks of the protocol. This is necessary when large blocks of data must be transmitted across the network. Large blocks of data can be time consuming and, if an error occurs during transmission, they can cause congestion on the network while retransmitting.

For this reason, blocks are broken down into smaller chunks, which make it faster and easier to control and transmit through the network. When a retransmission becomes necessary, only a small portion of the original data must be retransmitted, saving valuable network resources.

Encapsulation is the process of appending the original data with additional control information and protocol headers. This information is stripped off the message by the receiving node at the same layer it was appended. This information is transparent to the user.

Connection control is one of the most important tasks of a protocol. Connections must usually be established not only between two devices, but also between two application entities. These logical connections must be maintained throughout the data transmission. The establishment of a logical connection ensures reliable data transfer. The use of positive and negative acknowledgments informs the adjacent node of transmission status.

Sequence numbering is also used in these types of services to ensure that data is received in the same order it was transmitted. This type of protocol service is referred to as *connection oriented*. Each node may have multiple logical connections established at one time.

When the data transmission is complete, the logical connection must be released to enable another application entity to establish a connection and transmit data. Protocol messages (such as connect requests and disconnects) are used to manage these logical connections.

Connectionless services are supported in many protocols. Connectionless services enable data to be transmitted without establishing a logical connection between two application entities. The data is simply transmitted with enough information to enable the receiver to know how to process the data.

Sequence numbering and retransmission are not used with connectionless services. This type of service is not reliable and is typically found in applications such as electronic mail.

The SS7 network provides support for both types of services, but uses mostly connectionless services for data transfer. However, despite its use of connectionless services, the protocol in SS7 provides mechanisms that enable the emulation of connection-oriented services.

Flow control is used in most protocols to control the flow of messages to a particular node. This function is particularly important in SS7 networks because it is used to prevent congestion in any one signaling point. With flow control, protocol messages can be used to alert adjacent nodes of the congestion situation and invoke rerouting functions.

Stopping the flow of messages to any one node is also necessary in some cases when a node becomes unavailable and is unable to process messages. The protocols in SS7 are able to perform this task without human intervention. Often, congestion or outages can occur and routing can be changed without anyone even knowing what occurred until after the events have taken place and the problem has been resolved.

Error detection and correction are ways for protocols to determine if the data they are carrying has been corrupted. The methods for error detection vary, but they almost always rely on some technique such as a *cyclic redundancy check* (CRC). This runs an equation on the bit stream before it is transmitted and places the sum into a check sum field.

When the data is received by the distant node, the same equation is run on the data again. The receiving node then checks the sum and compares it against the sum in the check sum field. If they match, no error occurred. If they do not match, then an error occurred and the packet or data is discarded.

Overview of the OSI Model

The OSI model was developed and published in 1982 by the *International Standards Organization* (ISO) for use in mainframe environments. This protocol provides the procedures and mechanisms necessary for mainframe computers to communicate with other devices, including terminals and modems. Because the OSI model was developed after SS7, there will obviously be some discrepancies between the two protocols. Yet the functions and processes outlined in the OSI model were already in practice when SS7 was developed (such as layering protocol functions).

The OSI model divides data transmission into three distinct functions. There is the application itself, which is not included in any of these three functions. The application, or process, may be something like file transfer or electronic mail. The process is the user of the protocol and the entity transmitting data over the network.

The process will depend on a service or function within the protocol that will allow it to pass its data to the network for transmission. Before this can happen, information must be appended and certain tasks must be performed first. These tasks are the responsibility of the process layers.

The process layers use protocols that are unique to the application that uses them. The application has specific requirements of the protocol, yet the protocols used at the lower layers need not be concerned with any of these functions so they remain independent and transparent to the process layers.

The process layers interface with the transport layers, which provide the mechanisms necessary to reliably transfer data over the network. The transport layers include error detection and correction, as well as other tasks such as sequencing the individual segments.

Process layers are not dependent upon any particular network protocol. In fact, a successful protocol should be able to use the services of any network protocol. This is the main objective behind the OSI model. The various layers should be independent of one another and be able to use any protocol over the network.

The network protocol provides the mechanisms for actually routing the data over the network and getting it to its destination node. The network protocol has no knowledge of the process addresses and does not work with any of the transport information. Its only concern is moving the packet from one node to another node within the network.

Routing is accomplished by reading the device address and the network address. This is the only information needed by the network protocol. Some sequencing may also be used, but this is not to be confused with the sequencing used by the process layers. The sequencing at this layer is simply used to ensure that all the packets that were transmitted were indeed received. The numbering does not necessarily imply any order.

The network protocols are usually divided into two parts: node-to-node transfer and network-to-network routing. Node-to-node transfer is only concerned with the transmission of a data packet between two physical entities. This takes place over a physical connection between the two entities, which ensures that the sequenced data is received in the proper order.

The network-to-network part is concerned with the routing of information between two networks. This layer typically uses the network addressing and is not concerned with the device address in many cases. In fact, with many LAN protocols, the network layer does not know the device address because it is located in a different layer of the protocol header.

This layered approach provides specific functions and is used for specific applications. By using a layered approach, changes to the protocol do not affect all the layers. This is important to network users. Network equipment works at specific layers rather than at all layers. If a change is made to the protocol, the equipment needs to be changed only if the change affects the layer at which the equipment operates.

As we look at the OSI model, you will begin to see how the tasks assigned at each layer can easily be independent from the other layers. Begin thinking of simple devices used in networks (such as routers and bridges) and what their functions are in the network, and then match these devices to the layer at which they operate.

The OSI model addresses all the functions previously mentioned and divides the functions into seven different layers. Each layer provides a service to the layer above and below it. For example, the physical layer provides a service to the data link layer. The data link layer provides a service to the network layer. Yet each layer is independent, and if the function changes at any one layer, it should not impact the other layers.

The OSI model defines the following seven layers, as shown in Figure 3.1:

- Physical (layer 1)
- Data link (layer 2)
- Network (layer 3)
- Transport (layer 4)
- Session (layer 5)
- Presentation (layer 6)
- Application (layer 7)

First we will look at the bottom layer: the physical layer.

Physical Layer

The physical layer is the layer responsible for converting the digital data into a bit stream for transmission over the network. The physical layer must provide the electrical characteristics needed to transmit over the interface being used. Conversion of the

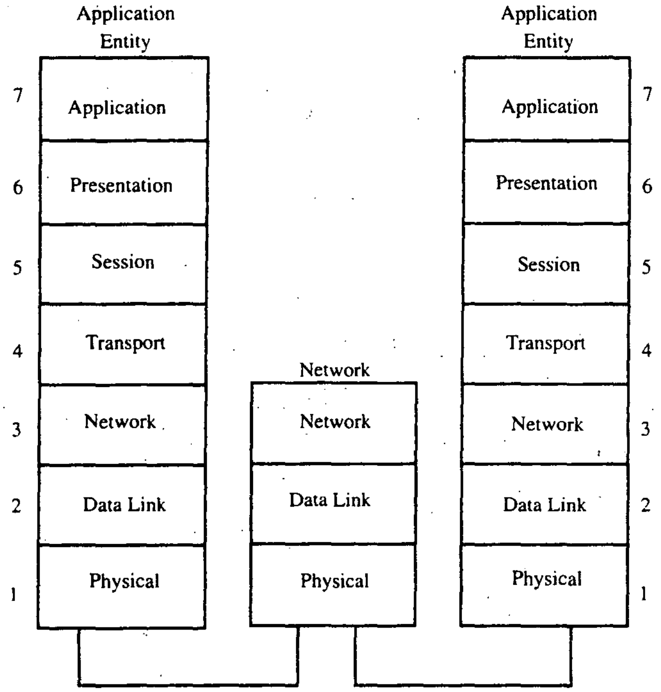


Figure 3.1 The OSI model defines seven layers of functions. The first three layers are primarily network functions that are used to transport information from source to destination.

digital signal from electrical to audible (as in the case of a modem) and even light (as in fiber optics) is the responsibility of the physical layer.

The responsibilities of the physical layer can be divided into several tasks. The most basic of these tasks is mechanical. The interface itself is a mechanical connection from the device to the physical medium that will be used to actually transmit the digital bit stream.

The mechanical specifications are dependent on the standard used. An RS-232 interface may use a DB-25 type of connection, whereas a V.35 may use an AMP connector. The mechanical specifications do not specify the electrical characteristics of the interface. The electrical characteristics are called out separately and are not dependent on the connector.

The electrical characteristics of an interface depend on the medium being used and the type of interface. Other factors, such as distance and the type of signals being transmitted, also play an important part when choosing a standard. The electrical properties of an interface include the signals used to actually transmit the bit stream and the control signals used to maintain the connection.

In the case of interfaces such as RS-232 and V.35, the data is transmitted on separate wires from the control signals. These control signals have nothing to do with the control information found in the upper layers of the protocol. In fact, they are also completely independent of the upper layers.

The control signals at the physical layer are hardware controlled and are used for flow control and maintaining a connection between the two devices. These signals may include *data terminal ready* (DTR) or *request to send* (RTS).

The upper layers have no knowledge of the control signals at the physical layer and do not attempt to influence their status at any time. The physical layer must be able to work on its own. This independence enables changes to be made to the physical layer (such as changing an interface type) without affecting the upper layers.

Although the control signals are independent of the upper layers, the physical layer has the responsibility to report any error conditions or line loss events to the data link layer. For example, if a clock signal is suddenly lost on the interface, the physical layer will report the loss of clock to the data link layer, which should invoke some sort of error recovery procedure (usually a reset of the hardware).

Electrical signals in their purest form consist of two states: on or off. To be more specific, an interface uses two voltage levels to represent binary digits. These levels remain constant until the binary digit changes to an opposite value. This is known as *nonreturn to zero* (NRZ). As long as a binary 1 is being represented, the voltage level remains high. When a change in the bit stream appears and a binary zero occurs, the voltage level changes to low.

Two terms are used to describe these transitions. The *bit rate* is the number of actual bits that can be transmitted over a line in one second. This is significantly different from the baud rate. The *baud rate* has nothing to do with the digital connection. The baud is a measurement of analog transitions that occur when using a device such as a modem. When using a modem, the digital signals are converted into analog audible tones that are transmitted at varying frequencies over the telephone line. Each time there is a frequency change, this is known as a baud. Understanding this fundamental difference between these two terms is important.

Many professionals mistake the two terms as being the same. Actually, it is quite possible to have a baud rate that is higher than a bit rate. When speaking of all-digital facilities, where modems are not used, the proper term is *bit rate*. The *baud rate* is measured after the digital interface on the analog side of the line.

Line encoding;line encoding is the process of altering the bit stream to force more transitions or fewer transitions, depending on the method of line encoding used. Line encoding becomes necessary for two reasons.

Digital signals cannot travel long distances and maintain their voltage levels. After some distance, the voltage level begins to drop. This can be critical when there are long series of consecutive 1s. To correct this problem, additional power can be applied to increase the wattage, which provides more push to the signal. This is also unfavorable because it requires larger power supplies, which give off more heat.

The optimum solution is to be able to use low power and maintain communications over long distances. Many types of line encoding are available to accomplish this. The simplest method is called *alternate mark inversion* (AMI). With AMI, every occurrence of a binary 1 causes the voltage level to change. For example, if the voltage level is at a positive level and there is another binary 1 in the bit stream, then the signal changes to a negative value. The use of negative voltages enables lower power requirements to be met.

Many other methods are used at the physical layer to overcome problems that occur when transmitting digital signals over long and short distances. None of these techniques are defined in the OSI model. The OSI model simply defines the processes that must take place, the various functions of the physical layer, the mechanical and electrical descriptions, and the services provided by the interface.

Data Link Layer

The OSI model defines the data link layer as the means to provide reliable communications between two devices. It is important to understand that the data link layer is only concerned with the data transmission between two devices, not the entire network. Communications through a network are handled at a higher layer.

The data link layer provides the services and functions necessary to transmit a bit stream between two devices using some method of sequencing and error detection and correction. Any management functions provided are only from the perspective of the physical interface used to interconnect the two devices. There is no knowledge of any other connections to the two devices from the perspective of the data link layer.

In addition to providing the services for reliable data transfer between two devices, the data link layer must also interface with the network layer above it and the physical layer below it. This is accomplished through the use of *primitives*. A primitive is a protocol between two layers. Because this interface is primarily software controlled, these primitives remain transparent to the end user.

The primitive to the network layer is used to pass received data from the physical layer to the network layer. Before this data is passed along, any information appended by the data link layer at the distant device must be removed. This includes sequence numbers and check sum fields.

When data is to be transmitted, it is passed to the data link layer from the network layer using this same interface. The data link layer must then append information to the original data. This information may include a device address, sequence number, and check bit sum. The address does not have to be that of the adjacent device. In fact, it would not make much sense if the address was always that of the adjacent device since that is the only destination known at this layer.

The destination address is usually the final destination for this data transmission. When received by another device, the device must search a routing table to determine how to route the data to the destination address provided at the data link layer.

Not all protocols use addressing at the data link layer. In SS7, the addressing is somewhat different than that of other protocols. This will be discussed with the network layer because the routing function is typically found at layer 3, the network layer.

Sequencing in SS7 is provided at layer 2 to ensure that data is received in the same order it was transmitted. If data transmission was always reliable and never errored, sequencing would not be necessary. However, this is never the case, and data transmission can get lost. When this occurs, the distant device has no indication that data was transmitted and lost.

Sequence numbering in SS7 provides a mechanism by which the distant device can tell if data was transmitted and then lost because the next data packet received will contain a sequence number that is not sequential with the previously received packet. Sequence numbers can be of any range, although they usually fall within two ranges. Modulo 8 provides sequencing in the range 0 to 7. Modulo 128 provides sequencing in the range 0 to 127.

However, even though the protocol enables 127 packets to be transmitted without acknowledgment, very few networks will permit this. A window size is configured in all network equipment to prevent the retransmission of too many packets. The idea is to send a burst of packets (15, for example) and if acknowledgment is not received after *n* seconds, retransmit the 15 packets. This is better than retransmitting 115 packets.

When acknowledging receipt of a message, the sequence number of the received packet is provided in the acknowledgment. Not every sequence number is acknowledged individually. One acknowledgment can be sent for a range of sequence numbers. For example, if an acknowledgment is sent with the sequence number of 6 and the last acknowledgment had a value of 3, the acknowledgment is for sequence numbers 4, 5, and 6.

Errors can be detected by using a check sum field. When an error is detected, the recovery procedure requests a retransmission from the originator of the errored packet. The errored packet is then discarded. Many methods are available for requesting a retransmission depending on the protocol used.

When a message is received, the data link layer must determine where the packet begins and what type of packet it is. Some sort of delimiter or flag precedes each packet. A *flag* is a specific bit pattern used before every packet. This bit pattern may never be duplicated in the packet itself because the data link layer will consider that octet as the beginning of a new packet.

For this reason, whenever a pattern is used for a flag, there must be some technique for ensuring that the bit pattern is never duplicated. The most common method is the use of *bit stuffing*. Bit stuffing inserts a bit in a fixed location (such as after every fifth consecutive binary 1).

The packet type will vary depending on the protocol. In some protocols, there are many different types of packets. A packet may be a supervisory packet, information packet, or unnumbered packet. SS7 uses three types of packets (called *signal units* in SS7).

The most important function of the data link layer is link management. The data link layer must be responsible for the integrity of the data link. When an error is discovered by the physical layer (such as a loss of timing), the data link layer is notified. The data link layer then invokes some method of error recovery to restore the link. In the case of a loss of timing, the link may be taken out of service and then reset. This enables the link to realign itself with the source clock.

Flow control is an important part of link management. In the data link layer, flow control can be performed through the use of protocol messages. In SS7 networks, special signal units are used with protocol messages, which indicate congestion conditions at an adjacent node or that an adjacent node is out of service and unable to process any messages.

Flow control initiates the rerouting of messages by the upper layers of the protocol stack so that messages will not be lost. This is not a function of the data link layer, but the data link layer must report congestion and out-of-service events to the network layer so that routing procedures can be invoked.

So far we have only discussed the procedures of a point-to-point configuration. Not all protocols and networks use point-to-point configurations. Many network topologies may also use multipoint configurations that have duplex or half-duplex transmission. The data link layer is impacted by the configuration, and its services and functions will differ depending on the configuration.

For the purposes of this book, we will only discuss point-to-point because all SS7 networks use point-to-point configurations between signaling points. A full-duplex link enables transmission in both directions simultaneously, as is the case in SS7 networks. This requires two separate paths per link. A half-duplex link uses only one path, but simultaneous bidirectional transmission is not possible.

SS7 networks use a simple data link layer protocol. Because of the point-to-point configuration and the nature of the transmissions, this layer does not require much complexity. In other networks, it may be necessary for the data link layer to inquire before sending a data packet. Without a positive acknowledgment, transmission cannot take place.

In SS7, data is transmitted continuously from a variety of sources. This transmission is always asynchronous in nature and does not require a session to be established with the receiving device. In fact, SS7 protocols are all connectionless-type protocols. Connection-oriented services are not used in today's SS7 networks.

Network Layer

The network layer provides routing services for data packets received from another node. In the case of a packet-switched network, packets may come into a node from a variety of locations. It is up to the network layer to examine the destination address and determine the link to be used to reach that destination.

The network layer is responsible for data transmission across networks. The transport layer provides a connection to an entity within a device, whereas the network layer provides a transparent transfer of the data for the transport layer. The network layer helps the transport layer free itself from the worries of internetwork data transfer.

There are two methods of reaching a destination. Some protocols require the establishment of a virtual connection with another node. Other protocols use *datagrams*, which are packets of information that contain all the control information necessary to advise the destination about how to process the received packet.

A virtual connection is established by sending a call request to another node. The purpose of the virtual circuit is to establish a consistent path through the network for all associated messages to follow. This method is used to overcome the inherent problem with packet switches, routing associated messages in multiple directions resulting in packets being received out of sequence.

When a virtual circuit has been established, the packets that follow use the same path through the network, ensuring that all messages are received in the same sequence they were sent.;This method is not favorable because it reduces the reliability factor in the network. If a node in the path becomes congested, messages are delayed. If a circuit fails, there are no alternate circuits and the message is lost.

Many packet-switching networks use datagram services to route packets throughout the network. This enhances the performance of the network as messages can be routed dynamically based on the status of the circuits and the nodes in the network. When congestion occurs at any one node, messages are quickly rerouted in another direction, avoiding the congested node.

This is much like the routing used in SS7. Although SS7 uses a datagram-type service, it also utilizes certain procedures for specific types of messages that emulate a virtual circuit. The difference is in the network management of SS7. Even though a message is routed over a virtual circuit, if a circuit fails in that path or a node becomes congested, it can be rerouted. SS7 enjoys the best of both worlds.

The addressing at this layer typically incorporates a multitier addressing scheme. The station or nodal address is found in the data link layer, while the network layer provides a higher level of addressing. Above the network layer is yet another layer of addressing: the logical connection, which is the final destination for all protocol messages. The logical address resides within a network entity.

SS7 addressing differs from this in that all addressing is located within the network layer. Addressing the node, the network, and even a group of signaling points within a regional area is accomplished with what is called a point code. The point code uniquely identifies all entities in the ANSI SS7 network.

Quality of Service (QoS) is a parameter that is used by the routing function to identify the quality of transmission that must be provided. For example, if a particular message requires sequencing and special handling, the network layer must identify the level of processing required to route the message throughout the network. This parameter is used by the network management function when congestion occurs or when messages get lost.

The SS7 protocol provides several mechanisms for QoS, including a priority parameter for prioritizing message types. The priority of a message determines when a message can be discarded and when it must be routed no matter what.

There are also network management functions at the network layer. When we discussed the data link layer, we discussed management procedures at the link level. Remember that the data link layer has no knowledge of the rest of the network. It is only concerned with the adjacent node to which it is connected.

Link management is the sole responsibility of the data link layer. As we discussed before, the status of the link is not broadcast throughout the network. This is of local significance only. However, if the status of the node itself should be impacted (perhaps by causing congestion), then the rest of the network must be notified.

This is the responsibility of the network layer. The network layer sends network management messages throughout the network, or, at least, to all of its adjacent nodes to inform them of degrading service at that node. This enables other nodes to make decisions about routing messages in different directions around the troubled node.

In many cases, the affected node sends a network management message to all of its adjacent nodes. They, in turn, must decide whether or not another network management message needs to be sent to all of their adjacent nodes, hence broadcasting out to the rest of the network. This usually depends on the type of network management message that is received.

As is the case with all levels, an interface to the layer above and below it is necessary. The OSI model talks about the use of *service data units* (SDUs). These are messages sent between layers of the protocol stack that contain the actual user data and information appended by the protocol (such as control information). This is passed in either direction, depending on the flow of the message. If a message has been received, it is always passed in the upward direction. If a message is being prepared for transmission, it is always passed in the downward direction.

In networks that use point-to-point architecture, there is little use for a network protocol. This is certainly the case in LANs. For this reason, protocols used in LANs do not use this layer unless other networks are bridged to the LAN. When other networks must be accessed by the LAN (internetworking), the network layer becomes a necessity. In the SS7 network, the network layer is also important because this network consists of many individual networks all bridged together.

The OSI model also talks about the difference between *data terminal equipment* (DTE) and *data communications equipment* (DCE). In OSI terms, the DTE is an entity that originates a data message and uses the services of the network to send this data to its destination—another DTE.

The DCE is the network device responsible for the actual handling and relaying of the message through the network. A DCE device can be a modem, router, packet switch, or any other intermediate node in the network for which the message is not the destination. The purpose of the DCE is to route the message to its destination, nothing else.

A DTE device is further defined to work at all seven layers of the OSI model, whereas a DCE device works at only the first three layers of the OSI model. These first three layers are the only layers necessary for actually transmitting data over the network.

In these simple terms, we can easily identify the *service switching point* (SSP) in the SS7 network as a DTE device. The *signaling transfer point* (STP) could be considered a DCE (although there are some functions of the STP that might also qualify it as a DTE). The *service control point* (SCP) could be considered a DTE.

The easiest way to remember this is to identify the endpoints of the network. The endpoints are where messages originate and terminate. The intermediate devices in the network work only at the first three layers and are considered DCEs.

In the world of networking, one of the most difficult achievements is the ability to interwork with other networks despite the differences in the protocols. This means that network layer and data link layer procedures must be converted. Conversion is not as simple as it may seem. Many times, one protocol may have procedures and functions that are not found in another.

When a message is received into a network from another, unlike the network layer, the interface between the two networks (the gateway) must provide direct one-to-one mapping of the message and all of its parameters to the equivalent in the other protocol. This can be difficult if such procedures and parameters do not exist and have no equivalents. The rule is to try and provide some sort of alternative when possible.

The conversion must always be transparent to the upper layers, which are not typically affected. Remember that the network layer operates independently of the upper layers, providing a service to the upper layers. When this service changes, the upper layers should not be affected.

In the SS7 network, interworking sometimes occurs at all levels of the protocol stack. Not only does the network layer require conversion, but the application layers must also be converted in order for the upper layers to be compatible between networks. This is done through the use of gateway STPs or protocol converters.

Understanding the network layer can help you understand the routing and network management that must take place within any network. Let us now take a look at the transport layer.

Transport Layer

The transport layer is used to ensure reliable communications over the network. This means that data must be received without error, in sequence, and without the loss of segments. The transport layer can be sophisticated or simple. However, if layer 3 is not capable of providing reliable transfer of data, then layer 4 must possess the capability to fulfill the role.

In essence, the transport layer relies on the reliability of the network layer so that it does not have to concern itself with this role. When a reliable network layer is provided, the transport layer is very simple. However, when the network layer is not reliable, the reliability factor must be built into the transport layer. Such is the case with protocols such as frame relay, which does not use any of the control parameters found in other protocols.

Addressing at this layer consists of the *service access point* (SAP). The SAP is a logical address within a node. The logical address is the interface from the network segments of the protocol to the upper layers.

Because the connection is taking place between two different devices, the transport layer must have some knowledge about the addresses in the other device. This is accomplished in a couple of different ways. The easiest method is to use predefined addresses for common entities. By using predefined addresses, all systems can address logical entities at the transport layer without having to query the distant device about addressing.

Another method is to broadcast the address any time a new function is added. This is commonly used in some LAN protocols today and enables functions not commonly used or too specialized to be predefined to notify other nodes of their function and address. The transport layer is the only layer that needs this information because it is responsible for the connection and termination.

The OSI model also talks about a naming convention, in which the particular task or logical function is called by name. This means that another device must provide the lookup capability of finding the physical address for the task name. This is commonly used in SS7 networks, where the signaling points may not know the actual address, but know the task with which they want to interface.

This is a very favorable method in large networks because it enables nodes to route to a function without having to know every address in the network. If another entity can provide the physical address, it saves memory space at each of the end nodes.

In X.25 networks, the transport layer also provides a multiplexing service. Virtual circuits may be used by many users, but only one transport service is used by all. The transport service must be able to multiplex its services among the many different users, even if they all come in on the same link. The transport service then splits the users to their various SAPs.

This function is not used in SS7 networks. In fact, the transport layer function is not even defined in SS7 today. As we will discuss a little later, the transport layer is not used in SS7 networks because SS7 does not currently support connection-oriented services.

Connection-oriented services, even with reliable network protocols, require the services of the transport layer to ensure the connection establishment and maintain the connection. Flow control is included in this layer to manage the data flow through the connection. The data flow is controlled to the layer below, the network layer.

It is clear that the OSI intended the transport layer to be used as a backup to the network layer, providing additional mechanisms for reliable data transfer. In today's networks, this is not an issue. Today's networks use reliable mediums and do not suffer from the maladies of networks five years ago. This is certainly evident to those who use modems for network access.

Not too many years ago, modem transmission was very unreliable at high speeds. Today, modem speeds of 14.4 kbps are possible because the telephone circuits have been improved. This is also the case with network mediums.

With protocols such as frame relay, where there are no control parameters, the transport layer becomes important. The philosophy in many of these networks is to let the upper layers worry about flow control and error detection/correction. This enables the lower layers to be simple and thus faster and cleaner. With dependable facilities, error detection does not become much of an issue.

Session Layer

The session layer is responsible for establishing a dialog, or session, with another entity. The session layer must also define the type of dialog to be established. This in itself implies a connection-oriented service.

The session layer also provides flow control procedures. Flow control at this layer is imposed on the interface to the transport layer. The peer entity at the remote destination does not interact with this flow control as it is of local significance only.

The session layer also manages what is called *synchronization points*. These are dialog units. An example of a dialog unit may be multiple file transfers, with each file representing one dialog unit. For example, if an entity needs to send several files to another remote entity, the session layer can establish each file as one synchronization unit. The entity can require that an acknowledgment be received for each synchronization unit before another can be sent. This is to ensure that each file is received properly before sending more data.

If a large data transfer is to take place and the transmission must be interrupted (for maintenance purposes or another task of a higher priority), the session layer must remember where the file transfer was interrupted so that it may start up at the place it left off. The session layer is not responsible for saving any data received; it is only responsible for marking the place of interruption and continuing on from that point.

The OSI model also specifies the use of a token at the session layer. The token is passed by the session layer to grant permission to transmit data. Several types of tokens are defined. One token grants permission to transmit data, another sets the synchronization points, and a third releases a connection.

Tokens are passed from one session layer user to another. Only the holders of a token may transmit data (if they are holding the data token). The holder of a token may also pass the token to the adjacent user.

As with the transport layer, the session layer is needed only when using connection-oriented protocols. If only connectionless services are provided, there is no reason to use this layer. In SS7 networks, the session layer is not necessary because SS7 does not support connection-oriented services.

Presentation Layer

While the application layer is concerned about the user's perspective, or view, of data, the presentation layer concerns itself with the view taken by the lower-layer protocols. Data encryption and compression are found at this layer.

Perhaps the best description of the presentation layer is to consider the function of compression. If data must be compressed before it is transmitted over the network, the presentation layer must perform the compression and provide a format (or syntax) that the session layer is going to be able to use.

The syntax of the data at the presentation layer does not necessarily match that of the layer above. The only requirement at this layer is to provide the data in a syntax that can be sent over the network and received at the distant node. The peer presentation layer at the distant node must be capable of decompressing the data for the upper layers.

Another function at this layer is encryption. Encryption involves scrambling the data in some format that can be descrambled at the distant end. The purpose of encryption is to provide security over the network.

The encryption technique used must be transparent to the session layer and to all layers below it. The presentation layer at the distant end is responsible for descrambling the data. In today's networks, encryption and compression are about the only applications really suited for this layer. In previous networks, where mainframes had to communicate with terminals, the presentation layer was used to present the data on the terminal.

Syntax is used by programmers who must write the procedures in software code for the various network devices. A standard notation for data is used in most programming languages. This layer uses an abstract syntax (such as *Abstract Syntax Notation One* [ASN-1]) to represent data types.

ASN-1 is the syntax used in SS7 applications. This syntax is commonly found in many network protocols and is widely used throughout the industry.

Application Layer

The application layer in the OSI model is the interface between the application entity and the OSI model. This interface is the first stage in processing the received data for transmission over the network.

The services listed in the OSI model relating to the application layer include information transfer, identification of the intended receiver, availability of the receiver, and any other functions not already defined in the lower layers.

Some examples of applications provided by the application layer include file transfer, job transfer, message exchange, and remote login. This layer also ensures that once an addressed entity commits to another entity, it cannot be interfered with by another entity. A database could be left in an unknown state if this were allowed.

Another principle to remember about this layer is that the application layer views data from the same perspective as the user. In other words, whereas the rest of the layers view the data from a network transmission perspective, this layer must view the data the way the user will see it. Thus, the data must be reconstructed as it was originally before it can be passed on to the application.

Overview of the SS7 Protocol Stack

This section will define the functions of the SS7 protocol in a conventional network based on *time-division multiplexing* (TDM). The SS7 protocol differs somewhat from the OSI model (see Figure 3.2). The OSI model consists of seven different layers, whereas the SS7 standard uses only four levels. The term *level* is used in the same context as *layers*.

The functions carried out by these four levels correspond with the OSI model's seven layers. Some of the functions called for in the OSI model have no purpose in the SS7 network and are therefore undefined.

It should also be noted that the functions in the SS7 protocol have been refined over the years and tailored for the specific requirements of the SS7 network. For this reason, many discrepancies exist between the two protocols and their corresponding functions.

Regardless of the differences, the SS7 protocol has proven to be a highly reliable packet-switching protocol, providing all of the services and functions required by the telephone service providers. This protocol continues to evolve as the network grows and

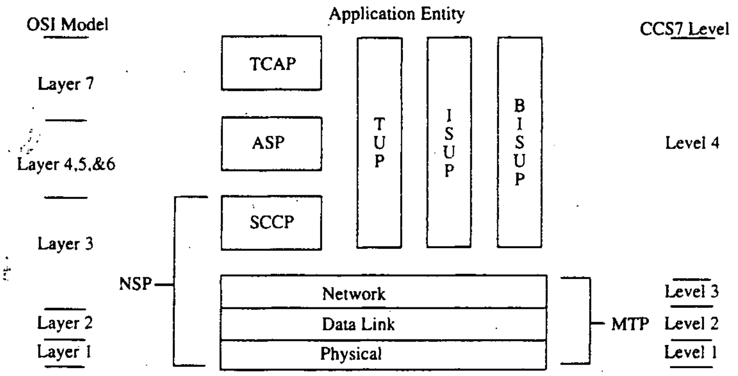


Figure 3.2 The SS7 protocol stack consists of only four levels and does not perfectly align with the OSI model. This is due in part to the fact that SS7 was developed before the OSI model. Many of the principles were in place, however, which explains the similarities.

the services provided by the telephone companies change. The following descriptions only apply to SS7 networks deployed using TDM circuit-switching networks. Networks using true packet-switching facilities (such as *Transmission Control Protocol/Internet Protocol* [TCP/IP]) do not use the same techniques.

Level 1—Physical Level

The physical level in SS7 is virtually the same as that of the OSI model. The OSI model does not specify any specific interface to be used, as this will always differ from network to network. In SS7, we can specify which interfaces will be used since the Telcordia standard and the ANSI standards all call for one of two types of interfaces—the DS0A or V.35.

The DS0A interface is the most favored for this application in North America, whereas the V.35 is widely used throughout the world. There is no inherent value for using DS0A in SS7 networks other than the fact that DS0A is already available. Because central offices are already using DS3 and DS1 facilities to link to one another, the DS0A interface is readily available in all central offices.

Many carriers have begun using *asynchronous transfer mode* (ATM) as a transport for SS7, but only when high concentrations of data links are needed due to heavy signaling traffic. Telcordia and Lucent have defined standards using an unchannelized DS1 at the physical level and ATM at the transport level. These are referred to as *high-speed links* (HSLs). As we have already discussed, TCP/IP has also received wide acceptance throughout the industry for use in SS7 networks. The SS7 standard does not specify the use of any one interface. In fact, the standards enable the protocol to use any interface at any rate. Performance requirements impact Telcordia requirements for switching entities and, in some cases, even the *International Telecommunications Union—Telecommunications Standardization Sector* (ITU-TS) performance standards will determine the type of interface to be used.

The theory, however, is that the protocol should be able to use any type of interface and any type of medium, maintaining true transparency throughout the layers. The other factors, of course, are the distance and transmission rates needed to support the traffic mixes in each unique network.

Level 2—Data Link Level

The data link level of the SS7 protocol stack provides the SS7 network with error detection/correction and sequenced delivery of all SS7 message packets. As with the OSI model, this level is only concerned with the transmission of data from one node to the next in the network. It does not concern itself with the final destination of the message. As the message travels from node to node, each node examines the dialed digits (contained in level 4) and uses that information to determine the next route for the message. Level 2 is given the information by level 3, which determines message routing. Level 2 then provides the functions necessary to transmit the packet to the next node.

Level 2 does not provide the routing for SS7. This is a level 3 function. Level 2 only provides the mechanisms needed to ensure reliable transfer of the data over the network. This is accomplished in several ways. First, level 2 provides the sequencing of messages between nodes. The sequence numbering is only of significance on one link. Each link uses its own sequencing series and is independent of the other links.

The sequencing numbering is used by this layer to determine if any messages have been lost during transmission. A lost message indicates an error, which is counted by an error counter maintained by level 3. After significant errors, the link is taken out of service and the network begins diagnostic and recovery procedures.

Another error-checking function maintained at level 2 is the *frame check sequence* (FCS). SS7 uses CRC-16 for error checking of the user data. The purpose of this mechanism is to maintain data integrity. The bit stream is subjected to the CRC-16 equation, and the remainder is placed into the FCS field. When the distant node receives the message, the same equation is used again, but this time the value is compared with the value in the FCS field of the received message.

If there is an error in a message or a message is lost, level 2 is responsible for requesting a retransmission. The retransmission may be accompanied with a message containing user data (user data in this context refers to level 4 information). Unlike most protocols, where retransmissions are unique messages that do not carry any bearer information, the SS7 method maintains this function at the lower level, enabling the upper layers to function independently.

This enables retransmission requests to be sent to the distant node while also sending a layer 4 message. This also allows a higher throughput of SS7 traffic rather than network management messages.

A length indicator is provided to enable level 3 to determine what type of packet (signal unit) it is receiving. Level 2 must know the type of signal unit being received so it knows how to process the message. If it determines that there is information intended for a higher layer, then this level will pass the contents of the message up to the network level, or level 3.

In packet-switched networks utilizing TCP/IP, the *MTP2 User Adaptation Layer* (M2UA) protocol is used to provide many of the preceding services.

Level 3—Network Level

The network level provides three functions: routing, message discrimination, and distribution. All three functions depend on the services of level 2. When a message is received, it is passed by level 2 to level 3 for message discrimination.

Message discrimination determines to whom the message is addressed. If the message contains the local address (of the receiving node), then the message is passed to message distribution. If the message is not addressed to the local node, then it is passed to the message-routing function. The message-routing function reads the called and calling party addresses in the message to determine which physical address to route to. The called and calling party addresses can be considered logical addresses and the physical address can be considered the node address.

The physical address in SS7 networks is referred to as a point code. Every node in the network must have a unique point code. The routing function determines which point code to route the message to based on information stored in its administrable routing tables. These routing tables are maintained by the service providers themselves and are network dependent.

The point code in many cases is not the final destination for a message, but the adjacent point code for this node. This enables messages to be routed through the network and rerouted to another node in the event of a network failure. The routing scheme is determined by the network providers and can vary depending on philosophy.

Message distribution is used when message discrimination determines that the address is a local address. Message distribution is responsible for identifying which user part the message is addressed to (based on the service information octet field of the message) and routes the message to its internal user.

There are three network management functions at level 3: link management, route management, and traffic. Each type of network management uses different mechanisms to achieve results.

The link management function uses the *link status signal unit* (LSSU) to notify adjacent nodes of link problems. A link problem does not necessarily mean that the link cannot transmit messages. Software errors or processor problems on link interface cards can cause a link to become unusable.

When this occurs, it is quite possible for a link to remain operational at level 2 and even level 3, but nonoperational at level 4. When this occurs, the adjacent node must be notified that the indicated link cannot be used for traffic because there is a problem at the affected signaling point.

Level 3 sends LSSUs via level 2 to the adjacent node, indicating the problems with the link and advising of its status. The link can be removed from service (which means that no MSUs are transmitted over the affected link) and diagnostics can begin. Diagnostics consist of realigning the link or resynchronizing the link.

Realignment occurs when traffic is removed, all counters are reset to zero, all timers are reset to zero, and *fill-in signal units* (FISUs) are transmitted for a prescribed duration of time, which is called the *proving period*. The duration of the proving period is dependent on the type of link being used. Telcordia has specified that the proving period for a DS0 at 56 kbps is 2.3 seconds for normal proving and 0.6 seconds for emergency proving periods. At 64 kbps, the normal proving period duration is defined at 2.0 seconds and the emergency proving period is at 0.5 seconds. When a 1.536 Mbps link is used, the normal proving period is defined at 30 seconds and the emergency proving period is defined at 5 seconds. During the proving period, any errors that may occur with the FISUs' transmission are counted.

When link management has determined that too many errors have occurred on the link, the entire process begins over again and timers and counters are reset to zero and FISUs are transmitted for a prescribed duration of time.

Another form of link management entails the use of changeover and changeback messages. These are sent using *message signal units* (MSUs) and advise the adjacent node to begin sending traffic over another link. The alternate link must be within the same linkset. During the time that all MSUs are being rerouted over different links, the affected link is being realigned by level 3.

A changeback message is sent to tell the adjacent node that traffic may be sent over the affected link once again because it has been restored to service. The changeback message is typically followed by a changeback acknowledgment message.

Route management provides the mechanisms for rerouting traffic around nodes that have failed or have become congested. This is a function of level 3 and works with the link management function.

Usually, when a link management message has been received, if the route of the node is affected, it may trigger the generation of a routing message depending on the impact on other nodes. Route management is used to inform other nodes in the network of the status of a particular node that has become unavailable or congested. This differs from link management, which only notifies an adjacent node about link status.

Route management messages use the MSU and are generated by nodes that are adjacent to affected nodes and not usually by the affected nodes themselves. These messages are the transfer-prohibited, transfer-restricted messages and are discussed in Chapter 6, "Message Transfer Part Level 3."

Traffic management is used as a flow control mechanism. Flow control is used in the event that a node has become congested, but only at a single level. For example, if a particular user part is not available (such as the *ISDN User Part* [ISUP]), a traffic management message can be directed at adjacent nodes informing them that ISUP at a particular node is not available without having any impact on *Transaction Capabilities Application Part* (TCAP) messages to the same node.

Traffic management, then, is different from the previous two functions in that it deals with a specific user part within an affected node rather than with the entire entity. This mechanism enables the network to control the flow of certain messages based on protocols without impeding other traffic that should not be affected.

In packet-switched networks using TCP/IP, *MTP3 User Adaptation Layer* (M3UA) provides many of the previous services.

Level 4—User Parts

Level 4 in the SS7 network consists of several different protocols, which are all called user parts and application parts. For basic telephone call connection and disconnection, the *Telephone User Part* (TUP) or ISUP protocols are used. TUP is used in Europe and other countries following ITU-TS standards, whereas ISUP is used primarily in North America (but is replacing TUP worldwide).

To access network databases, the TCAP protocol is used. TCAP supports the functions required to connect to an external database, perform a query of the database, and retrieve information. The information or data retrieved is then sent back in the form of a TCAP message to the signaling point that requested it.

TCAP also supports the remote control of other entities on the network. A network switch can invoke a feature or a function in another network switch by sending a TCAP message from one entity to another.

TCAP is being used more and more as the network evolves into a more intelligent network that is capable of many self-invoked functions. With the inclusion of wireless networks into the SS7 networks, the use of TCAP will increase for roaming and other wireless functions.

The *Operations, Maintenance, and Administration Part* (OMAP) is really an application entity that uses the services of the TCAP. The standard describes the syntax used for OMAP, relying on the ASN-1 standard. This is used to provide communications and control functions throughout the network via a remote operations center terminal. This terminal is typically located in a remote maintenance center, where control over all network elements is possible. The administration of system databases, maintenance access, and performance monitoring are all part of these centers.

The *Mobile Application Part* (MAP) is a relatively new level 4 protocol used in GSM wireless networks. The purpose of this protocol is to provide a mechanism that enables wireless subscriber information to be passed from one wireless network to another. The MAP parameters include information such as the *mobile identification number* (MIN) and the serial number of the radio unit itself. In North America, the IS-41 protocol is used for *Code Division Multiple Access / Time Division Multiple Access* (CDMA/TDMA) wireless networks.

Other level 4 functions exist and will be discussed in much greater detail in later chapters. For now, an understanding of the differences between the OSI model and the SS7 protocol stack is all that is necessary. While all the functions called for in the OSI model are addressed in the SS7 protocols, the SS7 protocol stack is condensed and does not address connection-oriented services used to establish a session with another user.

In addition to providing connection requests in the voice network, SS7 also provides for database access from any entity on the network. This is the most important feature of the SS7 network and the main reason why SS7 has been deployed in the PSTN all over the world: so that all telephone companies can share subscriber information and call-handling procedures on a call-by-call basis.

SS7 Protocols

Now that we have discussed the various layers, or levels, of the SS7 protocol, let us examine the protocols used within these levels to accomplish the specific functions required at each level. The protocols used within SS7 each have a specific application and are used according to the services they provide the network.

In TDM-based signaling networks, levels 1, 2, and 3 are combined into one part: the *Message Transfer Part* (MTP). MTP provides the rest of the levels with node-to-node transmission, providing basic error detection/correction schemes and message sequencing. In addition, MTP also provides routing, message discrimination, and distribution functions within a node.

In packet-switched networks using TCP/IP, M2UA and M3UA protocols replace the MTP protocols, providing the same services in the packet telephony environment. *SCCP User Adaptation* (SUA) has been defined to replace *Signaling Connection Control Part* (SCCP), while the *Simple Control Transmission Protocol* (SCTP) is used as a transport for these protocols to guarantee delivery of these protocols (a peer protocol to TCP and the *User Datagram Protocol* [UDP]).

When a database transaction is requested, MTP is accompanied by another higher-level protocol: the SCCP protocol. SCCP provides the addressing necessary to route a message to the correct database. Database addresses are called subsystem numbers and are the logical addresses used by the protocols to route to the appropriate database entity.

In the event that an originating node does not know the subsystem number, the dialed digits or other similar information is provided in a called address field. This information is then used for routing the message through the network. At some point, before the database is reached, the called party address must be translated into a point code and a subsystem number.

The point code is of the SCP connecting to the database, and the subsystem number is the logical address of the database itself. Once the SCP is reached, the subsystem number may be sent over another type of network, such as an X.25 network.

The SCCP message is then returned with the proper routing instructions to the end office requesting the global title. SCCP is also used as the level 3 protocol supporting TCAP, which is the protocol used for all database transactions. SCCP is required for routing TCAP messages to their proper database.

Another function of the SCCP protocol is to provide end-to-end routing, which is not possible with MTP. SCCP provides the means for routing a message transparently through the network using intermediate nodes as routers without the need to know the individual addresses of each of the intermediate nodes.

The addressing provided in the SCCP field enables each of the intermediate nodes to route based on the address in the SCCP protocol. The signaling points then base their routing on the SCCP address and generate the routing label for use by level 3 routing.

Although the standards often show a correlation between SCCP and the ISUP, there is no current definition supporting such services. SCCP at this time is only used in conjunction with TCAP protocol messages.

ISUP is the protocol used to set up and tear down telephone connections between end offices. This protocol was derived from the TUP, which is the ITU-TS equivalent to ISUP, but offers the added benefit of supporting *Intelligent Networking* (IN) functions and *integrated services digital network* (ISDN) services. ISUP is used throughout the United States today and provides not only call connection services within the PSTN, but also links the wireless network and the *Personal Communications Service* (PCS) network to the public telephone network.

Broadband ISUP (BISUP) is used for setting up and tearing down connections on ATM facilities. Because ATM and other broadband facilities identify virtual circuits rather than circuit numbers, a different protocol that matches these requirements is needed. BISUP identifies virtual circuits instead of TDM circuit codes.

Through the use of these protocols, SS7 is able to provide a variety of services that are not obtainable with the previous signaling methods. SS7 is a message-based packet-switching network that is capable of growing with the technology it must support. Because of SS7, the telephone companies have had to change their philosophies regarding service and are now finding themselves in a new industry—data communications.

Chapter

4

Overview of Signal Units

Overview of Signal Units

There are three basic methods of switching in a network: circuit switching, message switching, and packet switching. Circuit switching uses a physical connection between two entities for transmitting a data stream. The circuit remains connected until both entities have completed the transmission. A good example of a circuit-switching network is the *Public Switched Telephone Network* (PSTN).

Message switching came about in the 1970s and 1980s and uses a message structure to route data through a network. The data is accompanied by an address and a message (which serves as an instruction to the receiver). The data is sent in its entirety and does not include any error-checking schemes or flow control.

Packet switching arranges the data into a packet or a group of packets and transmits it in the form of a complete packet, providing all the information needed to route and process the received data. Network management and error detection/correction are included in packet-switching networks.

Signaling System 7 (SS7) is a packet-switching technology and uses data packets just like X.25 and other packet-switching technologies. A packet contains all the information necessary to route data through a network without establishing a connection to the destination.

Packet switching is a more efficient way of networking, and it makes better use of facilities. In the event of network failures or any other problems in the network, the packet protocol can dynamically change the routing for a particular destination and can provide higher reliability through error checking and correction.

Usually, packet-switching networks use different types and formats of packets depending on the purpose of the network. For example, if you send a data packet in an X.25 network, the packet type is called an *information frame*. The information frame has a distinct format and provides parameters that are specific to the transfer of data through the network.

If a packet is received in error and the packet must be retransmitted, a *supervisory frame* is used to inform the originator of the data packet that the data was in error and it needs to retransmit the errored packet.

The supervisory frame contains the parameters needed to inform another node of an error, but it does not support the transmission of any data. This packet serves a very unique purpose and cannot be used for anything else.

SS7 uses three different structures of packets, which are called *signal units*. These signal units provide three different levels of service in the SS7 network. The SS7 protocol uses all three signal units for the transmission of network management information, depending on the level of management. Information is sent using only one type of signal unit.

Another unique factor about SS7 networks is the source of information. In most networks, we are sending data from one user to another. In SS7 networks, the user is the telephone network. The information is control and signaling information from telephone company switches and computers, which must be shared from one device to another. This makes the SS7 network a machine-to-machine network rather than a user-to-user network. There is very little human intervention in this network because most of the procedures and processes are automated and do not require any operator control.

A signal unit is nothing more than a packet, but SS7 has many applications, requiring different packet structures and capabilities. The applications found in the SS7 network vary from standard networks. SS7 has circuit-related applications and noncircuit-related applications. These are the two basic foundations used to identify the functions within the network.

Circuit-related applications are directly related to the connection and disconnection of telephone circuits used to connect telephone subscribers. These circuits can be analog voice trunks or digital data circuits. They are located on a separate network outside of the signaling network and are used for the sole purpose of connecting telephone subscribers to other telephone subscribers.

The SS7 network does not have anything to do with the voice and data in these circuits other than identifying the type of data and voice transmission that will take place (for example, data rates and encoding methods used at the voice interfaces).

Noncircuit-related applications consist of all other traffic in the SS7 network. To support the circuit-related functions of the PSTN, telephone switches must be able to communicate with one another. Whether they are requesting information from a database stored in a central computer system or invoking a feature in a remote telephone switch, there needs to be a protocol for all other aspects of the telephone network that are not related to a specific circuit.

Network management information is another type of communication that must be supported within the network. This is the automated part of the network, which enables signaling points within the network to automatically recover from failures and signaling point outages. Network management is completely autonomous in SS7 networks.

All signal units in networks based on *time-division multiplexing* (TDM) rely on the services of the *Message Transfer Part* (MTP) for routing, network management and link management, and basic error detection and correction. *Transmission Control Protocol/Internet Protocol* (TCP/IP) networks use IP-based protocols. The *Internet Engineering Task Force* (IETF) has actually created new protocols to provide the functions of MTP in the TCP/IP network environment (such as the *MTP2 User Adaptation Layer* [M2UA], *MTP3 User Adaptation Layer* [M3UA], *SCCP User Adaptation* [SUA], and *Stream Control Transmission Protocol* [SCTP]).

Anytime information is transferred through the network from one signaling point to another, the *message signal unit* (MSU) is used. It is called the message signal unit because SS7, like many protocols, uses data messages to convey information to another entity in the network. Information is considered control information or network management information.

The MSU provides the fields of the MTP protocol, including two fields: the *service indicator octet* (SIO) and the *service information field* (SIF). The SIO is used by level 3 to identify the type of protocol used at level 4 (such as the *ISDN User Part* [ISUP] or *Transaction Capabilities Application Part* [TCAP]) and the type of standard. A standard can be national or international. If the protocol at level 4 is based on the *International Telecommunications Union—Telecommunications Standardization Sector* (ITU-TS) standard, then the SIO field indicates that this is an international standard. If the protocol was any other type of protocol (such as *American National Standards Institute* [ANSI]), the SIO field indicates that this is a national protocol.

This information is used by level 3 and the message discrimination function to determine the type of signal unit, the protocol, and how it should be decoded. The SIO field also has spare bits that can be used for priorities in the ANSI standard or for other functions in private networks.

The SIF is used to transfer control information and the routing label used by level 3. Consider this the payload of SS7. This field can contain up to 272 octets and is used by all upper-level protocols such as ISUP and TCAP.

Not all information in this field is considered level 4 information. For example, in the case of network management information, it is level 3. The same is true if the *Signaling Connection Control Part* (SCCP) is used to transport TCAP. The SCCP portion of the message is found in the SIF field of the MSU. SCCP is considered level 3 information.

Link status information is carried using another signal unit type called the *link status signal unit* (LSSU). The LSSU is actually used by level 3 at one node to transmit the status regarding the link on which it is being carried to its adjacent node. The LSSU is never used to carry link status messages through the network. It is only used to communicate this status between two adjacent signaling points.

When no traffic is being sent and the network is idle, the *fill-in signal unit* (FISU) is sent to provide constant error checking on the link. This enables the SS7 network to maintain its high reliability because even though no information is being sent, the signaling points can still perform error detection on the FISU to determine if the link is beginning to deteriorate.

In addition to the FISU transmission, the MTP protocol is constantly monitoring the status of the link. The MTP is used in all three signal units. These signal units are explained in more detail in the following section.

Fill-In Signal Unit (FISU)

The lowest level signal unit—that is, the one that provides the lowest level of service—is the FISU. The FISU acts as a flag in TDM-based SS7 networks. FISUs are sent when there is no payload to be delivered and the network is idle (see Figure 4.1).

This is different than in any other network, where flags are transmitted. A flag is usually a 1-byte pattern, consisting of a 0, six 1s, and a 0 (01111110). This 1-byte pattern is used to maintain clock synchronization in many asynchronous networks. There is no intelligence in this type of pattern, however, so it serves no other purpose.

In the event that a data link begins to degrade, there is no indication that the link can carry traffic any longer until a transmission is attempted. By this time, it is too late. The transmission will fail and the link will have to go through diagnostics.

In order to maintain a high level of reliability, the FISU is used in TDM-based SS7 networks. Besides being used for synchronization of the link, this signal unit is also used to acknowledge receipt of MSUs. The sequence numbers are used to acknowledge a previous signal unit.

The most significant field in the FISU is the *frame check sequence* (FCS) field. This field is used by level 3 to determine if there are any errors in the FISU. This is based on the bits in all the fields of the FISU. The FCS is found in all signal units and is used to transport the remainder of the CRC-16 equation performed by the transmitting signaling point.

The CRC-16 is the error-checking mechanism implemented by all transmitting signaling points when transmitting a signal unit. The purpose is to provide the remainder of the *cyclic redundancy check* (CRC) equation to the receiver of a signal unit. The receiver then uses the same CRC equation and compares it to the value received.

By using this field for error checking in the FISU, the MTP can constantly evaluate the status of any link, even during periods of idle traffic. In the event that a link has degraded to a point where it is causing too many errors, the link can be taken out of service by the MTP link management function before it is needed for actual traffic.

The FISU can also be used to acknowledge a previously received signal unit. This is done by sending an FISU with a *backward sequence number* (BSN) that is equal to the *forward sequence number* (FSN) of the signal unit being acknowledged. In other words, the BSN identifies the sequence number of the last good signal unit received.

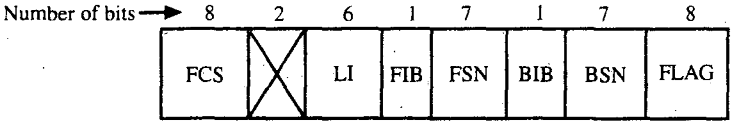


Figure 4.1 The FISU consists of the components necessary for routing (per MTP level 3) and is sent during idle periods instead of by flags. By sending the FISU, a signaling point can verify the integrity of a link by checking the FCS field for errors.

In the event that a signal unit is received and rejected by the MTP at level 2, the FISU can be used to send back a negative acknowledgment. A negative acknowledgment requires the use of a *backward indicator bit* (BIB). Usually, the BIB and the *forward indicator bit* (FIB) have the same value. However, when there is a negative acknowledgment, the BIB is toggled, assuming an opposite value from the FIB. This signifies a retransmission request.

The receiver of an FISU or any other signal unit with opposite values in the indicator bits examines the BSN to determine which signal units need to be retransmitted. This is more efficient than using the specialized supervisory frames that other protocols use.

When there are no errors, the indicator bits maintain the same value; that is, both the FIB and the BIB are exactly the same. When a retransmission occurs, the retransmitted signal unit is sent with the indicator bits set to the same value.

When the retransmission is acknowledged, the originator of the retransmission request toggles the FIB to match the BIB and maintains this value until another retransmission is required. This procedure is explained in full detail in Chapter 5, "Message Transfer Part (MTP)."

The FISU is never retransmitted if it is found in error. In fact, when FISUs are sent through the network, the sequence numbers do not increment (FSNs). There is no reason to retransmit these signal units because they do not provide any user information. They are only used to maintain the integrity of a signaling link.

The FSN assumes the value of the last MSU sent by the transmitting signaling point and stays at the same value until another MSU or LSSU is transmitted. The BSN follows the same procedure, unless it is used to acknowledge a previously received MSU.

The length indicator field in the FISU is always set to zero. The length indicator identifies the type of signal unit being received. The length is that of the information field, which does not exist in the FISU. The total length of an FISU is static at 48 bits long.

Although many drawings and publications show both an opening and closing flag, there is only one opening flag and no closing flag. The opening flag of one signal unit is the closing flag of the previous signal unit. This is defined in Telcordia GR-246-CORE.

In TCP/IP networks, the FISU is not used. A *keep-alive* signal is still needed, however, to make sure all entities on the network are still in service. The TCP/IP protocol does report when nodes fail; however, it may take several seconds before a node's status is reported. This is not acceptable for SS7 so the M2UA protocol provides a mechanism for constantly checking nodes on the network as the FISU does.

Link Status Signal Unit (LSSU)

The LSSU is sent between two signaling points to indicate the status of the signaling link on which it is carried. Therefore, the LSSU is only of significance between two signaling points and is not broadcast through the network (see Figure 4.2).

When a link is determined to have failed, the signaling point that detects the error condition is responsible for alerting its adjacent signaling point that the link is no

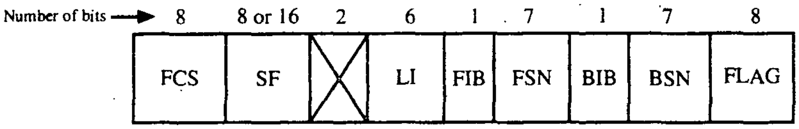


Figure 4.2 The LSSU is used by MTP level 3 to send link status information to an adjacent signaling point. The LSSU is not broadcast to any other signaling points.

longer available. The types of error conditions that warrant this procedure are alignment problems.

Alignment of a link means that all signal units received are of the correct length, and there are no ones-density violations. A ones-density violation occurs when the bit stream has more than five consecutive 1s, which is considered by the protocol as a flag. With link management and level 2 functionality, this should never occur. When it does, the link must be taken out of service and realigned.

Realignment is the procedure used by levels 2 and 3 to correct a link problem. The real problem is usually within a processor at either end of the link. Therefore, the processor that is at fault must be corrected. The first step is to remove all traffic from this link. The LSSU is sent to the adjacent signaling point to inform the adjacent node that all traffic should be removed from the link and the link should be realigned. No acknowledgment is required; this is simply an information signal unit.

This procedure is necessary for two reasons. To begin with, the two signaling points run independently of one another. Each end of a signaling link has its own processor. This processor and its accompanying software provide the functionality of levels 2 and 3. If the processor fails or is unable to process any more MSUs, only that processor would be aware of the trouble. The adjacent processor thinks that the link is working fine and is capable of carrying traffic. It is for this reason that the LSSU is necessary.

When level 2 determines that there is a problem (as notified by level 3), it will transmit an LSSU identifying the problem. The status indicators in the *status field* (SF) of the signal unit identify the specific status of the link; that is, the link is in alignment or there is a processor outage at the originating node.

The fact that the link is capable of sending this LSSU indicates the capability to send lower-level traffic, despite the incapability to process upper-level traffic. Depending on the status of the link, the receiving signaling point may send a network management message in the backward direction (to its adjacent signaling points), indicating the incapability to reach a particular signaling point. This would occur only if the route to a certain destination became inaccessible because of the link failures.

This means that the LSSU works in conjunction with other network management functions. The link management function uses the LSSU to notify adjacent signaling points of link status, while the route management function uses the MSU to notify adjacent signaling points of problems with a route to a destination.

The LSSU consists of the same components as the FISU, with the addition of the SF. The SF carries the link status information for the link on which it is carried. The LSSU is not transmitted on parallel links and does not carry information about other links. The SF indicates the status of the link on which the LSSU is carried.

Again, this implies that the link did not have a hard failure. A hard failure is one in which no traffic can be carried by the link, as in the case of a backhoe digging up a facility with SS7 links. The LSSU relies on levels 2 and 3 still being functional on the link.

As with the FISU, when an error occurs within the signal unit, the LSSU is not retransmitted. An errored LSSU is discarded, and the error is counted as an error on the link.

The value of the length indicator in an LSSU is either a 1 or 2. Currently, the LSSU SF is always one octet in length. Until further definitions are made for additional status indications, this rule will probably not change.

Message Signal Unit (MSU)

The MSU (see Figure 4.3) provides the structure for transmitting all other protocol types. This includes the ISUP, the TCAP, and the *Mobile Application Part* (MAP). The difference between the MSU and the previous two signal units is the addition of the SIO and SIF.

The SIO is used by level 3 message discrimination to determine the type of protocol being presented in the MSU. This enables message discrimination to identify the user at level 4. For example, if the SIO indicates that the protocol is ISUP, then ISUP will be the user at level 4.

The SIO also identifies the version of protocol being presented: international or national. The international protocol applies only to the ITU-TS-standard-compliant protocols. This is used when connecting to the international plane of the SS7 network. The national protocol applies to all other standards, including the ANSI standard used in the United States. It should be noted, however, that national does not imply ANSI. Many national standards are used throughout the world. For example, in Germany the national standard would be 1TR7, in Hong Kong it would be the Hong Kong standard, and in New Zealand it would be the New Zealand standard.

The use of two planes in the network enables all nations to internetwork on the international level using a gateway signaling point (usually a *signal transfer point* [STP]) to gain access into the international network from the national side, and vice versa.

Individual countries can then use their own individual flavcrs of the SS7 protocols, depending on the requirements of their own unique networks, without impacting the entire SS7 network worldwide.

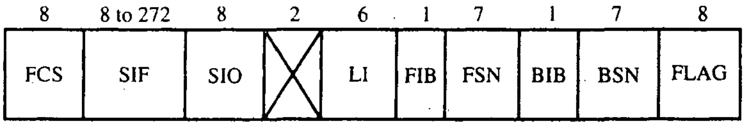


Figure 4.3 The MSU is used to deliver level 4 information to its destination. Level 4 information is found in the variable SIF.

All nations must comply with the ITU-TS standards at the international plane and use some method of interworking between the two planes. Interworking almost always requires protocol conversion. The SIO can be used to determine when this will be necessary.

Another reason this parameter is important is because of the difference between the point codes used between international and national. International point codes are formatted as a 3-bit zone identification, an 8-bit area or network identification, and a 3-bit signaling point identification. National point codes can be any variation as long as the total field length remains the same. The ANSI standard is an exception to this rule, where the point code is a 24-bit point code—8 bits for network identification, 8 bits for cluster identification, and 8 bits for member identification.

The MSU provides a signaling information field with a capacity of up to 272 octets of user data. In the case of SS7, user data consists of any data from an upper layer (such as ISUP or TCAP). The SIF does not necessarily have to be used for level 4 information. Network management also uses the signaling information field. Network management is a function of level 3.

The length indicator of the MSU can be any value over 2 and up to 64. The length indicator is a 6-bit field, which limits it to the range it can represent. Yet the SIF can be up to 272 octets long. In TCP/IP-based signaling networks, this limitation no longer applies.

When the SIF exceeds 64 octets in length, the length indicator of the MSU remains at 63. This is not an issue with the protocol because the only purpose of this field is to allow level 3 message discrimination to be able to determine the type of signal unit being received. There is no other use for this field.

Based on this fact, it is safe to say that any value in the length indicator over 2 is always an MSU, and any value over 2 is really insignificant. There is no reason to expand this field because the exact length is not important to level 3.

Primitives

In order for the various levels to interface with one another, some method of standard interfacing must be implemented. The use of primitives is not unique to SS7, although the particular primitive types used in this protocol are unique.

Communications between levels 2 and 3 and between levels 3 and 4 are all software controlled. We do not see any communications over the network, although we will see the results over the network. A primitive is the method used by software to pass information to the next level in either direction.

The important thing to understand is that a primitive is pure software. There is nothing for us to see or examine, unless we are looking at the source code of a signaling point itself. Primitives are discussed here for those who are actively writing software for SS7 network products and need to understand the full picture of what is taking place.

As seen in Figure 4.4, the primitive provides four fields. The first field, marked by the *X*, indicates the originator of the primitive. If the MTP was passing information up to the ISUP, the first field indicates *MTP*.

X	Generic Name	Specific Name	Parameter

X = MTP or N (SCCP)

Figure 4.4 Primitives are used to communicate with the various levels of the protocol stack within a network entity. Primitives are not seen in the network, but reside in software at each signaling point. This figure depicts the structure of a primitive.

The next field is the generic name. The generic name identifies the type of information being provided. For example, if information regarding the address of the originator (such as the calling party address) is being sent from ISUP to MTP, the generic name would be *unitdata*. The generic name will differ depending on the level. For example, the SCCP will have different generic names than, say, the ISUP. The functionality remains the same.

The field after the generic name is the specific name. The specific name describes the action that is to take place. The specific name can be any one of the following:

- Request
- Indication
- Response
- Confirmation

A *request* is used to invoke some type of service from another level. For example, in the case of network management, there may be the need to start a procedure. The request would be used to invoke that procedure at a higher level.

An *indication* is used to inform the requesting level that the requested service has been invoked. This is like an acknowledgment between levels. Using the preceding case, when a user part invokes a management procedure, it will send an indication to MTP to inform MTP of the invocation.

A *response* is sent to complete a particular transaction between a service element and a user. A user, in this sense, is the protocol, while the service element is something such as SCCP or the upper user parts. The response is used only when a service has been previously invoked and an indication has been sent.

The *confirmation* is sent to inform the user part that a connection has been established or a requested service has been invoked. Confirmation, in some aspects, is similar to an acknowledgment.

Many procedures surround the various primitives, depending on the level with which they are communicating and to which user part they are interfacing. The purpose of the primitive, once again, is to provide a means of communication between the various protocol levels within a signaling point.

Overview of SS7 Protocols

As we have discussed in the previous chapters, the SS7 network uses many different protocols. Each protocol is used for a specific purpose and provides the necessary functionality to accomplish specific tasks.

In this section, we will look at the various protocols used in SS7 networks and discuss their usage and applications. This section will only provide an overview of the various protocols. For a more specific explanation of these protocols, refer to their respective chapters.

The SS7 network provides some basic services to the PSTN. The impetus behind deploying this network was to remove all signaling information away from the voice network. In the early days of common channel signaling, this certainly seemed enough to justify the usage of another network. However, the SS7 network slowly evolved into much more than just a signaling network. It has also evolved into a control network.

The word "control" implies many different things. From a voice network standpoint, control refers directly to the capability to control features and tasks in a remote telephone switch or centralized computer. The user of this remote control capability is usually another telephone switch or computer system.

This network obviously forms the basis for an *Intelligent Network* (IN). In this discussion, IN is not referring to the IN or *Advanced Intelligent Network* (AIN) standards, but rather a network capable of supporting intelligent services through the use of call logic and/or service databases. Without the mechanisms for supporting remote control or other network entities, this would not be possible.

In today's convergent network, call control has evolved. Next-generation switches are computer platforms controlled by software. The *media gateway* (MG) provides access to the subscriber and transforms analog voice into digitized voice for transmission over the TCP/IP network (or the *asynchronous transfer mode* [ATM] network in some cases). Call control is distributed throughout the network to lower the cost of these network switches. The call control is now more centralized in *media gateway controllers* (MGCs), eliminating the need for expensive processing resources in the MGs. A control protocol such as Megaco or MGCP is used to communicate call control information between the MG and MGC.

However, the initial purpose of signaling cannot be ignored either, especially because this is currently the principle function of this network. As networks evolve and the many features and services being added require more intelligence, this will gradually change.

As we discussed in Chapter 1, "Signaling System #7," the SS7 network evolved from the earlier CCS6 network, which was more limited, yet of similar technology. The primary difference between the two technologies is in the protocols used and the technologies' structures. CCS6 used a very stringent structure that had fixed-length signal units. This did not allow for variable-length signal units and limited the protocol as far as the type of information that could be provided.

It was for this reason that SS7 was structured the way it is today. By providing a basic structure that various protocols can depend on for transport, it enables the upper-level protocols to be more dynamic. This means they can grow and evolve with the network without affecting the transport mechanism. This, of course, was the main limitation in CCS6 networks. The structure was too constrained and did not allow for sufficient growth of any kind in the upper layers, due largely to the absence of an independent transport function.

Message Transfer Part (MTP)

The MTP is the transport protocol used by all other SS7 protocols in the TDM SS7 network. This protocol is actually divided into three different levels. In comparison with the *Open Standards Interface* (OSI) model, MTP provides the same functionality as layers 1, 2, and 3.

The physical level of MTP (level 1) allows for the use of any digital-type interface supporting the data rate required by the networks. The bandwidth requirements are based on a number of variables related to the number and size of packets. Common interfaces in most SS7 networks today include DS0A and V.35.

For the most part, MTP works independently of all other levels. This enables the upper levels to evolve to meet the ever-changing demands of the network without affecting the interface.

There is one exception to this rule: broadband networks. Existing TDM links are limited not only in bandwidth, but also by the fact that a maximum of 32 links can be supported to any one network entity (this is less in some cases, as explained later). To overcome this limitation, high-speed links are used (ATM and IP based).

There are two standards (one by Telcordia and one by Lucent) for the use of DS1 facilities using ATM. Both are referred to as *high-speed link ATM* (HSL-ATM) with subtle differences. The DS1 bandwidth is 1.544 Mbps with no channelization.

Standards for IP are still evolving; however, a number of networks are already transporting SS7 over IP using Tekelec's TALI interface. The IETF Sigtran group is working on M2UA, M3UA, SUA, and SCTP to support the transport of SS7 over IP networks.

The level 2 function of MTP provides the functions necessary to provide basic error detection and correction for all signal units. This protocol is concerned only with the delivery of signal units between two exchanges or signaling points. There is no consideration outside of the signaling link.

This implies that level 2 has no knowledge of the final destination. This is a fair assumption. Level 2 does not need to concern itself with this information. In the true spirit of the OSI model, this is left up to the upper levels. Level 2 provides reliable transfer of information over a signaling link to the adjacent signaling point. Once the information reaches the adjacent signaling point, it is up to level 2 to determine how to route the message any farther.

Level 2 is maintained at the signaling link level. Each circuit card in an SS7 device must be able to provide and support this functionality independently of the rest of the system. For example, if several links are connecting to the same signaling point, each link runs independently and does not concern itself with the activities of the other links.

Sequence numbering is a function found at this level. Now that we understand that this level works independently of all other levels and all other links, we can assume that the sequence numbering is significant only on each particular link. In other words, if one link is transmitting messages using sequence numbers 1, 2, and 3, there is no synchronization of sequence numbers on the other links. They may use a completely different range of numbers as long as they are all sequential. Each link maintains its own sequence numbering.

This is also true for the adjacent signaling point on the same link. One signaling point can send sequence number 10, while the adjacent signaling point sends sequence number 121. This is because these links have independent processing that is not synchronized, enabling links to be much more efficient.

Another function of level 2 is error checking. Two methods of error checking are available: *basic* and *preventive cyclic redundancy* (PCR). Basic error detection/correction is used with all terrestrial signaling links. This is by far the most favorable because it is much more efficient than PCR.

PCR is only used with satellite signaling links, and it uses constant retransmission rather than error checking. With basic error detection/correction, when an error is detected, a retransmission is requested. The sequence number is provided for the last received signal unit that was good, enabling the originator of the bad signal unit to determine which signal units to retransmit.

In PCR, all transmitted signal units are retransmitted automatically during idle periods until they are acknowledged. Once they have been acknowledged, they are dropped from the transmission buffer. They are continually retransmitted until the distant end acknowledges them. This, of course, is not efficient use of the network and creates a lot of overhead.

The reason for this method lies in the propagation delay introduced when using satellite-signaling links. If a signal unit is sent, a retransmission may cross an acknowledgment because of the delay encountered. This would lead to some confusing situations in which a signal unit is retransmitted (due to a timeout, for example) and, at the same time, an acknowledgment is received. The receiving end would also find itself somewhat confused if it sent an acknowledgment, only to find the same signal unit being retransmitted.

Procedures to alleviate this are implemented whenever a satellite is used. The general rule is not to use a satellite for signaling links whenever possible, but when there are no other alternatives, the protocol supports the use of satellites and microwaves.

Level 2 also detects the presence of an opening flag for the delineation of an incoming signal unit. The flag is always a fixed pattern of 01111110 and is located in the first octet of the signal unit. As mentioned earlier, the opening flag is also the closing flag of the previous signal unit.

MTP level 3 provides four functions: message routing, message discrimination, message distribution, and network management. Network management is probably the most important. Network management maintains the integrity of individual signaling links by continuously monitoring them and counting the number of errors that occur on any single link.

When excessive errors have been counted, the link is removed from service (messages are blocked from the link) and the link is reinitialized. Because most errors are the result of clock signal degeneration and other related factors, resynchronization of the link usually resolves any problems that may occur.

When a link is said to be functioning properly and messages are of the correct length, the link is in alignment. When messages are received that are not the correct length or if there is a ones-density violation, the link is out of alignment.

Network management can rectify this problem. There are several functions within network management. Each function looks after a specific area of the network. They are

- Link management
- Traffic management
- Route management

Link management is concerned with the integrity of an individual link. Although this is a level 3 function, it relies on the service of level 2 to notify adjacent nodes when there is a problem on a link. The types of problems are typically errors, such as signal unit length and synchronization.

Link-management blocks messages from being transmitted over the affected link, routing traffic to other links within the same linkset. It also generates messages about the status to be sent to the node on the other end of the link. The adjacent node performs the same functions.

Only adjacent signaling points are notified about link troubles. Link management does not inform other signaling points on the network. Therefore, link management is a local function and does not directly affect the performance of the overall network.

There is a subtle impact, however, on the rest of the network when links fail. Link failures cause traffic to reroute to another link, possibly causing that link to become congested. If too much traffic is directed to other links and the processor cannot keep up with it, the signaling point can enter congestion.

When a signaling point becomes congested, the adjacent signaling points are notified to reroute all traffic around the congested signaling point. This can result in delays in the network and can even create congestion in other signaling points.

Link management is also responsible for activating and deactivating links and, in some cases, even automatic allocation. Automatic allocation is a feature offered in some *service switching points* (SSPs) that provide both voice circuits and SS7 links. Automatic allocation removes voice circuits from service and automatically places them in service as SS7 signaling links. The circuits must be preconfigured for this capability. Not all systems offer this capability, but when it is offered, it can be valuable in temporarily handling sudden bursts in link demand.

Traffic management provides the mechanisms for routing traffic around failed links within a linkset. Traffic management uses the MSU to send changeover and changeback messages to an adjacent node, informing the adjacent node of the failed links.

This is not to be confused with link management, which is responsible for turning links up and taking links out of service. Link management is what controls the status of a link and informs the adjacent signaling point of the link status. The difference lies in the mechanism used to inform the adjacent signaling point.

Link management uses the LSSU, which is carried on the link that is affected. Traffic management uses another link within the same linkset and is used when a link fails for any reason to advise the adjacent signaling point to use another link within the same linkset. This mechanism is necessary when a link is unable to carry any level of traffic, such as when a backhoe digs up a link facility.

Route management is used to inform other signaling points on the network about the incapability of one signaling point to reach another signaling point. For example, if a signaling point becomes inaccessible by an adjacent signaling point, the adjacent signaling point will send a route management message to its adjacent signaling points to tell them that it can no longer reach the specified point code.

Transfer-restricted and transfer-prohibited messages are two of the most commonly used route management messages. In the event that a link becomes unavailable and link management sends a link management message to an adjacent signaling point, it is possible that, in time, if congestion occurs, route management will be implemented to alert other signaling points on the network (only those adjacent to the originating signaling point) that the destination (or affected signaling point) can no longer be reached.

Besides the network management procedures just described, three other major functions are within level 3:

- Message discrimination
- Message distribution
- Message routing

First, message discrimination uses the routing label of the MSU to determine to whom a message is addressed. If the routing label contains the address of the local signaling point, then the message is handed off to message distribution. If the address is of another signaling point, the message is handed off to message routing.

Message distribution uses the SIO to determine the user of a message. If the SIO indicates that the user part is ISUP, the message is handed off to the ISUP. If the SIO indicates that the user part is the TCAP, the message is handed off to TCAP.

Message routing attaches a new routing label to an outgoing message and determines which signaling link should be used to route the call. The signaling point's routing table works with this function in determining the destination point code and the linkset that should be used to reach the destination.

M2UA

M2UA was developed by the IETF to be used to support the back haul of SS7 traffic over an IP facility. Currently in draft stage, this protocol will rely on SCTP or an equivalent transport.

One example of where this protocol would be used is between a *signaling gateway* (SG) and an MGC, where the SG is terminating conventional SS7 links (using MTP level 2). The SG would send messages to devices resident on an IP network. These are referred to as *application servers* (ASs), running multiple *application server processes* (ASPs). An example of an AS would be a server supporting services to telephone service subscribers.

The M2UA at the SG is responsible for maintaining the state of all ASPs that have been configured in the SG as destination addresses. They can be active ASPs (receiving traffic from the SG) or inactive ASPs (backup or redundant ASPs). In the event that there is more than one active ASP for a specific application, the SG is responsible for declaring one as active and the other as standby. Three redundancy modes for ASPs are supported: active/standby, load sharing, and broadcast.

Although the M2UA protocol does not specifically resolve the reliability and integrity of data sent/received, the document does specify the use of redundant servers whenever servers are used as well as the diversity of logical connections wherever possible. This is the same philosophy followed with traditional SS7 networks.

M2UA also supports client/server implementation, which means that the protocol stack itself can be run on servers configured as either client or server. The default configuration calls for the SG to be configured as server and ASPs to be configured as clients.

The mapping between interface identifiers and physical link interfaces is also the responsibility of M2UA. This is vital to ensuring that messages from the SS7 network are routed to the proper ASP residing on the proper AS. Each SS7 link can be assigned to one SCTP stream, allowing for a one-to-one correlation between links and SCTP streams. The standard also enables a link to be split across multiple SCTP streams (as would be necessary for broadband links).

One area not defined in M2UA is flow control and congestion. These are currently left to individual implementation, which could be an area of concern when performing interoperability testing.

M2PA

This is a new edition from the Sigtran committee that is designed for use with nodes connecting via IP without any TDM-based SS7 connections. In this case, messages are never sent to a conventional SS7 network.

A SG can connect to other nodes within its own network using nothing but IP connections. When this is the case, a point code is assigned to the SG, and it behaves the same as an STP in conventional SS7 networks without using TDM facilities. This protocol is used in place of MTP2 and even supports functions of MTP3, such as data retrieval for changeover procedures.

M3UA

This protocol provides the functions of MTP level 3 in TCP/IP networks used for SS7 transport. Address mapping from SS7 point codes to IP addresses is provided by this protocol. The actual mapping takes place at the SG. Routing is based on several parameters depending on whether or not the message is ISUP or TCAP.

For ISUP messages, routing is based on the *destination point code* (DPC), *origination point code* (OPC), SIO, and *circuit identification code* (CIC). These parameters are used to determine the IP address associated with MGCs on the IP network.

For SCCP/TCAP messages, routing is based on the DPC, OPC, SIO, and *subsystem number* (SSN). These parameters are used to determine the IP address associated with *service control points* (SCPs) on the IP network.

One significant difference between MTP3 and M3UA is the absence of protocol length limitations. MTP3 is limited to 256 bytes, whereas M3UA supports longer lengths. The *broadband ISUP* (BISUP) and BICC (ISUP1) protocols allow these longer lengths for broadband applications. Of course, when connecting to an IP-enabled STP within the network, messages over 272 octets will not be able to convert to MTP-based SS7; therefore, the larger message size is only used when interconnecting within IP. M3UA uses the services of the SCTP.

SCCP User Adaptation (SUA)

This protocol provides support for the transport of TCAP messages coming from or going to the SS7 network via an IP-based signaling network. The protocol provides full support for SCCP connectionless and connection-oriented services, dependent on the protocol being carried in the upper levels (such as MAP, INAP, and so on).

SUA was intended for use where telephone databases are connected to SGs using IP rather than conventional TDM-based signaling links. This is a favored approach in many networks due to the current bottleneck that exists when using TDM-based links. Only 32 links can be supported on any one database platform, creating a huge throughput issue for services such as *Local Number Portability* (LNP). By using IP, this bottleneck is eliminated and carriers can take advantage of 100 Mbps of bandwidth supported by IP.

If we look at the hierarchy of protocols in the previous scenario, IP would be the transport and SCTP would be the protocol providing services to SUA. The SUA protocol in turn provides services to TCAP or any other SCCP user.

The use of IP to connect to multiple servers providing a variety of network services is expanding and will continue to be one of the first implementations for IP in many networks.

Simple Control Transmission Protocol (SCTP)

Also developed by the IETF, this protocol is used as a transport in IP networks and is used in association with M2UA and M3UA protocols. SCTP is used in place of TCP or *User Datagram Protocol* (UDP) and has been developed specifically for use in SS7 networks where IP is the transport.

The difference between TCP and SCTP is in the delivery mechanism. SCTP uses packet-oriented delivery, whereas TCP uses stream-oriented delivery. SCTP also provides a means to protect against masquerade attacks, where messages are injected into the network from unknown points using false addresses (or addresses from other nodes on the network).

Network flooding is another concern in IP networks. Network flooding could occur when a network is attacked from unknown sources by sending thousands of messages (that is, network management messages to several destinations on a network). SCTP detects and prevents network flooding.

One of the unique features of MTP is the use of FISUs as keep-alive signals. As long as FISUs are detected on a link, the link is known to be capable of carrying live traffic. As soon as a link fails, the transmission of FISUs fails and network management takes over. This provides a means for detecting facility trouble even while the facility is idle. The FISUs also provide a means of acknowledgement on links when messages are transmitted.

SCTP replaces MTP; therefore, no FISUs are used in the IP portion of the network. However, SCTP does provide a keep-alive signal much like MTP to emulate FISUs.

In short, SCTP provides the functions needed to support a real-time protocol such as SS7 over IP networks. TCP was not designed for real-time functions and therefore does not make a good protocol for SS7 transport. This is what drove the development of SCTP. M2UA, M3UA, and M2PA all use SCTP.

Signaling Connection Control Part (SCCP)

The SCCP is only used with the TCAP, although the standards indicate its use with the SUP. The purpose of SCCP is to provide a means for end-to-end routing. The MTP is only capable of point-to-point routing. This means that a message can be routed based only on the physical links available from a signaling point.

SCCP provides the addressing to route a message through the entire network. This information is used at each signaling point by MTP level 3 routing to determine which linkset to use.

The difference between MTP and SCCP is the way the information is used and the nature of the addresses. The MTP provides both the OPC and DPC. In both cases, the point code is from a node-to-node perspective.

In the case of SCCP, the address consists of three parts: called/calling party, point code, and SSN. The routing can be based on any of the three, although when routing by point code, the address is a combination of the point code and SSN.

When routing a TCAP message, the signaling point must be able to identify the destination, which is almost always a computer database or a specific signaling point. In many cases, there may not be any dialed digits associated with the transaction (although in today's applications this is not the case). SCCP provides the addressing needed by MTP to route a TCAP message through the network.

The address information in SCCP remains fairly static unless the point code and SSN are unknown to the originator. In this case, an STP will have to provide translation. This is usually the case when a number is dialed that cannot be routed by the network.

The digits provided in the called party address are called *global title digits*. When the signaling point originating the message does not know the point code or the SSN of the database that will be providing a routing number for the requesting exchange, the global title digits have to be used by MTP level 3 for routing. At some point, the point code and SSN have to be provided so the message can reach its final destination. This function is known as *global title translation* (GTT) and is usually provided by the STP adjacent to the destination database.

When a number is dialed, such as an 800 or 900 number, the network cannot route the call based on conventional routing methods. This is because the numbering plan uses the area code of a number to determine to which area in the nation's network the call should be routed (the area being handled by a specific toll office). Likewise, the prefix usually denotes a specific central office that can route this call to the subscriber. In the case of 800 and 900 numbers, these do not have area codes that denote a toll office.

The SS7 network will provide a routing number by which the end exchange can route the call. This requires the services of the TCAP and SCCP. The called party address of SCCP will provide the dialed digits, although not all the digits are necessary. Only the area code (800) and the prefix are necessary.

The number is compared in a database, which provides the routing number to TCAP. The routing number is then returned to the requesting exchange via TCAP and SCCP so that a connection can be established for the call.

This is just one example of how SCCP can be used. The called party address does not have to be dialed digits. In the wireless network, the *mobile identification number* (MIN) is placed in the called party address for roaming information.

When used by the ISUP, SCCP enables ISUP messages associated with an already established connection to be routed using end-to-end routing, like TCAP messages. This functionality has not yet been implemented in SS7 networks; however, with new services and the evolution of the IN, this may become necessary.

ISDN User Part (ISUP)

The ISUP is a circuit-related protocol that is used for establishing circuit connections and maintaining the connections throughout a call. ISUP is only associated with voice and data calls on low-speed facilities. Broadband facilities such as ATM and frame relay are supported by BISUP.

ISUP supports both analog and digital voice circuits and was adopted by ANSI to replace the *Telephone User Part* (TUP). The TUP does not support data transmission or digital circuits. ISUP added the parameters necessary to support digital circuits and data transmission (specifically, *integrated services digital network* [ISDN] facilities, although standard *Plain Old Telephone Service* [POTS] facilities and virtually all other nonbroadband facilities are supported by ISUP).

ISUP is compatible with the ISDN protocol, which was developed as an extension of SS7 to the subscriber. There is direct mapping of ISDN Q.931 message types to ISUP message types, even though the message types are not the same. The purpose of the ISDN compatibility is to enable subscriber switches to send signaling information to remote subscriber switches during the call connection phase. After the connection has been established, ISUP supports communications between the two endpoint subscriber switches. This feature may be necessary to support caller-invoked features, such as conference calling or automatic callback. The capability to invoke features and share information between two subscriber switches and/or networks is the unique capability of ISUP and the purpose for its development.

Broadband ISDN User Part (BISUP)

To support the *broadband ISDN* (BISDN) and ATM architectures, the ISUP protocol was modified. This new version of ISUP provides additional message types and parameters, which provide the support necessary for ATM and broadband networks. For example, addressing in these network protocols is very different than in standard telephony, requiring changes to the protocols that must establish connections on these facilities.

The most significant difference between the ISUP and BISUP protocols is in the circuit assignment procedures and the type of circuits supported. ATM and BISDN circuits are virtual circuits rather than physical circuits. This places new demands on the SS7 network because it must be capable of assigning these virtual circuits and maintaining them. Because of the number of circuits available in broadband networks, a new circuit-numbering convention was adopted.

In addition to the new circuit requirements, broadband networks also support the dynamic allocation of bandwidth on a per-call basis. Now, when a call connection is established, the available bandwidth for that call is negotiated between the originating exchange and the destination exchange.

A few other, more subtle changes appear in this newer version of the ISUP protocol. They are described in more detail in the Chapter 9, "Overview of ISUP." In addition to the procedure descriptions, this chapter also has a section that explains the various message types and their parameters.

Telephone User Part (TUP)

The TUP is used in international networks. This protocol is compatible with the ISUP, with differences between the two mainly in the message type and parameters. Regardless of these differences, the two protocols can be mapped to one another successfully, even if a one-to-one mapping relationship does not exist. TUP is being replaced by ISUP at the international level as well.

The United States and ANSI decided early on to replace this protocol with the ISUP protocol in order to support the evolving services provided in many U.S. networks. The international market is slowly evolving to this protocol.

Before ISUP, support for data services and digital facilities was provided through a now-obsolete protocol called the *Data User Part* (DUP). DUP is no longer used in U.S. networks and has been omitted from this book. Because of the migration toward ISUP at the international level, TUP has been omitted from this book as well.

Probably the most noticeable difference between the structures of ISUP and TUP is in the header field. The ISUP protocol uses message types, whereas the TUP protocol uses an H0/H1 header. The H0/H1 header was originated from the CCS6 protocols. Messages are grouped into classes, which are represented by the H0 field. The H1 field denotes the specific message within that class. ISUP uses message types without any classification. This provides more flexibility in the protocol and more room for growth.

Transaction Capabilities Application Part (TCAP)

The TCAP is probably the most versatile of all the SS7 protocols. TCAP is used for two purposes: accessing remote databases and invoking features in remote network entities.

A network entity does not have to be a switch. Any network device, provided it is equipped with the proper interfaces and can provide all four levels of SS7 support, can be accessed by this protocol.

In today's networks, TCAP is limited to database access, although more networks are providing new advanced services that require the use of TCAP to invoke those services. Custom calling features provided by the IN will most certainly require the support of TCAP to invoke features and services in remote switches.

The TCAP protocol is not being used to its fullest potential today. The mechanisms currently provided in this protocol reach far beyond database access. As the IN evolves, the traffic mix in all SS7 networks will rapidly change to consist of mostly TCAP traffic.

The TCAP protocol has been designed to provide for the remote control of other network entities, which holds many possibilities in itself. For example, say a subscriber wants to change his or her telephone service. Normally, this would require a telephone call to the telephone company, which could remotely access the subscriber database and add the new service to the customer record. A service order would be generated and the new services programmed into the switch serving the subscriber. With TCAP capability, subscribers could enter the order entry system themselves. With an interface between the order entry system and the switch, subscribers could then change the program within the switch serving their telephone numbers.

Another use for TCAP is in wide use today. Wireless carriers have found that TCAP and SS7 networks are a good use for the transport of simple text messages used with pagers and many cell phones. *Short Message Service* (SMS) sends short text messages through the SS7 network using the services of TCAP as an alternative to building out more expensive next-generation wireless networks. Known throughout the industry as 2.5G and 3G networks, these wireless networks are costly to deploy, but support the use of wireless devices to access Internet services.

Carriers are finding that subscribers are perfectly happy in the short term with SMS-based services and don't necessarily need full Internet access through their cellular phones today. For this reason, we will continue to see SS7 networks expand their capacity to support the growth of SMS services.

Chapter

5

Message Transfer Part (MTP)

The MTP acts as the carrier for all *Signaling System 7* (SS7) messages in networks based on *time-division multiplexing* (TDM), providing the reliable transfer of messages from one signaling point to another. This function includes levels 1, 2, and 3. In addition to providing signaling-point-to-signaling-point communications, the MTP also provides error detection and correction.

The methodologies of MTP are very similar to those used in other *bit-oriented protocols* (BOPs), such as X.25. Sequence-numbering and error-checking mechanisms are very similar.

The signal unit structure used in all SS7 messages provides all the information required by MTP levels 2 and 3. Flow control is provided through the use of a special signal unit called the *link status signal unit* (LSSU), which is described in this chapter.

MTP is defined in the *International Telecommunications Union—Telecommunications Standardization Sector* (ITU-TS) documents Q.701 through Q.704, Q.706, and Q.707. They can be found in the Telcordia document GR-246-CORE, Volume One, Chapter 1.111.1 through 1.111.8. *American National Standards Institute* (ANSI) publications referring to the MTP protocol are numbered T1.111-1992, "Functional Description of the Signaling Message Transfer Part (MTP)."

The Telcordia recommendations add reliability and versatility to the network. The Telcordia publications are almost identical to the ANSI and ITU-TS publications, other than the additions made by Telcordia for network reliability and availability.

Overview of MTP Level 2

The MTP provides all functions of layers 1, 2, and 3 in the *Open Systems Interconnection* (OSI) model. We have already discussed the types of interfaces used at level 1 of the SS7 network. These are industry-standard interfaces and do not necessarily require an in-depth discussion here. Level 2 of MTP provides error detection/correction as well as error checking through the check bit field.

Besides error detection/correction, the MTP also provides link alignment, which could be associated with error correction. When a link is deemed unable to carry data (which is caused by clock corruption, among other things), the link can be removed from service and resynchronized. The following are the functions of the MTP:

- Signal unit delimitation
- Signal unit alignment
- Signal unit error detection
- Signal unit error correction
- Signaling link initial alignment
- Signaling link error monitoring
- Flow control

The purpose of the MTP is to ensure that the transmission facility is always functioning and capable of transmitting data. The MTP possesses the unusual capability to detect errors even when no data is being transmitted. This is done through the transmission of *fill-in signal units* (FISUs). The following is a description of the various procedures used to monitor the integrity of the data link and how the link is realigned in the event the protocol deems it necessary.

Performance

The performance of the entire MTP can be expressed in simple terms. The availability of a set of signaling links used to reach a specific route, which is defined to reach a specific destination, must meet the criteria outlined in the various standards.

Telcordia has defined three possibilities for performance. The network can be divided into three segments: the user portion, the network backbone, and the network access. The user interface is where switches and databases (*service switching points* [SSPs] and *service control points* [SCPs]) transmit and receive *ISDN User Part* (ISUP) and *Transaction Capabilities Application Part* (TCAP) messages. The network access portion involves the *access links* (A-links) from the SSPs and SCPs to the network *signal transfer points* (STPs). The links connecting STPs on the network are considered the network backbone.

The user interface unavailability should not exceed three minutes a year per interface. Network access unavailability is not to exceed two minutes a year. The backbone portion of the network unavailability should be a negligible amount of time.

Signaling Link Error Monitoring

First there must be a means for determining when a link is no longer capable of reliably carrying traffic. Apart from detection and correction, there must also be a means for monitoring the number of errors that occur on a link and making the decision to take the link *out of service* (OOS).

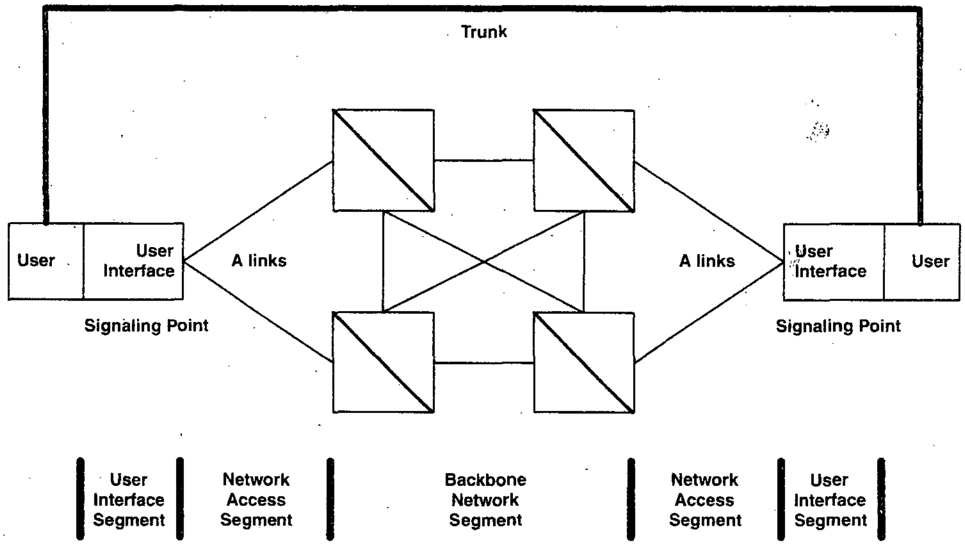


Figure 5.1

Three types of error rate monitors are used. Two are used when the link is in service and the other is used when the link is going through an alignment procedure. The *signal unit error rate monitor* (SUERM) and the errored interval monitor are used while the link is in service, and this is often referred to as the "leaky bucket" technique because of the way they decrement the counter after *n* number of good signal units or intervals.

The SUERM is an incremental counter that is incremented by one whenever an error is encountered. An error includes signal units received out of sequence, with a bad *cyclic redundancy check* (CRC) or one of improper length. After 256 signal units have been received without error (consecutive signaling units), the SUERM is decremented by 1.

When the SUERM reaches a value of 64 errors, a link failure is reported to level 3 and the link is taken OOS and placed through an alignment procedure. Level 3 controls the link management function and directs level 2 during the alignment procedure. Level 2 does not initiate the alignment procedure; it simply reports the errors and takes direction from level 3 link management.

When 1.536 Mbps links are used, the errored interval monitor method is used. The link is monitored for a determined period of time (defined by Telcordia as 100 milliseconds). If a flag is lost during the interval or a signal unit is received in error, then the interval is considered in error. An errored interval increments a counter in the same fashion as the SUERM. The counter is decremented when 9,308 intervals have passed without error (also defined by Telcordia). When the counter reaches a value of 144,292 intervals, the link is removed from service and alignment procedures are started. The actual values may be different, especially in international networks, but these are the recommended values defined by Telcordia.

The *alignment error rate monitor* (AERM) is used during the alignment procedure and is an incremental counter. Each time an error is encountered during alignment, the counter is incremented by 1. When the AERM determines that there have been excessive errors, it causes the link to be taken OOS and the alignment procedure begins again.

Signal Unit Error Detection

Errors are detected using the check bit field and the sequence number of the signal unit. If the check bit field is in error, the signal unit is discarded and a negative acknowledgment is sent to the originating signaling point. An error is also counted by the SUERM.

The level 2 timer T7, which is used for *excessive delay of acknowledgment*, prevents a signaling point from waiting too long for a positive or negative acknowledgment. Usually, an acknowledgment is sent when a signaling point becomes idle and does not have any more traffic to transmit. When congestion occurs at a signaling point or an extreme amount of traffic is present, it is possible that T7 could time out and force the retransmission of messages.

The recommended value for T7 is 11.5 seconds. This, of course, depends on the network. Although the actual value of T7 is optional, the timer is usually a nonadministrable timer, which means that once it is set, it cannot be changed by system administration.

Signal Unit Error Correction

When an error is detected in a signal unit, the signal unit is discarded. MTP level 2 counts the error (SUERM or errored interval monitor) and requests a retransmission if basic error control is being used. *Preventative cyclic retransmission* (PCR) treats errors differently and is explained in greater detail later in the section "Preventative Cyclic Retransmission (PCR)."

When excessive errors are detected on any one link, the link is taken OOS. The link is then placed through an alignment procedure to test the link and place it back into service automatically. The link will not be placed back into service until it has passed the *proving period* of the alignment procedure.

Signal Unit Alignment

A link is considered in alignment when signal units are received in sequence without ones-density violations and with the proper number of octets (based on the message type). The signal unit must be a total length of 8-bit multiples. If the signal unit is not in 8-bit multiples or if the *signaling information field* (SIF) of a *message signal unit* (MSU) exceeds the 272-octet capacity, the signal unit received is considered in error (the 272-octet rule only applies in conventional TDM-based SS7 networks and is eliminated when the *Internet Protocol* [IP] is used).

The link is not taken OOS until there has been an excessive number of errors. This is determined by the SUERM. This counter is used to count the total number of errors on a signaling link. Each link keeps its own unique counter.

The purpose of the counter is to determine when an excessive number of errors have occurred (64) and take the link OOS. The type of errors is limited to alignment errors. The typical cause of alignment errors is usually clock signals that are not properly synchronized on both ends of a link.

The network management procedure at level 3 is responsible for realigning the link (by taking it OOS and resynchronizing the link). Level 2 is responsible for reporting any errors to level 3 link management.

When the link is taken OOS, it must be tested for integrity before it is available for MSUs again. This process is known as the *alignment procedure*. Two types of alignment procedures are used: normal alignment procedure and emergency alignment procedure. Both these methods are discussed in greater detail later in this section.

Signal Unit Delimitation

Every signal unit is preceded by a flag. The flag is an 8-bit pattern, beginning with 0, followed by six consecutive 1s, and ending in 0 (01111110). The flag is used to signify the beginning of a signal unit and the end of the preceding signal unit. Although the protocol actually enables both an opening and closing flag, only one flag is used in the United States.

Signal unit delimitation is important to the upper layers. In most networks, there is traffic constantly flowing through each signaling point, even though the messages may not contain any information. More detailed information is provided in the following section.

Flow Control

Flow control enables traffic to be throttled when level 2 becomes congested at a distant signaling point. The LSSU is used to send congestion indications to the transmitting nodes. When the LSSU of busy is received, the receiving signaling point stops sending MSUs until the congestion is abated.

Flow control also uses a priority for signal unit types to ensure that important signal units such as MSUs are transmitted, even during a congestion condition. Congestion is not the only condition indicated by flow control. Processor failures are also indicated by level 2, meaning that level 2 can no longer communicate with levels 3 or 4.

Flow control at this level indicates congestion with the specified link, not the signaling points. In addition, flow control at level 2 provides priority consideration for signal units, with no regard to the user part.

This is different from level 3 network management, which gives consideration to the user of a signal unit. Level 3 network management controls the flow of messages to a particular level 4 user, whereas flow control at level 2 controls the flow of messages to a link processor.

If the congestion condition should continue, the link will be taken OOS and realigned using the alignment procedure. This prevents the link from becoming locked in the congestion state.

Structure of MTP Level 2

The signal unit components used by level 2 are shown in Figure 5.2. These components can be found in all three types of signaling units. The *backward sequence number* (BSN) and the *forward sequence number* (FSN) are used for sequencing packets and are used by level 2 to ensure that all transmitted packets are received. They are also used for positive and negative acknowledgments. The indicator bits are used to request a retransmission. The *length indicator* (LI) enables level 2 to determine the type of signal unit being sent, and the CRC field is used to detect data errors in the signal unit.

Flag

The flag is used to indicate the beginning and end of a signal unit. As mentioned in the previous chapter, the flag in U.S. networks is used to indicate both the beginning of one signal unit and the end of another. In some other networks, there can be both an opening and closing flag.

The flag bit pattern can be duplicated within the information field of an MSU (ones-density violation), causing an error to occur. To prevent the data from duplicating the flag pattern, the transmitting signaling point uses bit stuffing. *Bit stuffing* is the process of inserting an extra bit before transmission after every series of five consecutive 1s. The bit value is always 0.

By inserting a 0-after-every-five-consecutive-1s pattern before transmitting a signal unit, the transmitting signaling point can ensure that there is never an occurrence of

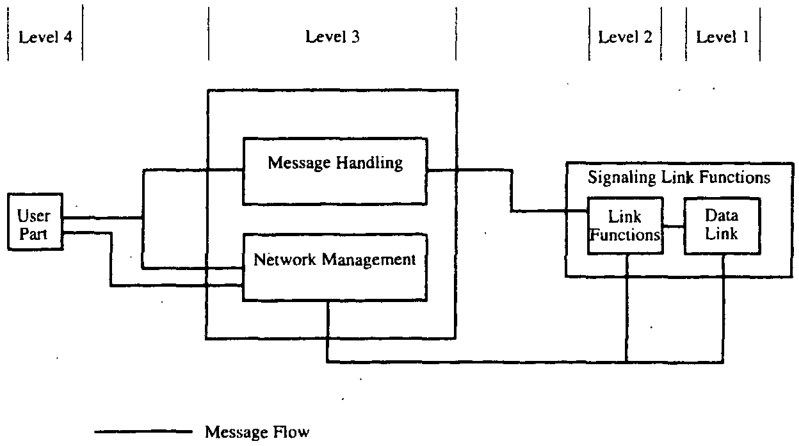


Figure 5.2 Level 2 MTP is responsible for link functions, which include aligning the link and reporting link status to MTP level 3.

six consecutive 1s except for the flag, which is inserted just before transmitting the signal unit.

The receiving signaling point, upon receipt of a signal unit with five consecutive 1s, removes the inserted 0s from the signal unit. Because this is an absolute rule, a 0 is always going to be inserted after five consecutive 1s and is always removed after five consecutive 1s.

When a ones-density violation occurs, the signal unit is considered out of alignment and level 3 is notified of a link failure. The link then enters what is referred to as *octet counting mode*. During this mode, every octet is counted, and the number of errors per octet is monitored rather than errors per signal unit. The link is then taken OOS and put through an alignment procedure.

Sequence Numbering

The SS7 protocol uses sequence numbering like many other layer 2 BOPs. The SS7 technique is just a little different from, say, X.25, but the principle is the same. Sequence numbering is achieved in several ways. When 56 kbps links are used, a 7-bit sequence number is used. Both an FSN and BSN are used on these links. The sequence number used on 1.536 Mbps links is 12 bits in length (if MTP level 2 is used on these links). If *asynchronous transfer mode* (ATM) links are used, MTP is replaced by the *signaling ATM adaptation layer* (SAAL) protocol, which uses a 24-bit sequence number.

Here is how sequence numbering works within MTP level 2. The FSN indicates the number of the signal unit now being sent. This sequence number is incremented by 1 after every signal unit transmission, except in the case of the FISU or LSSU. The FISU and LSSU assume the FSN of the last sent MSU or LSSU and never increment.

The BSN is used to acknowledge received signaling units. For example, if sequence numbers 1 through 7 have been sent and received by the distant signaling point, the next signal unit sent by the receiving signaling point could have a BSN of 7, which acknowledges that all signal units, 1 through 7, were received without error.

The transmitting signaling point maintains all transmitted signal units in its transmit buffer until an acknowledgment is received. When a signal unit is received, the BSN is examined to determine which signal units are being acknowledged. All acknowledged signaling units are then dropped from the transmit buffer.

If signal units remain in the buffer unacknowledged, they will remain until timer T7 times out. When T7 times out, a link failure indication is given to level 3. This causes the link to be taken OOS and placed through the alignment procedure.

Signal units received are checked for integrity (*check bit field* [CRC]), and they are also checked for proper length. A signal unit must be at least six octets in length, or it is discarded and the error rate monitor is incremented. A negative acknowledgment is then sent to request a retransmission of the bad signal unit. A signal unit's length must be in 8-bit multiples; otherwise, the signal unit is in error.

Indicator Bits

The indicator bits are used to request a retransmission. There are two indicator bits: a *forward indicator bit* (FIB) and a *backward indicator bit* (BIB). During normal conditions, both indicator bits should be of the same value (0 or 1). When a retransmission is being requested, the signal unit being sent by the signaling point requesting the retransmission will have an inverted BIB. The FIB retains its original value. This indicates to the distant signaling point that an error occurred and retransmission must take place.

The BSN indicates the last signal unit received without error. The receiver of the retransmission request then retransmits everything in its transmit buffer with a sequence number higher than the BSN of the retransmission request. The indicator bits in the retransmitted message are independent of the originating exchange's indicator bits and do not provide any indication. Therefore, they will both be of the same value during the retransmission.

When the retransmitted signal units reach the distant exchange that originated the retransmission request, an acknowledgment is sent. The FIB in the acknowledgment will be toggled to match the BIB and will remain at this value until another retransmission request.

The process we are describing is referred to as basic error detection/correction. These procedures are discussed further in the following section.

Length Indicator (LI)

The LI is used by level 2 to determine which type of signal unit is being sent. The values of the LI can be

- 0 (FISU)
- 1 or 2 (LSSU)
- 3 or more (MSU)

The LI should match the length of the field immediately following it before the CRC field. This field does not exist in the FISU. In the LSSU, this is the *status field* (SF), which can be one or two octets in length. If the signal unit is an MSU, there are two fields: the *service indicator octet* (SIO) and the SIF. The SIO is an 8-bit field, whereas the SIF is a variable-length field used by level 4.

The maximum length of the SIF is 272 octets. Obviously, the LI cannot accommodate such a large number. Therefore, whenever the SIF length equals 62 octets or higher, the LI is set to the value of 63. Only level 2 uses this LI, and its only purpose is to indicate what type of a signal unit is being received. LIs can be found throughout the level 4 packet to indicate the length of the variable fields within level 4.

Cyclic Redundancy Check (CRC)

The CRC field is the last field in the signal unit. This field is calculated using the fields immediately following the flag up to the check bit field itself. The fundamental process is rather simple. The transmitting signaling point performs the check before bit stuffing and transmission. The remainder is carried in the transmission in the *frame check sequence* (FCS) field. The receiving signaling point then performs a similar calculation and compares the remainder to the FCS field of the received signal unit. If there is a discrepancy, the signal unit is discarded and an error is counted in the SUERM. The CRC-16 method of error checking is used in SS7.

Description of MTP Level 2 Procedures

The previous section provides a basic overview of how MTP level 2 works. The following section will provide more details about the various procedures and functions used within the MTP at this level.

Basic Error Control Method

So far, we have discussed how a link is placed into service when it is initially started and a little about negative acknowledgments and retransmission. After a link has been placed in service, error detection/correction is used at level 2 to maintain proper transmission of SS7 messages. There are two methods of error control: basic error control and PCR. First, we will discuss the basic error control method.

Basic error detection/correction is used whenever a signaling link uses a terrestrial facility. *Land* links can be of any type of medium—it makes no difference to the MTP at this level. Basic error detection/correction is the most favored method.

Basic error control uses the indicator bits within the MTP portion of the signal unit to request the retransmission of signal units received with errors (see Figure 5.3). When an MSU is transmitted, the transmitting signaling point sets the FIB and BIB to be the same. This is the state the indicator bits should always be in when there are no errors. Both the FIB and BIB should be of the same value, but it makes no difference if the value is a 1 or 0.

When a signaling point detects an error, the signal unit in error is discarded and a negative acknowledgment is sent to the transmitting signaling point. The negative acknowledgment may use an FISU, LSSU, or MSU. The BIB is inverted to indicate a retransmission request. Again, the actual value is of no significance; just the fact that the bit is different from the FIB is of significance here.

The BSN should acknowledge receipt of the last good MSU. The BSN is then used by level 2 at the transmitting signaling point to determine which signal units in the

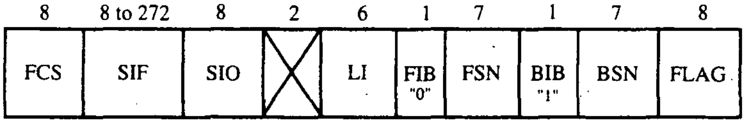


Figure 5.3 This figure illustrates a retransmission request. The BIB has been changed from its previous value, while the FIB remains the same.

transmit buffer to retransmit. All those signal units that have been acknowledged are removed from the buffer, and the remaining signal units are then retransmitted.

When the retransmission begins, the FIB is inverted to match the value of the BIB of the received retransmission request. Both indicator bits should now match again. They retain this value until a retransmission is requested again, in which case the BIB inverts to a value different from the FIB.

Sequence numbers in the SS7 network can be a value from 0 to 127. This is known as *modulo 128*, meaning that 128 sequence numbers can be sent before recycling again. The number of sequence numbers that can be sent before an acknowledgment must be received is referred to as the *window size*.

The larger the window size, typically the better the throughput. This is only true, however, when you have reliable transmission. When the link has too many errors, retransmissions can congest the signaling point at the link level. When there is a large number of signal units to retransmit, the problem is compounded.

The window size of any signaling point is something that must be determined based on the type of transmissions, the capacity of the link, and the average number of retransmissions that occur.

Figure 5.4 shows a sequence of good transmissions with positive acknowledgments. The positive acknowledgment is the sending of a signal unit in the opposite direction with an acknowledgment (the BSN) for previously transmitted signal units. When the acknowledgment is received, the signal units can be dropped from the transmission buffer.

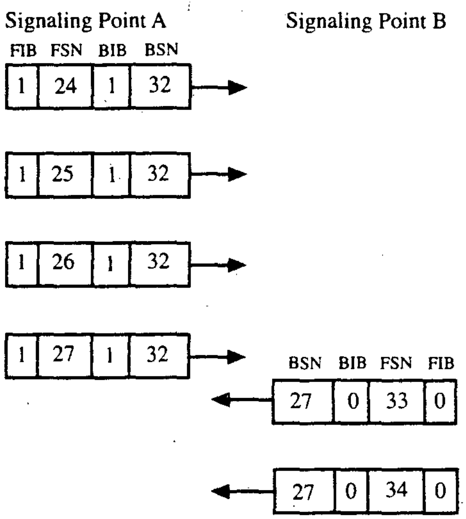


Figure 5.4 This figure depicts a successful transmission sequence between two exchanges. The BSN acknowledges receipt of the previously sent messages. Unlike most protocols that indicate what sequence number they expect next, the BSN value is of the last received sequence number.

It should be mentioned that an acknowledgment does not have to be received after every signal unit. A common practice in all asynchronous protocols is to permit several signal units to be acknowledged in one acknowledgment.

Likewise, an acknowledgment could be received for only some of the transmitted signal units. This does not indicate an error. As seen in Figure 5.5, the acknowledgment could come a little later. Sometimes processors get busy and cannot acknowledge everything at once. As long as the timer T7 does not expire, this does not pose a problem.

Preventative Cyclic Retransmission (PCR)

PCR is used whenever satellite transmission is required for signaling links. This is not the favored method because it uses a higher number of retransmissions than basic error detection/correction does.

PCR does not use the indicator bits when a retransmission is needed. Instead, all unacknowledged signal units are automatically retransmitted during an idle period. Indicator bits could prove to be a frustrating tool with satellite transmission because of the propagation delay introduced with this method of transmission.

The PCR method waits until there are no more MSUs to be sent and then retransmits everything in its transmit buffer. Only unacknowledged signal units will be in the buffer, so, in essence, the signaling point is retransmitting what it perceives to be signal units that have not been received.

If a signal unit has been received and another of the same sequence number is received, the retransmitted signal unit is discarded. If a signal unit is received with a new signal unit, it is processed as normal. Eventually, the distant signaling point will send an acknowledgment for all received signal units.

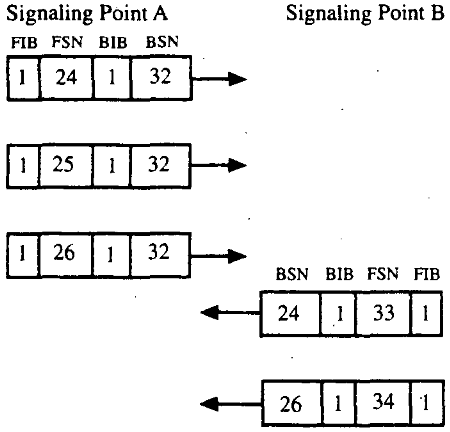


Figure 5.5 This figure appears as if there may have been an error; however, no error occurred. Signaling point B sent an acknowledgment for sequence number 24 and then sent an acknowledgment of 25 and 26. Timers allow this asynchronous acknowledgment of messages without causing retransmission.

If a signaling point is sending retransmissions and it receives additional MSUs to transmit, it stops the retransmission and sends the newly received MSUs. The distant signaling point must be capable of determining which are new MSUs and which are retransmissions.

If the transmit buffer of a signaling point should become full, it stops sending any MSUs and sends the entire contents of the buffer. No new MSUs can be sent during this period. If an acknowledgment is still not received, the buffer is retransmitted again until an acknowledgment is received. This is known as a forced retransmission.

To prevent this condition from occurring, a counter is typically implemented to allow the forced retransmission before the buffer becomes full and MSUs become lost. If the buffer is full, MSUs received are going to be discarded and a busy status will be sent to adjacent signaling points. By using an administrable counter, a threshold can be set to force retransmission when the buffer reaches a predetermined capacity.

The only rule in PCR is the number of signal units that can be sent without acknowledgment and the number of octets that can be sent without receiving an acknowledgment. No more than 127 signal units can be sent without acknowledgment. The number of octets must not exceed the time to send a signal unit and receive an acknowledgment.

Figure 5.6 depicts a sequence of events in which several MSUs have been sent, but not yet acknowledged. The transmitting signaling point does not have any more MSUs to transmit so it automatically begins retransmitting what is in its transmit buffer.

During the retransmission, a group of new MSUs is generated. This results in the retransmission being stopped and the new MSUs being transmitted. Once these have been transmitted, the retransmission begins again.

Finally, the distant signaling point has reached an idle state and begins sending acknowledgments back to the originating signaling point. The theory behind this method is that eventually a signal unit will reach the other signaling point, with or without retransmission. The retransmission may flood the link and push the link beyond its 40 percent capacity, but the messages are guaranteed a higher rate of success if they are continually transmitted in this fashion.

Structure of the LSSU

The LSSU, as shown in Figure 5.7, is used at level 2 to notify adjacent nodes of the status of level 2 at the transmitting signaling point. Link status is carried over the same link for which it applies and is not carried over other links (assuming, of course, that the link is functional at level 2).

If level 2 fails completely, then nothing is received over the link, including FISUs. Level 3 is notified that acknowledgments have not been received, and it initiates a link failure recovery. Link recovery is accomplished through the alignment procedure.

The LSSU is transmitted by the hardware and software associated with a particular link. The link itself may or may not be at fault. The trouble most usually can be found at the termination point of the link—for instance, a link interface card. Because this is where level 2 software resides, this should be the first point of maintenance testing.

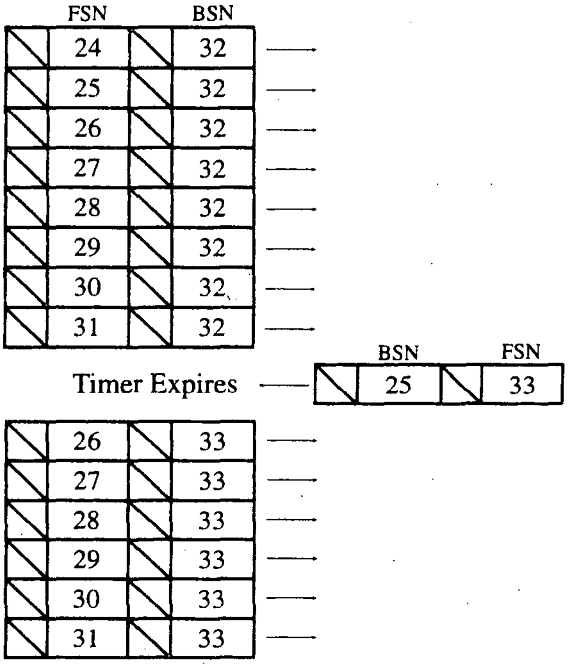


Figure 5.6 With PCR, the transmitting signaling point maintains all messages in its transmit buffer until an acknowledgment is received. The signaling point retransmits all unacknowledged messages after the expiration of a timer or when there is nothing else to transmit.

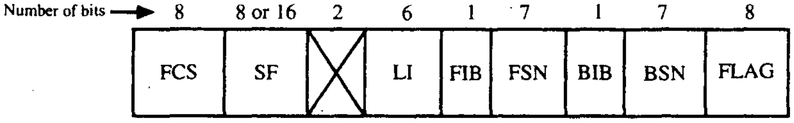


Figure 5.7 Components of the LSSU.

A hardware or software failure at the link interface card causes level 2 software to initiate LSSUs.

The LSSU provides three types of status information:

- Level 2 is congested.
- Level 2 cannot access levels 3 or 4 (processor outage).
- The alignment procedure has been implemented.

The SF is used to provide the information to the adjacent signaling point. The information does not include the link number, which means that this signal unit cannot be used to notify nodes about link status over other signaling links. It pertains only to the link on which it was received.

Another important concept with the LSSU is that the status is really level 2 and 3 status in the transmitting signaling point rather than the transmission facility. Throughout the various SS7 publications, when they talk about the signaling link status and signaling link functions, they are really talking about the level 2 functionality in the signaling point itself.

The SF is an 8- or 16-bit field, although only 3 bits are actually used. The three most significant bits provide the actual status information, whereas the remaining bits are set to 0.

The *status indicator of busy* (SIB) indicates that level 2 is congested at the transmitting signaling point. When a signaling point receives an SIB, it stops the transmission of all MSUs and begins sending FISUs. If the condition persists for three to six seconds, level 3 is informed of a link failure. Level 3 then initiates the alignment procedure to begin on the affected link.

The congested signaling point continues sending SIBs at regular intervals (timer T5) until the condition abates. Timer T5 is a value of 80 to 120 milliseconds. Any MSUs already received by the transmitting signaling point will be acknowledged, but no new MSUs should be sent.

At the receiving signaling point, the timer T7 is reset every time an SIB is received. Timer T7 is used for excessive delay of acknowledgment. If T7 should time out, the link is considered at fault, and level 3 initiates the alignment procedure. To prevent this from occurring, the T7 timer is reset after every SIB.

To prevent an excessive delay caused by a received SIB, a signaling point begins timer T6 upon receipt of an SIB. This timer is used to time the congestion period, preventing congestion at a remote signaling point from causing the network to bottleneck. If T6 should time out, the link is considered at fault and level 3 initiates the alignment procedure on the affected link. The value for T6 is one to six seconds.

To indicate that congestion has subsided and normal processing can begin, the affected signaling point will begin transmitting MSUs again and stop the transmission of SIBs. The receiving signaling point recognizes the absence of SIBs and begins receiving MSUs as normal.

The *status indicator of processor outage* (SIPO) indicates that the transmitting signaling point cannot communicate with levels 3 and 4. This could be caused by a *central processor unit* (CPU) failure or a complete nodal failure. If maintenance personnel have manually placed a link OOS, the link will send SIPOs to the adjacent signaling point as well. Level 2 is usually functional because it resides within the link interface hardware. The signaling point sends an SIPO to notify the distant signaling point to stop sending MSUs.

Upon receipt of an SIPO, the transmission of all MSUs is stopped, and FISUs are sent to the affected signaling point. The transmitting and receiving nodes stop level 2 timers T5, T6, and T7. If the condition persists for too long, then the link has failed and level 3 initiates the alignment procedure on the affected link.

Even though there has been a processor outage, level 3 may still be capable of controlling level 2 functions. When this is the case, level 3 may request level 2 to empty its buffer (both receive buffer and transmission buffer). When this is the case, all MSUs in the buffer are discarded. When an MSU is received from a remote signaling point, level 2 of the affected signaling point sends an FISU with the FSN and FIB set to the same value as the BSN and BIB of the last MSU received from the remote signaling point. Normal processing of all messages then resumes.

The LSSU also sends a *status indicator of out of alignment* (SIO). This condition occurs when a signal unit is received that has a ones-density violation (the data field simulated a flag) or the SIF has exceeded its maximum capacity of 272 octets. The SIO is sent when a link has failed and the alignment procedure is initiated.

An LSSU *status indicator of out of service* (SIOS) indicates that the sending signaling point cannot receive or transmit any MSUs for reasons other than a processor outage. Upon receipt of an SIOS, the receiving signaling point stops the transmission of MSUs and begins transmitting FISUs. The SIOS is also sent at the beginning of the alignment procedure.

Link *status indicator of normal* (SIN) or *emergency* (SIE) indicates that the transmitting signaling point has initiated the alignment procedure. The link is made unavailable for MSUs and only FISUs are transmitted to the affected signaling point until a proving period has been passed (see the section "State 03—Proving" for information on proving periods). After the successful completion of a proving period, MSUs can be transmitted over the link to the affected signaling point.

In the event that an LSSU is received with errors, the receiving signaling point discards the signal unit. Retransmission is not requested of the LSSU. The FSN of an LSSU does not increment, but it assumes the value of the last transmitted MSU. The BSN does increment when an acknowledgment is being sent to the distant signaling point.

The LSSU is processed within level 2 and does not get passed to level 3. However, level 2 may pass control information to level 3, depending on the status of the LSSU. For example, if an LSSU with a busy status is received, level 2 notifies level 3 to stop the transmission of MSUs.

It is also important to remember that the LSSU works independently of network management, which is a level 3 function. In fact, network management uses the MSU to send management information to other nodes and can use any link or route to reach adjacent nodes. Level 3 is used when links have failed altogether and LSSUs cannot be transmitted over the affected link.

Most level 2 problems are caused by hardware failures. Therefore, they do not require the same level of sophistication as software problems. Level 3 and 4 errors are normally software related, and they require more sophisticated reporting mechanisms. The recovery procedures used by maintenance personnel will also be very different for level 3 and 4 problems from those at level 2.

Signal Unit Alignment Procedure

The purpose of the alignment procedure is to reestablish the timing and alignment of signal units so that the affected signaling point can determine where signaling units begin and end. In other words, when a node receives a signal unit, it needs to know that a bit begins on a particular clock tick.

As previously mentioned, the out-of-alignment condition occurs when the flag has been simulated within the data (ones-density violation) or the SIF is too long (longer than 272 octets), which would indicate that a flag was missed.

The procedure resets both the transmitting and receiving nodes at level 2 and does not affect other links at either signaling point. The procedure also provides testing for a given period of time to ensure that the link transmission is reliable, preventing further errors from occurring.

Two alignment procedures are used: normal alignment and emergency alignment. Normal alignment is used when other links are associated with the affected link (such as in a linkset). The other links must be going to the same destination. An emergency alignment is used when no other links to the adjacent signaling point are within the linkset. The emergency alignment uses the same procedure, but within a shorter time period. Level 3 is responsible for determining which alignment procedure to use.

Four states are entered during alignment. Timers associated with each state ensure that the signaling point does not get stuck in any one state. When any of the timers times out, the alignment starts over again. The following explanation describes each state and the events that occur during these states.

State 00—Idle

This state indicates that the procedure is suspended, and it is the first state entered. State 00 is resumed whenever the alignment procedure is aborted (due to excessive errors). During the time a link is in a proving period, level 3 network management reroutes signal units to other links. If a link should fail the proving period, level 3 places the link back into state 00 for a specified time period (level 3 timer T17).

To prevent rapid link oscillation between in-service and out-of-service states, the level 3 timer T32 is used. When a link is placed back into the alignment procedure, timer T32 is started. If the link fails during T32, the link is placed back into state 00 until T32 expires. Attempts by the remote signaling point to begin alignment of the link are ignored during this time. The signaling point responsible for initiating the state 00 will transmit LSSUs with the SIOS until timer T32 expires.

State 01—Not Aligned

This state is entered when initiated by level 3. The signaling point initiating the link state will send an LSSU with an SIOS and start the level 2 timer T2 (not aligned). Timer T2 for normal alignment is set to 11.5 or 23 seconds (this is determined during the configuration of the signaling point itself).

In some networks, the signaling points have the capability to make links inactive, allocating them for other traffic (such as bearer traffic). When the link is in need due to traffic, it can then be reassigned as a signaling link. This is referred to in the standards as *automatic link allocation*. When signaling points use this function, timer T2 must be set with different values at either end of the signaling link.

The signaling point continues to send LSSUs with the value of SIO until level 3 initiates the next state.

State 02—Aligned

This state;aligned state indicates that the link is aligned and is capable of detecting flags and signal units without error. Remember that out of alignment means that the signaling point can no longer delineate signal units based upon receipt of a flag. What this really implies is that the link has lost its timing and it no longer recognizes the beginning and end of a signal unit. This state indicates that the link is now capable of detecting flags and recognizes the boundaries within the signal unit itself.

During the time that the link is in state 02, the level 2 timer T3 is started. When the link leaves state 02 and enters state 03, T3 is stopped. If T3 should time out, the link is returned to state 00 and the process begins all over again.

State 03—Proving

The proving period is used to test the integrity of the link and level 2 at the signaling point. During the proving period, LSSUs with the value of SIN or SIE are sent and errors are counted. There are two proving periods: normal and emergency. The proving period of normal lasts for 2.3 seconds, during which time no more than four errors may occur while in state 03. The AERM keeps count of all errors received during the proving period. The AERM is an incremental counter, which counts all transmission errors, including CRC errors and ones-density violations.

During the proving period, FISUs are sent on the link. The LSSU of normal or emergency (SIN or SIE) is also sent to indicate that the link is in a proving period.

The emergency proving period lasts for 0.6 seconds, during which time no more than one error may occur. This is also monitored by the AERM. When excessive errors have occurred according to the procedure, the link is returned to state 00 and the process begins all over again.

If a link cannot be restored, it is returned to the idle state and the alignment procedure is repeated until either the link is restored or maintenance personnel detect the repetitive failure and take corrective action. When a link is found to be continually failing or in constant alignment, it usually indicates an equipment failure, which can be resolved by manual intervention (replacing the link interface card usually fixes the problem). Maintenance personnel should always be monitoring the status of signaling links and watching for links that continuously fail. Traffic measurements provided by most signaling points can be a useful tool in determining the number of failures during a given period of time.

Level 3 Alignment Processes

After a link has successfully passed the alignment procedure, the link is returned to an in-service state, where MSUs are transmitted and normal processing is allowed. The link is placed into a probationary period by level 3, which lasts a period set by level 3 timer T33. If the link fails during the probation, it is placed back into suspension (state 00), during which time level 1 sends LSSUs with an SIOS. All attempts to place the link into alignment are ignored until the level 3 timer T34 expires.

Timer T34, which is the suspension timer for link oscillation, is also used to prevent links from rapidly oscillating from an in-service to out-of-service state. This set of level 3 timers, T32, T33, and T34, are all used to filter link oscillation and are only required in Bell System networks at STPs.

Upon completion of the alignment procedure at level 2, level 3 begins a signaling link test to determine if the link is capable of carrying traffic. This is an option in the ANSI recommendations, but it is a requirement within the Bell System networks.

The Bell System networks also use a mechanism for the automatic allocation of signaling links in the event that there is a signaling link failure within the linkset and additional links are required. This procedure uses a predefined set of links that are used for other applications (such as voice transmission) and, when needed, places them into the alignment procedure for preparation as signaling links within the SS7 network.

This process ensures that there are always links available, even when the designated active links have failed and cannot be restored. Level 3 is responsible for the activation and restoration of these links, and for assigning them to the proper linkset. Link management at level 3 is the function that looks after this.

Level 2 Signaling Link Test Procedures

Some automated test messages are used on signaling links to verify the integrity of a link as well as detect possible looping. Looping is a condition that occurs when errors are made in provisioning routing tables. They can be difficult to detect and costly to the carrier.

The *signaling link test message* (SLTM) is sent on a link whenever the link is activated or restored by level 3 (after link alignment, for example). The content of the SLTM is a test pattern. The signaling point responsible for activating the link or restoring the link is also responsible for initiating the SLTM and determining whether or not the test pattern originally sent is actually returned.

The SLTM is first sent by a signaling point and routed back (via loopback) by the adjacent signaling node. The purpose is to verify that the routing in both nodes is correct (otherwise, the SLTM would be sent to another destination), and the data contained in the SLTM is not corrupt. If the SLTM is sent and returned correctly, the link is passed.

This test is performed every time timer T2 expires, regardless of the congestion status on the link. In other words, this is a mandatory test. The test can be performed automatically and manually.

Chapter

6

Message Transfer Part (MTP) Level 3

This chapter will discuss *Message Transfer Part* (MTP) level 3 functions as used in channelized networks. Some of these functions have been implemented in new *Internet Protocol* (IP) protocols specifically for the transport of *Signaling System 7* (SS7) over IP networks. The purpose of adding the functions and procedures discussed in this chapter to IP networks is to ensure the reliability of the signaling network and guarantee the delivery of signaling messages when using *Transmission Control Protocol / Internet Protocol* (TCP/IP) facilities.

Therefore, it is important to understand MTP level 3 even though you may not be using *time-division multiplexing* (TDM) networks. You will find that MTP level 3 is still supported in the signaling nodes (*service switching points* [SSPs], *signal transfer points* [STPs], and *service control points* [SCPs]), although IP is the transport.

There are two categories of functionality at this level: signaling message handling and signaling network management. Signaling message handling is used for routing messages to the appropriate link and determining if messages will be addressed to the received node or if they will be forwarded. Signaling network management is used to reroute traffic to other links when nodes become unavailable.

MTP level 3 relies on the services of level 2 for the delivery of all messages. The interface between the two levels consists of a set of primitives. Primitives enable parameters to be sent to level 2 for routing over the network in the form of SS7 messages. At the same time, the primitives enable level 2 to send parameters to level 3 for message processing.

Message processing begins at level 3. Level 3 must determine the destination of a message and the user of a message, and maintain the status of the network. Level 3 uses primitives to communicate to level 4 users. Parameters are sent to level 4 through these primitives.

In the same fashion, primitives enable level 4 to send parameters down to level 3 for inclusion in a signal unit and transmission over the network. These primitives use the same structure as the primitives used between level 2 and 3.

Message Handling Overview

Message handling includes three different functions: message discrimination, message routing, and message distribution. The interface with level 2 is the message discrimination function. Message discrimination determines what the destination signaling point is by reading the routing label of the message. If the *destination point code* (DPC) is the same as the signaling point's self-identification (its own address), the received signal unit is given to message distribution.

Message distribution is the mechanism used to deliver a message received by a signaling point when the DPC is its own. The distribution function must deliver the message to the proper user part or network management function.

The message-handling function uses the routing label found in all level 4 messages to determine who the originator is and who the destination is. It should be noted that a signaling point might have more than one point code. In addition to its own self-identification, the signaling point can also possess a capability point code, or an alias point code. This enables a signaling point to be partitioned into separate entities. For example, it may be advantageous to assign a point code to an STP for gateway services and another point code to the same STP for global title services.

The *service indicator octet* (SIO) is used by message distribution to determine the user part. The user (usually level 4, but not always) can be any user part or network management. In the case of network management, the SIO indicates whether the user is network management or network management testing.

Message routing is used to pass a message back to level 2 for routing over the network. If message discrimination determines that the message is not addressed to the receiving signaling point, it sends the message back to message routing. If level 4 has generated a message for transmission, it is also given to message routing.

Message Discrimination Overview

Discrimination applies to all received messages. Message discrimination uses the routing label of the *message signal unit* (MSU) to determine the destination of a message. The routing label provides the origination address (*origination point code* [OPC]), the destination address (DPC), and the *signaling link code* (SLC).

Message discrimination reads the routing label, specifically the DPC, to determine if the destination is the receiving node or if the message must be transferred to another node. If the message is intended for the receiving node, then the message is given to the message distribution function, which must determine which user part will receive the message (user part is the level 4 function, such as *ISDN User Part* [ISUP]).

The user is determined by reading the SIO field. This field is divided into two parts: the service indicator and the subservice field. The service indicator identifies the user part to receive the message. A user part does not have to be a level 4 function. Network management and network management testing are level 3 functions that use the information field of an MSU.

The service indicator can also indicate whether special or regular testing messages are being sent by network management. This is also used by level 3 network management, each requiring a unique procedure.

The SIO subservice field identifies whether this message is from an international or national network. This part of the message is used to identify the type of point code structure used, so the discrimination function can determine how to read the routing label. If the value of the network indicator in the subservice field indicates an international network, the routing label uses a different structure than it would for a national message.

The national network indicator does not necessarily mean that this message came from an *American National Standards Institute* (ANSI) network. The national network indicator is used by the *International Telecommunications Union—Telecommunications Standardization Sector* (ITU-TS) to differentiate between various structures used within different countries. This enables each country to use its own point code structure within its own national network while still conforming to the international standard set forth by the ITU-TS.

If the SIO indicates that this is a national message, the SIO can also be used to indicate two different versions of a national structure. The spare, or reserved, code in the subservice field can be used to indicate a different version of the national structure.

For example, a country may have two unique point code structures used in different segments of its network. This could be indicated by using the reserved bits in the national indicator of the subservice field. In ANSI networks, these bits are used to indicate the priority of an MSU.

Network management during periods of congestion and rerouting procedures uses priorities. Recommended values for the priorities are given in the ANSI and Telcordia standards, but they can be network dependent.

If used in a private network that does not access the *Public Switched Telephone Network* (PSTN), the network indicator does not have to be used (because the network is closed and it will always access the same network). This field can then be used as an extension of the service indicator, providing additional user part identification. The implementation of this feature is network dependent, and it is not defined in the ITU, ANSI, or Telcordia standards.

When it is determined that the message is to be routed to another signaling point, it is given to the message-routing function. The message-routing function must determine which link to choose to reach the destination node. In many cases, the next node to receive the message will not be the final destination. Other STPs may receive the message before it reaches its destination. A routing table is used to determine the shortest path to reach the destination. The objective in routing is to prevent multiple hops between nodes, introducing delay in the transmission. The routing table uses a priority mechanism for determining the most efficient route, that is, the route with the most direct path and the fewest hops.

If the message is for another node and the message-routing function determines that it cannot reach the destination, network management must respond. Network management is invoked when a signaling point failure or link failure is detected. We will discuss these messages later in the chapter.

If the point code in the routing label of a message is the point code of an alias, the message is given by message distribution to the *global title translation* (GTT) function. If the GTT function is not available at the signaling point, the message is given to message routing for transmission back onto the network.

The GTT function is provided by STPs and can be located regionally or within several STPs. Two levels of GTT exist: partial and final. Partial global title routing provides the point code of the STP responsible for providing final global title, whereas final global title provides the point code and subsystem of the SCP responsible for providing the data being queried.

Message Distribution Overview

If the DPC is the address of the receiving signaling point, then the message is given to the distribution function. Message distribution must determine who the user is for any given message. This is determined by examining the SIO service indicator field. As mentioned before, the user can be a level 4 function or level 3 network management.

In the event that a user is not available, a *user part unavailable* (UPU) message is sent by network management as an indication to the originating point that the message was discarded because the user part function was not equipped at the destination or was unavailable. The UPU message also provides a cause code, although this cause code is not very specific. Three causes are provided: the user part function was not equipped at the destination, the user part function was not accessible (that is, out of service due to a processor outage), or the user part function could not be reached for an unknown reason. The UPU message is shown in Figure 6.1.

Some user parts are processes that reside within a central processor at the signaling point. For example, the *Signal Connection Control Part* (SCCP) may provide GTT, which is considered a user part in this case. When the processor at the signaling point becomes congested or unavailable, the UPU message is generated.

This does not indicate a problem with the signaling point, however. The signaling point may be able to carry on with other processing, such as levels 2 and 3, and even level 4. This, of course, depends on the architecture of the signaling point itself. When distributed processing is implemented, the impact of a UPU message should be minimal from a signaling point perspective.

Message Routing Overview

If a message is received from another destination, then message discrimination passes the message onto message routing for transmission back onto the network. This is a common function of the STP. An end signaling point, such as an SSP, would not likely receive messages that need to be transferred, unless *full associated links* (F-links) were used. Level 4 uses the routing function to send messages originated by the point code out onto the network.

FCS	Cause	User ID	Destination	H1	H0	Routing Label	SIO		LI	FIB	FSN	BIB	BSN	Flag

Figure 6.1 This figure illustrates the components of the UPU message.

The routing function must first determine what the destination address is by reading the routing label and looking for the DPC. In an ANSI network, the subservice field of the SIO indicates a national network, which means the point code will be a 24-bit field. If the subservice field indicates that the message came from an international network, then the point code will be a 14-bit field.

The international point code structure is very different from the ANSI structure. The first value indicates the zone of the address. The zone is a 4-bit value and can be used to address a country or a group of countries. The next field is the area identifier. This is an 8-bit field and identifies the network of the address. The third field is the individual signaling point identifier, which identifies the actual member of the network. A member is any signaling point on the network.

It is important to note that the international point code structure is only valid at the international hierarchy. Once a particular gateway is reached, messages are converted into national formats, where the point code structure and protocol rules can change. This enables each country to deploy SS7 in its network to meet the needs of the individual country rather than to define a standard that does not address everyone's needs. The ITU-TS distributes point codes for use in the international network.

In the United States, the national standard is ANSI. The ANSI standard is used in the networks of the *Regional Bell Operating Companies* (RBOCs), but it is modified (as published by Telcordia) for the specific requirements set forth by the BOCs. The point code structure defined for use in the United States is quite a bit larger than the ITU-TS version.

The first field in the point code is the network identifier, and it is used to identify the individual network being addressed. This is an 8-bit field. The network identifier is reserved for only the largest of service providers. Those who receive a unique network identifier can also use all numbers in the cluster and member fields.

Network identifiers 1 through 4 are reserved for medium networks. Network identifier 5 is reserved to identify small network clusters. There is a network cluster code associated with each state and territory in the United States and Canada. These are used to identify smaller networks. Medium networks are those that do not have enough signaling points to warrant the distribution of an entire network number. Currently, only network identifiers 1 and 2 have actually been assigned to service providers. The cluster field then identifies the individual networks.

The cluster field is also an 8-bit field and is used in two different ways. In the case of those networks that have their own network identifiers, the cluster number can be used to group STPs together to provide more efficient route management. In medium networks (network identifiers 1 to 4), the cluster field identifies the individual network.

Small networks (network identifier 5) use cluster value 1 or 2. The member codes are then divided into three code blocks and given to very small networks with only a few signaling points. The codes are assigned in blocks, depending on the location and number of signaling points on the network. The member value of 0 is reserved for STPs.

Two forms of routing may be deployed within any signaling point: full point code routing and partial point code routing. In the case of full point code routing, the signaling point looks at the entire point code to determine how the message will be routed. This means the routing table for that signaling point must include all point codes for all connecting signaling points and endpoints on the network.

This can be quite cumbersome in larger networks with many point codes. For this reason, some networks use partial routing. Partial routing enables signaling points to route messages based on the network identifier or the cluster identifier.

With cluster routing, only the network identifier and cluster identifiers are entered into the routing tables of intermediate STPs. The STPs are all given the value of 00000000. All other signaling points are given full point codes as the STP will perform final routing to these destinations.

In some networks, only the network identifier is used for intermediate routing (known as network routing). The rules for network routing are the same as with cluster routing; it can only be used for intermediate routing. Network routing is also beneficial for gateway STPs, which provide access into different carriers' networks.

If the STP is providing global title functions, then the cluster number assigned to the STP must be different than the rest of the network. If this is not possible, then an alias point code must be used with a different cluster number for global title. This is also a benefit with network management, enabling the STP to be addressed directly by level 3.

Normal Routing Procedures

Before a message is transmitted, the routing table must identify which link and linkset to route the message to (see Figure 6.2). Every routing table must indicate the primary route to a DPC and an alternate route(s). Each route is assigned a priority, which indicates whether it should be the first choice, second choice, and so on. Some equipment manufacturers use different techniques for identifying the priority of a route, but the basic concept is the same.

The rule is to always choose the most direct route to any destination. The most direct route is always the route with the fewest hops through other signaling points. In the

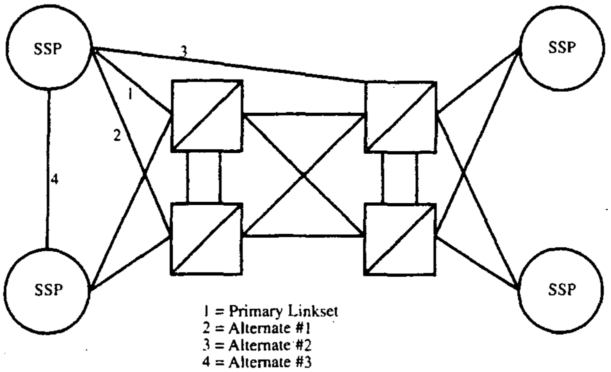


Figure 6.2 This figure shows a typical routing scheme for SS7 entities. The numbers indicate the choice, or weighting factor, assigned to each of the links. For example, the 1 indicates a first-choice route, and so on.

case of an end signaling point, if an F-link is available, it should be first choice if it terminates at the destination. If an *extended link* (E-link) is available, it should be the second choice. Access links (A-links) are last-choice routes. Of course, in many cases, the A-link may be the only route to a DPC, in which case it will be the first choice.

In networks where traffic is sent through other carriers, the routes are selected based on carrier agreements. Depending on the traffic destination, some routes through specific carriers may be cheaper than other routes (as determined by intercarrier connection agreements). This must also be considered when creating routing tables.

For an STP, if the destination is an endpoint (endpoints are SSPs and SCPs), then an A-link should be first choice. If there are no available A-links to the DCP, then an E-link should be used (if one exists). If the message must go through another STP to reach the DPC, then the STP that the destination "homes" to should be the next destination for the message. The home STP is that which connects directly with the signaling point (see Figure 6.3).

In the event that the home STP cannot be reached directly, then the message should be routed to a primary STP in a two-level hierarchy. The primary STP of the DPC should be the STP used to reach the home STP. In the event that the primary STP cannot be reached, then the primary STP of the message originator should be used.

When IP facilities are used for signaling links, the routing is somewhat different. The link selection becomes more of a virtual selection; however, the SLC code is still significant. Depending on the protocol being used to provide level 3 services, the SLC can be maintained by either encapsulating MTP level 3 into an IP message and transmitting to the distant end for processing, or emulating the SLC by an equivalent code within the IP stack.

Link Selection

The signaling link used to route the call is determined by the value of the *signaling link selection* (SLS) field located in the SLC field of the routing label. The SLC code is a preassigned code representing a physical link. The SLC is not the link number, but is a logical connection to a physical link.

All links are assigned an SLC and terminal number. The terminal number is a logical connection to a physical link. The SLCs range from 0 to 127. This means that each

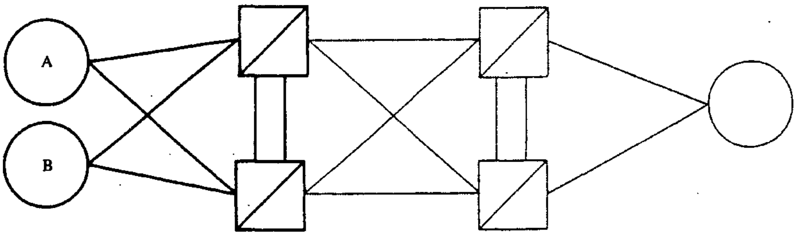


Figure 6.3 In this figure, both A and B have the same home STP. A home STP is one that is directly adjacent to the SSP.

link will have more than one SLC. When a link is deployed, it is up to the equipment software to reallocate the SLCs to accommodate the newly equipped link. When a link fails, software will reallocate the SLCs again so that the failed link no longer has an SLC.

Before transmission, level 3 routing determines which link will be the next outgoing link. This is based on several factors. First, all links within a linkset must carry equal traffic (load sharing). This means no link will carry more than the others. The MSU size must also be figured in the traffic capacity of every link so that one link does not become burdened with large messages while another only sends small messages.

After a link is selected, the SLC of the physical link is placed in the SLC field of the routing label and sent to level 2 for transmission. When the message is received at the remote signaling point, the bits in the SLC field are rotated one position to the right by level 3 routing. Currently, only the five *least significant bits* (LSBs) are rotated, even though the standards now support an eight-bit SLC field. The three *most significant bits* (MSBs) are used for link selection in networks where 8-bit rotation is supported (this is optional today and will slowly be implemented into all networks).

There are exceptions when bit rotation is not to be used. Call setup messages and database query messages are examples of messages that should not use bit rotation. These messages usually require more than one signal unit to be sent. These signal units must be received in order; otherwise, an error will occur. To prevent them from taking different routes and being received out of order, the messages relating to one transaction or to the same circuit connection always follow the same path (use the same links from one signaling point to another signaling point). *Cross links* (C-links) between mated STP pairs do not use bit rotation.

There are also certain maintenance messages that must be routed over specific links. For example, a *changeover order* (COO) is always sent over an alternate link, but the acknowledgment of a changeback order can be routed over any available link. Level 3 routing must also take these rules into account before selecting a link and rotating the SLS bits.

Load sharing must be used whenever there is more than one link or one linkset. In the event that there is more than one linkset, the LSB is used to select the linkset to be used. The remaining 4 bits then identify the link within the linkset. In the event that only one linkset exists, the LSB is used as part of the link identifier, but the MSB (bit 5) is left unused (maintaining a 4-bit link selection code).

In the international network, load sharing is not used. The SLC field is used to identify the voice *circuit identification code* (CIC) for all *Telephone User Part* (TUP) messages. For *Data User Part* (DUP) messages, the bearer identification code is identified in this field. Bit rotation is not used in international networks.

Network Management Overview

Network management at level 3 provides the procedures and functionality to reroute traffic through alternate links and linksets or to control the flow of traffic to a specified DPC. Three separate network management functions are used in SS7:

- Traffic management
- Link management
- Route management

Traffic management is used between two signaling points to divert traffic away from failed links. Traffic management messages are originated by the signaling point, which detects a problem in a link. The traffic link management message is sent over an alternate link to inform the adjacent signaling point not to route messages over the affected link. Traffic management also triggers route management from the receiving signaling point to its adjacent nodes.

Traffic management messages are not propagated through the SS7 network. They are only point to point. The difference between traffic management and link management (which incorporates the use of level 2 *link status signal units* [LSSUs]) is the fact that the LSSU is carried over the signaling link that is in question.

This fundamental difference points out that the link must be capable of carrying level 2 traffic before LSSUs can be used while not being able to carry level 3 and 4 traffic. In the event that a link cannot carry level 2 traffic (such as in the case of a complete link failure), then level 3 traffic management will be implemented to inform adjacent signaling points that the signaling link can no longer be used.

Link management consists of activation and deactivation procedures, as well as link restoration. These functions are used with level 2 to restore failed links back into service. The link management function triggers level 2 link alignment procedures and guides the transitions through the various states of alignment.

Route management is used to divert traffic from a specific signaling point. This is a function provided only by STPs, adding an important function to the network. Route management does not pertain to any one specific link as traffic management does, but to an entire signaling point. Route management is triggered by traffic management messages and is sent to the adjacent STPs when a traffic management message is received.

For example, in Figure 6.4, signaling point F has failed its link to signaling point D. Signaling point F sends a changeover message to signaling point D through signaling point E (traffic management). Signaling point E has no knowledge of what the traffic

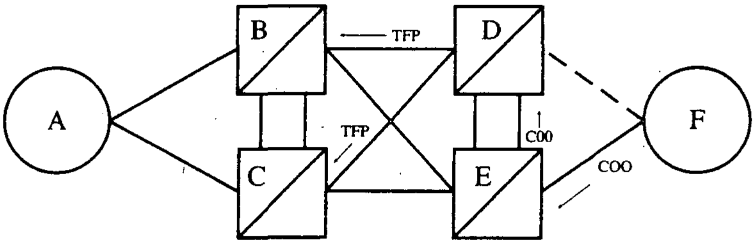


Figure 6.4 This figure depicts traffic management and route management.

management message is about, nor does it care. It is transferring the message onto its final destination.

Signaling point D, upon receipt of the changeover message, will send a *transfer-prohibited* (TFP) message to its adjacent nodes. The TFP message indicates the DPC that is no longer accessible through signaling point D. All adjacent signaling points then invoke rerouting procedures to begin routing traffic destined for signaling point F through signaling point E because this is the only available route. If there were other links available through D, then those links would be used to route traffic to F and the TFP would probably not be sent because signaling point F can still be reached through D.

Link management is then invoked by signaling point F to restore the failed link. This triggers level 2 to begin the alignment procedure and controls the transitions from one state to the next through the use of timers. When the link has been restored by level 2, level 3 link management is notified and traffic management sends a changeback message to restore traffic back to the newly restored link.

Upon receipt of a changeback message, signaling point D invokes route management to alert adjacent nodes that traffic destined for signaling point F can now be routed back through signaling point D.

This is only one example of how these three network management functions work together. As we discuss each management function, we will discuss specific examples in greater detail.

Signaling Network Management Procedures

As we mentioned before, there are three types (or categories) of network management. Traffic management diverts messages away from failed links, link management is responsible for the activation and deactivation of signaling links, and route management is responsible for the rerouting of messages around failed signaling points. Route management also controls the flow of messages to a signaling point.

Network management provides a layered approach to managing troubles on the network. Procedures are provided that deal with network congestion and outages from the link level all the way up to the route level.

When an error is encountered, level 2 reports the error to level 3, which must then determine which procedures to invoke. Procedures begin at the lowest level, the link level, and work their way up to the route level.

Link management directs the procedures at the link level. These procedures do not have a direct impact on routing or the status of signaling points. They trigger other network management events at level 3, however.

Link management does have an impact on other functions, but it is indirect. Traffic management is affected by link management, primarily because traffic management must divert traffic away from a link that link management has failed and removed from service. Traffic management ensures the orderly delivery of diverted traffic, providing for the transfer of unacknowledged messages to another link buffer and the retransmission of messages on a different link.

Traffic management does not divert traffic away from a signaling point. The purpose of traffic management is to redirect traffic to different links. However, traffic management does impact routes and routesets to specific destinations. If a particular route is used by another signaling point to reach a destination and traffic management has diverted traffic away from that route, adjacent signaling points may have to invoke route management procedures.

Route management diverts traffic away from signaling points that have become unavailable or congested. The reasons vary, but regardless of the reason, traffic management and link management will be involved at the affected signaling point. In the meantime, all the signaling points around the affected signaling point invoke route management procedures to prevent messages from becoming lost.

This layered approach to network management is what makes SS7 networks as robust as they are. Very few network troubles actually cause the network to fail. In fact, there are very few network outages at all in SS7. The protocol maintains a very high level of reliability, which is mostly due to the MTP and the network management procedures discussed in this section.

In this section, we will review each of the messages used in network management and the procedures used. We will look at the structure of each message type and the parameters used.

Network management messages use the MSU structure, as shown in Figure 6.5. The routing label is used for routing the message to the appropriate signaling point. The *signaling information field* (SIF) holds information concerning the point code experiencing the failures or the link that has failed. In addition, status codes, priority codes, and other maintenance codes can be included.

Because of the nature of these messages, the information field varies from one network management message to the next. As we discuss each type of network message, we will discuss the structure of the information field and how it is used.

Link Management Procedures

Link management provides the procedures necessary to manage the individual links within a signaling point. Link management does not view links from a linkset or even a route perspective, but from an individual perspective.

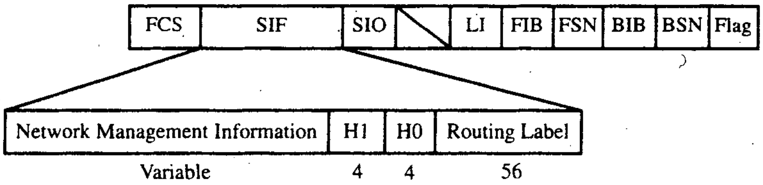


Figure 6.5 Network management messages use the MSU. The H0/H1 field is the label that identifies the type of network management network message being sent.

The management message (as shown in Figure 6.5) uses an H0 and H1 heading code to identify the type of management message being sent. The H0 heading code identifies the type of message, whereas the H1 heading code identifies the type of message for the procedure being declared. The following are the values for the H0 heading codes:

H0 Heading Codes	
0000	Spare
0001	Changeover and changeback messages
0010	Emergency changeover message
0011	Transfer-controlled and signaling-route-set-congestion-test messages
0100	TFP-allowed and -restricted messages
0101	Signaling-route-set-test messages
0110	Management-inhibiting messages
0111	Traffic restart messages
1000	SDLC messages
1001	Spare
1010	MTP user flow control messages
STOP	

The link management procedures require the use of MTP level 2 functions or *signaling ATM adaptation layer* (SAAL) to report failures and link status to an adjacent signaling point. These messages are not propagated through the network. Link management is only concerned about the local end of a link.

Link management provides three functions:

- Link activation
- Link restoration
- Link deactivation

These three functions are explained as follows in greater detail.

Link Activation. When a link is first activated, level 3 directs level 2 to begin the alignment procedure (as described in Chapter 5, "Message Transfer Part (MTP)") and place the link in service. Before messages can actually be transmitted over the link, link management also sends test messages over the link to ensure the integrity of the link.

Level 2 is used to inform the adjacent signaling point of the activities at level 3. This is accomplished using the LSSU, as described in Chapter 5. The LSSU identifies the status of the link, but does not instruct the adjacent signaling point on any procedures.

Once the link has been activated and is considered in service, a *signaling-link-test message* (SLTM) is generated and transmitted over the link. Upon acknowledgment of the SLTM, (*signaling-link-test acknowledgment* [SLTA]), the link is restored to service and traffic is enabled over the link. In the event that the link has a failure (determined and reported by level 2), link management invokes the restoration procedures.

Link Restoration. Restoration involves the LSSU to inform the adjacent signaling point of the events taking place. The signaling point that detected the errors is responsible for invoking alignment procedures and notifying the adjacent signaling point about the status.

Level 3 timers are used to control the procedures and ensure that the link does not get caught in an endless loop of alignment procedures. When a link has successfully passed the alignment procedures, level 3 generates an SLTM and transmits the test message over the network (as it did during activation). When an SLTA is received for the SLTM, the link is considered restored and level 3 changes the local status to available and in service.

Link Deactivation. Link deactivation is used when a link is found in error and needs to be placed in alignment. Link deactivation must first stop all traffic to the link, which invokes traffic management procedures (the diversion of traffic from the failed link). Link management then disconnects the link from its logical connection (the SLC).

Every link has a logical connection. In any signaling point, a number of logical connections are available for a link. All logical connections must be associated with an SLS code. This code is a dynamic code assignment, which changes according to the link status.

Each link within a linkset is given a unique SLS code, which is used by routing to determine which links a message should be routed over. This is directly related to the bit rotation procedure discussed earlier in the section, "Message Routing Overview."

The bits rotated are the SLS bits. A link may have more than one SLS code. All codes must have an assignment because of the way the bit rotation is used to select a link. When any link becomes unavailable and must be placed out of service, link management must disconnect the link from its logical connection. This means the table will change. One can view this table as a mapping to every link in the system. Each signaling point must have a similar table in software that enables it to choose signaling links based on status.

Link management changes the table based on the local status (reported by level 2) or the remote status (reported by level 2 via the LSSU). When a link becomes available again, it is reconnected to its logical connection.

Notice that all links are listed in numerical order. This ensures that the assignment is linear and not random, which enables both ends of the link to remain synchronous. Because the SLC is transmitted in maintenance messages, the code must be the same at both signaling points.

Link management can also drive a function referred to as *automatic link allocation*, which enables links from one linkset to be disconnected (logically) by link management and reassigned to another linkset. This requires a link to be connected to the same signaling point as the failed link that it is replacing. This feature is usually invoked by a threshold value, which is typically administrable. When a linkset falls below the predefined threshold, a predefined link can be removed from its linkset and reassigned by link management to replace failed links in another linkset.

Automatic link allocation also enables voice circuits to be used as links. The requirement is that the facility must be digital and of the same data rate as the failed link. This is not a concern because most voice circuits in central offices today are DS0A circuits (interoffice trunks, not local trunks).

Link management is able to remove the voice circuit from its connection and reassign the voice circuit to a linkset. A predetermined SLS code is assigned to the new link. When the failed link returns to service, the voice circuit is removed from the linkset and returned to service as a voice circuit.

This feature requires the voice circuit to be connected to the same signaling point as the failed link. This could be the case between two signaling points, provided the voice circuit was terminated in the same signaling point as the failed link. This forces the assumption that the signaling points must be end offices, which provide both voice circuits and SS7 circuits (SSPs).

To accommodate link management and automatic allocation, a *signaling-data-link-connection* (SDLC) message (see Figure 6.6) must be generated and sent to the adjacent signaling point to inform the adjacent signaling point of the new assignment. The adjacent signaling point must make the same assignment for the link to work.

In the SDLC message, the signaling link number is the physical link number (such as link 1, 2, or 3), which in turn may be SLC 0, 6, or 12. When identifying the SLC, only the first code is provided. The other SLCs, even though they are assigned to the same link, are considered secondary and are not used in network management messages.

Once the SDLC message has been received, an acknowledgment must be sent to confirm that the adjacent signaling point has made the same assignment in its own database. This is accomplished using one of three messages: connection-successful, connection-not-successful, or connection-not-possible messages.

If the connection was successful, then traffic can begin on the newly allocated link. If connection was not successful (for any of the reasons provided), then the procedure is aborted.

In the unlikely event that two signaling points invoke automatic allocation simultaneously, the signaling point with the highest point code will be considered first, overriding the other signaling point.

To understand the impact that link management has on other portions of the signaling point, we must look at the other network management procedures. Keep in mind the layered approach discussed earlier in this section; it will help you to understand the individual roles played by all these functions.

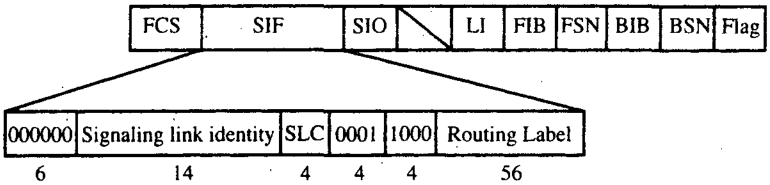


Figure 6.6 The SDLC message uses the network management structure and consists of the components shown here.

Traffic Management Procedures

Traffic management is used to divert traffic away from failed signaling links. It uses several messages that are sent using the MSU over adjacent links or adjacent linksets to adjacent signaling points.

Traffic management also deals with the source of congestion and provides flow control procedures. The objective of traffic management is to deal with the source of a problem whenever possible.

Traffic management also handles problems on a more direct level than other network management procedures. In essence, network management is offered at several layers, with traffic management in the middle layer.

Traffic management provides the mechanisms for managing traffic diversion due to the following:

- Signaling link unavailability
- Signaling link availability
- Signaling route unavailability
- Signaling route availability
- Signaling point restricted
- Signaling point availability

Signaling Link Unavailability. In the event that a signaling link should become unavailable or be manually blocked, deactivated by network link management, or inhibited, traffic management provides for the diversion of all traffic normally routed over the affected link to alternate links within the same linkset or other linksets that can route to the same destination.

Signaling Link Availability. In the event that a failed, blocked, inhibited, or deactivated link should become available again, traffic management diverts messages back to the affected signaling link. Procedures are provided to ensure that messages are not lost and the transmission is controlled to ensure an orderly delivery of all buffered messages.

Signaling Route Unavailability. In the event that an entire route should become unavailable, forced rerouting is used to divert the traffic away from the affected route. A route is a linkset or a group of linksets with a common destination.

Signaling Route Availability. Controlled rerouting is used to divert traffic back to a previously unavailable route. It involves an entire linkset rather than just an individual link. The diversion of traffic to another alternate linkset or route must be conducted in an orderly fashion to prevent messages from being lost.

Signaling Route Restricted. When a route becomes restricted, traffic must be diverted to a route of equal priority (or cost). In effect, this procedure invokes load sharing over two routes to prevent a route from becoming unavailable due to congestion (from multiple link failures, for example).

Signaling Point Availability. The MTP restart procedure is used to divert traffic to a signaling point now made available.

Traffic Management Messages. The traffic management message provides the SLC of the failed link and, in some cases, the *forward sequence number* (FSN) of the last good MSU received on the failed link. This information is sent to adjacent signaling points so they can assume the traffic for the failed link and ensure that no messages are lost. In order to ensure that no messages are lost, the traffic management procedure includes a method for copying all MSUs remaining in the transmission buffer of a failed link to the newly selected link. This procedure will be explained in further detail.

In all cases of traffic management, existing traffic on any one signaling link must not be interrupted. This means the procedures must permit normal traffic to continue while links are assuming the traffic of other failed links.

Changeover. The changeover message is used to divert traffic away from a failed link. LSSUs are sent by level 2 to indicate the status of the link throughout this procedure. This enables the two signaling points to maintain current status while the link is being realigned. When LSSUs are not being sent, *fill-in signal units* (FISUs) are transmitted.

Figure 6.7 shows the contents of the changeover message. The FSN is the FSN of the last MSU received by the failed link. This serves as an acknowledgment for any unacknowledged signal units received on that link. The following are the H1 heading codes used for changeover:

0 0 0 1	COO signal
0 0 1 0	Changeover acknowledgment signal
0 0 1 1	Extended COO signal (SAAL links)
0 1 0 0	Extended changeover acknowledgment signal (SAAL links)

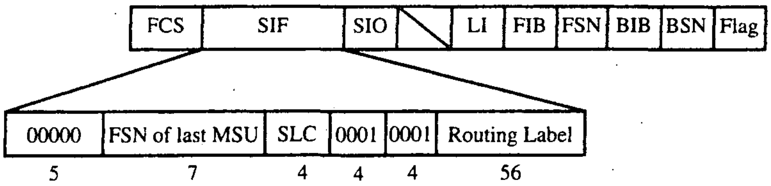


Figure 6.7 The COO consists of an SLC field, indicating the SLC of the failed link and the FSN of the last good MSU received on that link.

The SLC identifies the failed link. As we learned in our previous discussions about the SLC, this number is not necessarily a physical link number, but the logical link number assigned to the link.

It should also be mentioned here that all changeover-related messages (changeback, acknowledgments, and so on) use the same structure, using the same fields. These related messages are discussed separately for clarity.

When a link fails (see Figure 6.8), there are still MSUs in its transmit buffer that have not been acknowledged. Before these MSUs are discarded and the link realigned, they need to be dealt with in such a way that the messages will not be lost.

When level 2 detects a link failure, it first performs a buffer update. The messages in the transmit buffer of the failed link must be placed in the transmit buffer of the alternate link, so that they can be retransmitted in the event that an acknowledgment is not sent from the adjacent signaling point.

After the transmit buffer has been updated, the changeover message is sent over the alternate link to inform the adjacent signaling point that all messages to the affected node must be rerouted over the alternate link. The message will contain the SLC of the failed link, and the FSN (for MTP L2) or the sequence number of the last sequenced data protocol data unit (for SAAL) of the last good MSU received by the affected signaling point. The FSN in this message is used as an acknowledgment to the adjacent node so that it does not have to retransmit all MSUs in its transmit buffer.

The traffic is diverted to a link or links within the same linkset if there are any links available. However, there are times when there may not be any links available in the linkset. If there are no links available in the same linkset, another alternate linkset or linksets may be used. The destination must be the same for both linksets. In the event that there are no linksets available, routing management is triggered to reroute messages around the signaling point.

In the event that an STP does not normally carry traffic for the affected signaling point (the changeover was to a linkset to a different signaling point), a TFP message is sent by route management at the concerned signaling point. This is to prevent messages from being routed to the concerned signaling point and possibly causing congestion to occur (because the signaling point is now handling twice the traffic).

A TFP message informs adjacent nodes that no traffic addressed to the affected signaling point should be sent to the concerned signaling point. The concerned signaling point is that which originated the changeover message and is no longer carrying traffic to the affected destination.

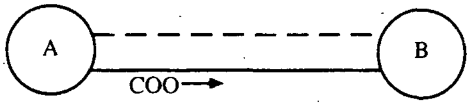


Figure 6.8 In this figure, the link indicated by dashes has failed. Signaling point A sends a COO to signaling point B, indicating the SLC code of the failed link.

The affected destination is that which has the failed link. The affected point code is usually provided in network management messages. Note that messages being carried over the concerned point code are being sent from the signaling point that used to carry traffic to the destination, but lost its path. The concerned signaling point has a path and is now "relaying" those messages, but should not be sent all traffic for this destination.

A good example of a situation like this is depicted in Figure 6.9. A signaling point that is adjacent to the affected signaling point has received a changeover, but does not normally carry traffic for this destination; therefore, the concerned signaling point sends the TFP message to any adjacent signaling points. When the concerned signaling point receives the changeover message, a *changeback acknowledgment* (CBA) is sent to indicate that the changeover was received and traffic is being diverted.

If there are no available links to the affected point code, then a time-controlled diversion procedure is implemented. This is also true if a processor outage is received, or if the signaling link is marked by the signaling point as inhibited, but is receiving traffic. The time-controlled diversion procedure prevents the signaling link from being failed and realignment from starting.

This is accomplished by setting level 3 timer T1, "Delay to avoid message missequencing on changeover." When timer T1 expires, new traffic can then be transmitted on the alternate link. This helps prevent the missequencing of messages, which may occur during a changeover initiated by the receipt of a processor outage or link-inhibited state.

Figure 6.10 shows when the use of time-controlled diversion may be necessary. Notice that signaling point A has lost its path to B, which is the STP for signaling point D. The changeover procedure is diverting traffic through signaling point C, which is adjacent to the affected signaling point D.

If the signaling link becomes available and the timer T1 has not yet expired, the time-controlled changeover is cancelled and traffic may resume on the affected link.

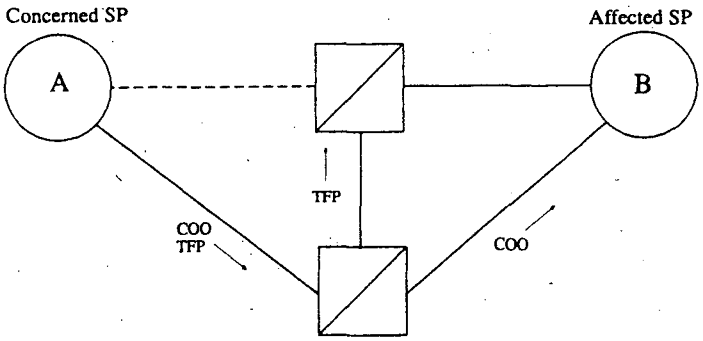


Figure 6.9 A COO may be sent through another signaling point, as depicted here. There is no other path for the message to get to signaling point B so it must be sent through an alternate route. The TFP is sent to all adjacent signaling points to prevent messages from being sent through A to B.

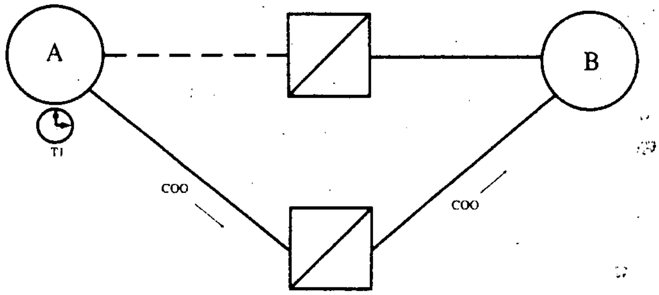


Figure 6.10 When timer T1 expires, the COO is sent to B. This is time-controlled diversion.

However, if the MTP becomes unavailable at the affected point code, the MTP restart procedure is initiated.

MTP restart is used to reset all timers and counters used by the MTP and to resynchronize the link. This means that all sequence numbering and error rate monitors are restarted as if for the first time. MTP restart is discussed in a later section.

If there is a processor outage at the affected link and the timer T1 expires, all messages that are available for retransmission are discarded. The processor outage means that level 4 and possibly level 3 are not functional and the messages could not be processed anyway. The affected link will not have any recollection of these MSUs ever being received because its receive buffer will have been reset.

If there is no acknowledgment received within timer T2 ("Waiting for changeover acknowledgment"), retransmission on the alternate link begins. This means that the changeover procedure starts without the acknowledgment. Any new traffic will transmit on the new link(s). This prevents a bottleneck in the event that the changeover message or the changeover acknowledgment was lost.

If a changeover acknowledgment is received, but there was no COO sent, the acknowledgment is ignored and discarded. No further action is necessary.

Changeback. The changeback message is used when a failed link has been restored, and traffic may now resume over that link. The message structure, as seen in Figure 6.11, consists of the SLC of the now-restored signaling link and the changeback code. The following H1 heading codes are used in changeback messages:

0 1 0 1	CBD signal
0 1 1 0	CBA signal

The changeback code is a unique pattern assigned by the originator of the changeback message. The changeback code enables a signaling point to initiate the changeback procedure for a number of signaling links. When the acknowledgment is returned, the acknowledgment must also carry the unique changeback code. This allows for the discrimination between acknowledgments and enables the signaling points to begin sending traffic in relation to each of the individual changebacks.

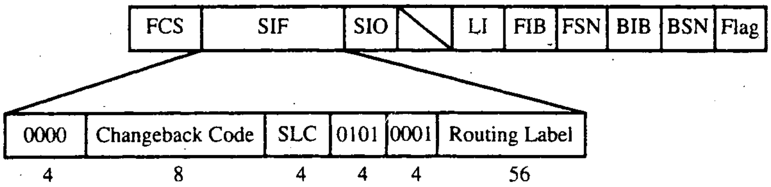


Figure 6.11 The *changeback declaration* (CBD) is sent to an adjacent signaling point to indicate that the link that previously failed has now returned to service.

Without this code, there would be no mechanism to allow for the orderly diversion of traffic over multiple signaling links. This is an issue only when there have been multiple link failures to one destination and all of the signaling links have been made available at the same time.

When the changeback is initiated, all transmission of MSUs on the alternate link is stopped. The changeback message is sent over the alternate link to the affected signaling point to inform it that the changeback procedure has been stopped and transmission over the failed link will now resume (see Figure 6.12).

The affected signaling point must then send a CBA (see Figure 6.13). This will trigger the procedure. All MSUs sent by level 4 or by other links destined to the affected signaling point are stored in a buffer until the changeback procedure is complete.

The CBA may be sent over any available link as long as the message is routed to the originating point of the changeback message. Once the acknowledgment has been received, the MSUs that were stored in the changeback buffer are sent over the now-available signaling link. There is no need to transfer unacknowledged MSUs from the alternate link transmit buffer to the now-available links transmission buffer.

In the event that the changeback originates from a signaling point that sent a TFP, a *transfer-allowed* (TFA) is sent to adjacent signaling points, enabling messages to be routed to the signaling point (see Figure 6.14). Likewise, if the affected signaling point became isolated due to the failed link, the signaling point is now made available by route management.

If an acknowledgment is not received within timer T4, "Waiting for changeback acknowledgment-first attempt," the CBD is repeated and timer T5, "Waiting for changeback acknowledgment-second attempt," is started. If timer T5 should expire before an acknowledgment is received, traffic is automatically started on the now-available signaling link. Maintenance functions within the signaling point are alerted in the event that there was an error in the acknowledgment transmission.

Emergency Changeover. In the event that a changeover procedure is initiated but the transmit buffer cannot be read, an emergency changeover procedure is used. The emergency changeover does not provide the FSN (MTP L2) or sequence number (SAAL) of the last good MSU received because the buffer has been cleared and that information is not available.

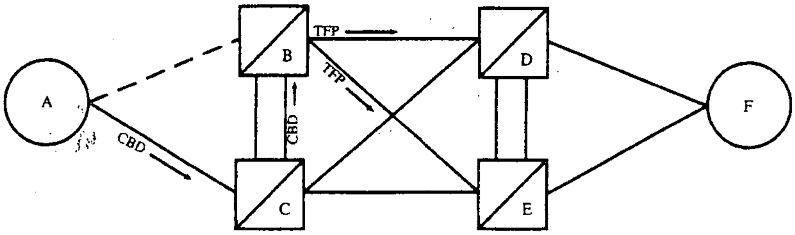


Figure 6.12 This figure shows the CBD being sent to signaling point B by using an alternate link to C. If there had been an alternate link directly to B, it would have been the path used for the CBD.

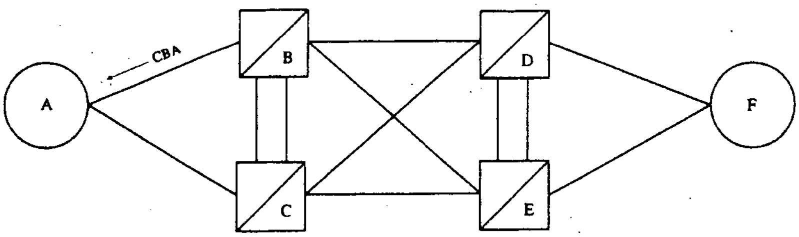


Figure 6.13 The CBA is sent over any available link. In this case, the previously failed link is used for the CBA.

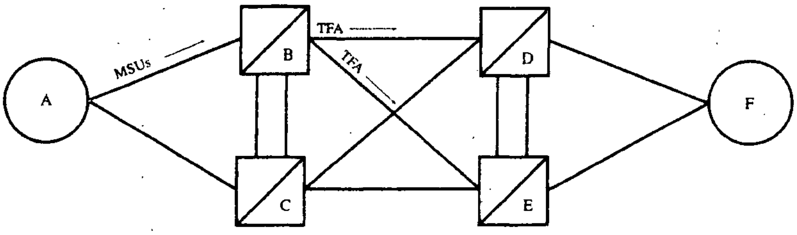


Figure 6.14 When MSUs are sent by the affected point code STP B in this example, the TFA is sent to all of B's adjacent signaling points.

The following H1 heading codes are used in emergency changeover messages:

0 0 0 1	Emergency COO signal
0 0 1 0	Emergency changeover acknowledgment signal

Level 2 begins sending the LSSUs on the failed link and FISUs when the LSSUs are not being sent. All new MSUs are diverted to the alternate link or linkset.

As was the case in the changeover procedure, the traffic can be diverted to multiple links or alternate linksets. Load sharing is invoked when there is existing traffic on these links to prevent a congestion condition on any one link.

In the event that there are no available paths to the affected signaling point on which changeover and emergency changeover messages may be transmitted, the time diversion procedure is invoked. As mentioned in the changeover procedure description, time-controlled diversion enables traffic to be diverted without failing the link (link status of out of service).

Forced Rerouting. The forced rerouting procedure is initiated in the event that a route to a specific destination becomes unavailable. The purpose is to reroute traffic around the concerned signaling point to the destination without losing messages or causing any other route to become congested.

Figure 6.15 shows MSUs destined for signaling point A. Signaling point A becomes unavailable through the route using signaling point C. Signaling point C sends a TFP message to signaling points D and E to inform them that the route to signaling point A is inaccessible, and messages should be rerouted through an alternate route.

In Figure 6.16, the alternate route is through signaling point B. All traffic to signaling point C (destined to signaling point A) is stopped. The forced rerouting buffers store all MSUs with the destination of A. When the alternate route is determined (B in this example), all diverted traffic is transmitted to signaling point B. The contents of the forced rerouting buffer are sent first.

When the route to signaling point A through signaling point C becomes available again, signaling point C sends a TFA to signaling points D and E (see Figure 6.17). Messages destined to signaling point A can now be rerouted back through signaling point C, or signaling point B, depending on the network configuration.

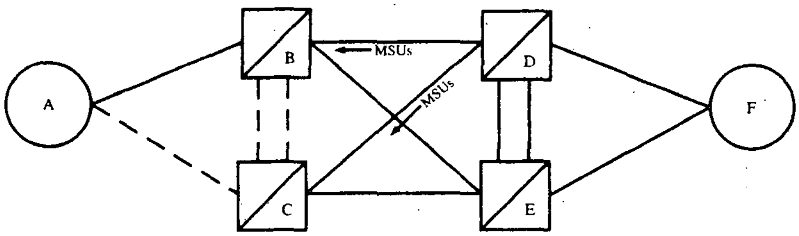


Figure 6.15 In this figure, MSUs are destined to A through C. The link from C to A has failed. The links from C to B have failed as well, causing messages to be sent back to D and E (circular routing).

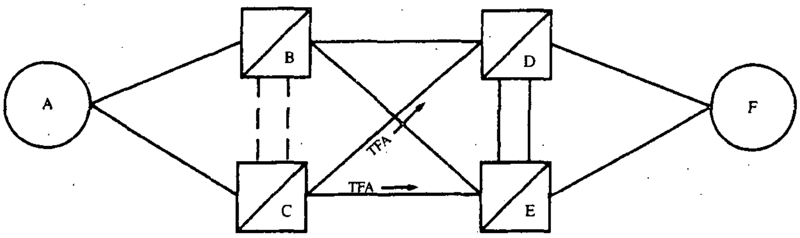


Figure 6.16 The link from A to C has been restored. C sends the TFA to its adjacent nodes, enabling traffic to A.

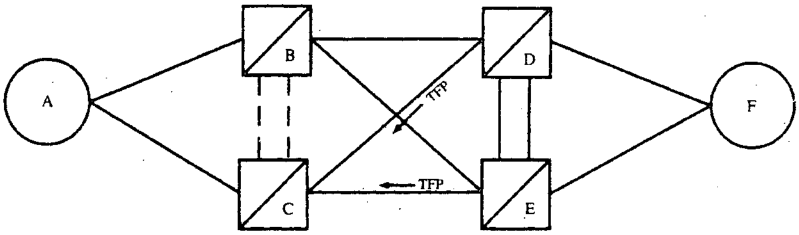


Figure 6.17 To prevent circular routing, D and E send TFPs to C. This prevents C from sending MSUs destined for A or B back to D and E.

Any existing traffic on the link should not be interrupted in any way. If there is a lot of traffic on the alternate route, load sharing is used to spread the traffic over the links evenly. This procedure should not cause the alternate route to become inaccessible due to failure or congestion.

If there are no alternate routes available, traffic is blocked, and messages stored in the forced reroute buffer are discarded. Flow control is used to inform user parts to stop sending traffic to the affected point code. A TFP message is also sent to the adjacent signaling points to stop traffic from being sent to the concerned signaling point (see Figure 6.18).

Controlled Rerouting. The objective of the controlled rerouting procedure is to restore traffic to the most favorable route in cases where a particular route was previously restricted. This is probably best explained by Figure 6.19. In the figure, traffic is destined to signaling point A. The primary routes to signaling point A from signaling points D and E use signaling point C, with signaling point B used as an alternate.

The route from signaling point C to signaling point A fails. In addition to this route failing, the route from signaling point C to signaling point B fails (see Figure 6.20). This scenario in the previous example (forced rerouting) caused a TFP message to be sent from signaling point C to signaling points D and E. This forces signaling points D and E to route all traffic destined for signaling point A through signaling point B.

The route from signaling point C then becomes available. All messages destined for signaling point A can now be routed through the primary route, signaling point C. To initiate this, a TFA message is sent to signaling points D and E (see Figure 6.21).

Signaling points D and E then initiate the controlled rerouting procedures. To prevent circular routing, signaling points D and E send a TFP to signaling point B (see Figure 6.22). This prevents signaling point B from sending messages destined for signaling point A.

A timer T6, "Delay to avoid message missequencing on controlled rerouting," is set and, after its expiration, the controlled rerouting buffers at signaling points D and E are transmitted to signaling point C (see Figure 6.23).

MTP Restart. The MTP restart procedure (see Figure 6.24) is an addition to the Telcordia standards, even though it has been in the ANSI publications for a while. The

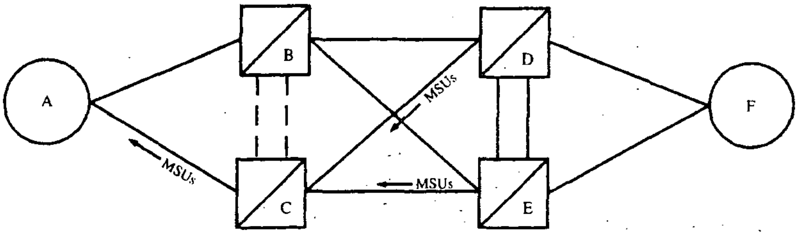


Figure 6.18 Once the TFP has been sent, MSUs in the buffers of D and E can be sent to A through C.

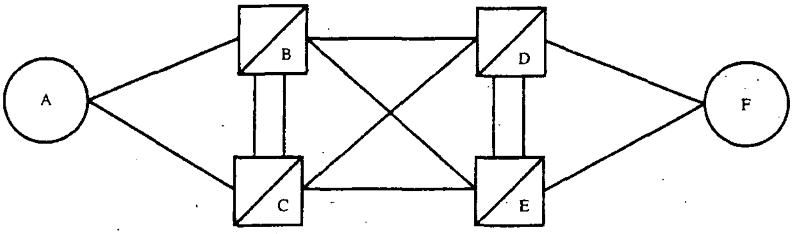


Figure 6.19 The links from C to A are primary linksets for routing and signaling point A.

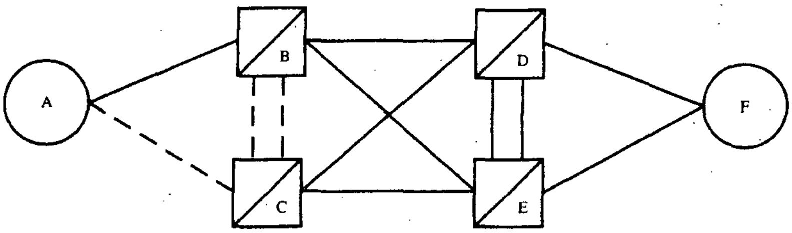


Figure 6.20 Links from C to A and C to B have failed.

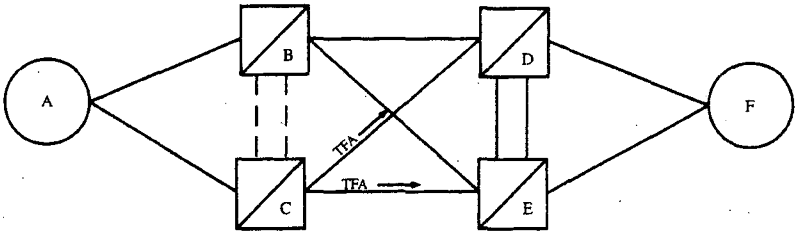


Figure 6.21 The link from C to A has now been restored. The TFA is sent to D and E to enable traffic for A to be sent to C.

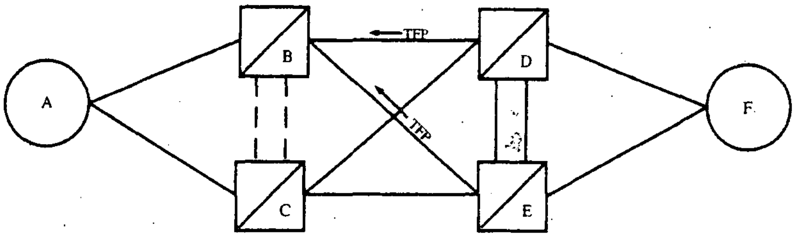


Figure 6.22 Signaling points D and E send TFPs to signaling point B to prevent B from sending traffic destined to A back to D and E (circular routing).

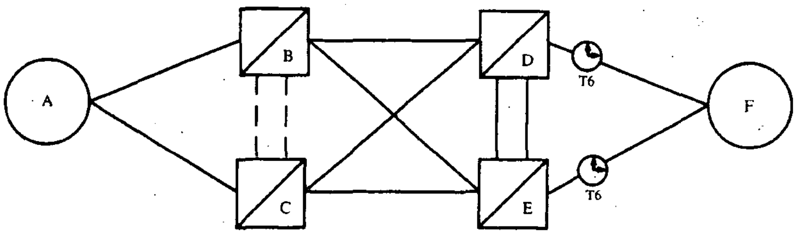


Figure 6.23 After T6 expires, MSUs resume on the links to signaling point C.

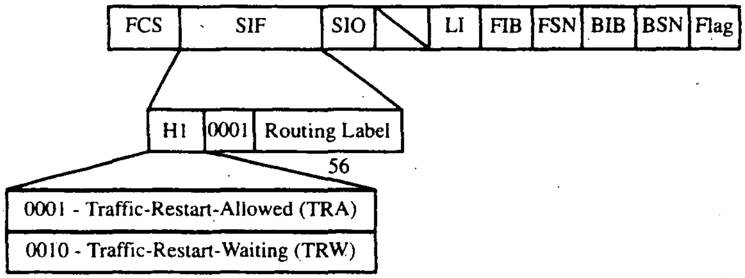

FCS
SIF
SIO
LI
FIB
FSN
BIB
BSN
Flag
H1
0001
Routing Label
56
0001 - Traffic-Restart-Allowed (TRA)
0010 - Traffic-Restart-Waiting (TRW)

Figure 6.24 The *traffic-restart-allowed* (TRA) and the *traffic-restart-waiting* (TRW) messages are used with the MTP traffic restart procedures. There is no information provided other than the management message.

purpose of this procedure is to protect the network and a signaling point in the event that a signaling point becomes isolated for a period of time and then becomes available.

When this occurs, the signaling point must exchange a great deal of routing status with its adjacent signaling points. The failed signaling point, because of the length of time it was out of service, may not have current routing status. Events involving adjacent signaling points may have occurred while it was isolated and, because no messages were routed to the signaling point, it cannot be aware of congested signaling points or other inaccessible signaling points.

To ensure that the signaling point has ample time to update its routing status information and exchange information with all of its adjacent signaling points, the MTP restart procedure is implemented with a set of timers. The timers ensure that the signaling point has ample time to retrieve routing status on each linkset or route.

There really are no set rules for when an MTP restart procedure should begin. One suggestion by Telcordia is to start the procedure upon receipt of an LSSU with a value of processor outage (SIPO).

There are two levels of MTP restart: a full restart and a partial restart. It is up to the management functions at level 3 to determine which is implemented and under what conditions. This is network dependent and can be implemented in a number of conditions.

One of the suggestions by Telcordia for initiating this procedure is that, when a route becomes available after a remote processor outage, the route initiates a restart procedure. This procedure at the remote end of the route involves the use of timers (T25 and T28), which are initiated upon receipt of the first link status of in-service (determined by level 2).

When a partial restart is initiated at the isolated signaling point, it sends out a TRA message to its adjacent signaling points. The remote signaling points can then initiate their own restart procedures on the direct routes to this signaling point, as specified in the ANSI and Telcordia standards. This procedure involves the use of timers T25 and T28 (as previously mentioned).

If a full restart is required, then timer T27, "Minimum duration of unavailability for full restart," is started. This timer is used to ensure that all adjacent signaling points see the unavailability of the routes leading to this signaling point and have ample time to respond. During the duration of T27, the isolated signaling point sends level 2 processor outage messages to its adjacent signaling points using the LSSU.

After the expiration of timer T27, the signaling point then attempts to place predetermined links back into service. The primary link of each linkset should be the first to be restored. To expedite the alignment procedure on these links, the emergency alignment procedure is suggested for the first links within the linksets.

Once these links have been brought back into service, route management messages can be exchanged. The route management messages will enable the isolated signaling point to determine the current status of all direct routes. While the in-service links are exchanging route status information, the other links can begin their alignment procedures.

When the first link in a linkset goes into service, timer T22, "Timer at restarting signaling point waiting for signaling links to become available," and timer T26, "Timer at restarting signaling point waiting to repeat traffic restart waiting message," are started. Timer T22 is stopped when sufficient links have become available ("sufficient links" is a network-dependent parameter). If T26 expires, a TRW message is sent to all adjacent signaling points. Timer T26 is then restarted. This timer (T26) ensures that enough time is allowed for the procedure to be completed. Timer T26 is stopped upon the expiration of several other timers, depending on the events taking place.

There are many other timers and events that take place during the MTP restart procedure. The intent here is to give you an idea of what this procedure tries to accomplish.

Without this procedure, each link is left to its own accord, and the alignment procedure is begun on a link-by-link basis.

When the links are left to their own devices to get restored, there is no orderly fashion in which routes are reinstated. This procedure brings more order to the alignment procedure and the procedures that take place when an entire signaling point has been isolated.

Management Inhibiting. Link management to block a signaling link from level 4 uses management inhibiting. The status of the link does not change at level 2. The purpose of this procedure is to enable personnel to send test messages over the inhibited link or to enable link management to send test messages over the link without interference from any of the user parts.

However, inhibiting a link is not permitted if the signaling point is under a congestion status or if there are no other links available. If the link being inhibited is the last available link, the procedure is denied. If a signaling point should suddenly become isolated (due to other link failures) or if all other links within the same linkset as the inhibited link should become unavailable, inhibiting is canceled, and the link is returned to normal service.

If no other links fail and the inhibit procedure is uninterrupted, only the originator can uninhibit the link. The link inhibition can be initiated either through a command entered at a system terminal or by network management.

The link is inhibited by sending an inhibit request to the remote signaling point. This informs the remote signaling point that the originator wants to inhibit the link and the remote signaling point should mark the link as inhibited. If, for any of the reasons mentioned here, the link cannot be inhibited, the request is denied.

To ensure that the link is marked as inhibited at both ends of the link, both signaling points periodically send test messages to check the status of the link at the adjacent signaling point. This is accomplished through the inhibit test message.

During the time the link is marked as inhibited, the local signaling point sends a local inhibit test message. This is to ensure that the link status in the remote signaling point is still shown as inhibited. If the remote does not acknowledge the local inhibit test message, the procedure begins again with an inhibit request. The local signaling point must first force an uninhibit of the link before inhibiting the link can start.

Likewise, the remote signaling point will also periodically send a remote inhibit test message. If the remote inhibit test message does not get an acknowledgment, the link status at the remote signaling point is changed to available (through a forced uninhibit), and traffic is enabled on the link. These two test messages ensure that both ends of the link are aligned properly and show the same status. A signaling point is allowed two attempts at inhibiting a link.

When the inhibit link message is received by a signaling point, the receiving signaling point initiates a changeover, diverting traffic away from the link onto other links within the same linkset or other linksets. If the signaling point receives an inhibit message, the link is marked as remotely inhibited.

The originator of the inhibit marks the link as locally inhibited. A changeover procedure is initiated to divert traffic to other links. The local signaling point uses the time diversion changeover procedure.

If the local signaling point has not received an acknowledgment of its inhibit message within timer T14, "Waiting for inhibit acknowledgment," the procedure is started again. Two consecutive attempts are enabled. If the signaling point is still unsuccessful after two attempts, the inhibit procedure is aborted, and the link remains available for traffic.

Inhibiting a link is usually a manual procedure, used by maintenance personnel for testing the reliability of a link. Test messages with specific patterns can then be exchanged over the inhibited link and checked for accuracy. The signaling point has the responsibility of ensuring that the inhibited link does not remain inhibited after testing is complete (caused by lost uninhibit messages never received by remote signaling points) and that the inhibited link is not the last available link to another signaling point.

Flow Control. Flow control at level 3 is used to control the flow of user part messages from the source. Much about flow control is implementation dependent. The procedures implemented for congestion or unavailable user parts are dependent on the manufacturers of SS7 equipment. The standards only define the need for such procedures and make suggestions as to how they can be addressed.

The intent of these procedures is to deal with congestion at the source, where the messages are being generated. This, of course, is at level 4—user parts. The protocol has no interaction, really, at this level because these are internal functions. However, the protocol does trigger these internal functions.

If a TFP message is received for a particular destination, level 3 will interact through network management with the protocol and will direct level 4 as well. Communications from the protocol to these internal functions are accomplished through the use of primitives, which were discussed in Chapter 5.

The primitives offer a structured communications format to interact with other levels. In the case of flow control, level 3 must be able to notify level 4 of a congestion condition at another signaling point. The result is a reduction in traffic being generated for the affected signaling point.

The advantage to this is twofold. Only the user part experiencing the congestion is affected, rather than the entire signaling point. The congestion flow control is directed at a specific user part (such as ISUP) rather than an entire signaling point. This enables other traffic to continue without impedance.

The other advantage is that congestion is dealt with at the source, rather than by trying to redirect messages around the congestion. If a particular node is causing congestion, the amount of traffic generated by that node is throttled, instead of redirecting all that traffic to another destination.

There are procedures invoked by route management at level 3 that also deal with congestion conditions; however, they deal with signaling point congestion. Route management does not communicate with the source, the user parts, directly. Traffic management deals with the source.

The MTP uses traffic management flow control to deal with traffic destined to a user part that has become unavailable. This is from the perspective of the receiver, rather than that of the source.

If an MSU is received by the MTP and the message discrimination function has determined that the MSU is addressed to the local signaling point, the message is sent to message distribution to be given to level 4. However, if message distribution is unable to give the message to level 4 because of a congestion condition or because the user part at level 4 is not available, traffic management MTP flow control is invoked to inform the originator of the problem (see Figure 6.25).

An example of how this could occur is best explained using the feature of global title translation. An STP performs GTT when it receives an SCCP message with a called party address that contains only digits.

These digits are referred to as global title digits. The digits are typically nonroutable (that is, 800 number or 900 number). The SCCP must deliver the message to a user part (SCCP in this case) at the receiving signaling point for the GTT function. If the processor dedicated to that function has failed, there are no resources available to perform the global title translation.

The signaling point in this case would return a UPU message to the originator of the message indicating that the message processing could not be completed because the necessary resources are not available to perform the function.

In Figure 6.25, the UPU message provides the DPC (the point code of the failed user part), the user part that is not available (SCCP, ISUP, and so on), and the cause. The MTP user code is the same code used in the SIO of the MSU.

There are only three causes today: reason unknown, an unequipped remote user part, or the remote user part is inaccessible. The condition for each of these causes is purely implementation dependent and may have different meanings in different networks.

In the event that traffic management should create a routing problem to any other signaling point, the routing management function at adjacent signaling points will be invoked. Likewise, if a signaling point that has invoked traffic management has lost a route to a destination, it will also invoke routing management to resolve routing issues. Route management is described in the following section.

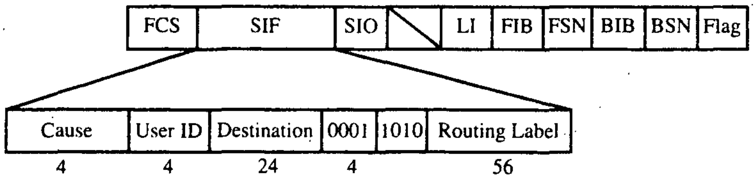


Figure 6.25 The UPU indicates that a user part is not accessible at the specified destination. The user ID is the same value as that provided in the SIO field.

Routing Management Procedures

Routing management is used by a signaling point to notify its adjacent signaling points of a routing problem. The routing problem is usually attributed to the loss of a signaling link or linksets, which together comprise a route.

The purpose of routing management is to redirect traffic around the failed route. This is accomplished through the use of transfer messages, which identify the failed destination by point code and instruct receiving signaling points on how to react (see Figure 6.26).

When a network is using cluster addressing, cluster routing management can also be invoked. In cluster addressing, each STP is assigned a unique cluster address. All of the signaling points that "home" to that STP are assigned the same cluster address, with unique member addresses (see Figure 6.27).

This enables routing management cluster messages to be sent to an STP and distributed among all of its cluster members. The advantage of cluster routing management is that one message can be sent to address an entire group of signaling points, rather than individual signaling points.

In addition to cluster routing, all networks must use some prioritizing on their routes. This is typically done by weighting each route in order of efficiency (see Figure 6.28). For example, all signaling points will have a primary route. The primary route (one for every destination) is the fastest path to the destination. Therefore, it is considered the most efficient.

In addition to the primary route, there should also be a secondary route, which is not the most favorable path but provides an alternative in the event that the primary path should fail. This is not the best path, but it will get messages to the same destination.

Additional routes are weighted in a similar fashion. The more paths provided, the more reliable the network. The objective is to provide diverse routes to the same destination so that, in the event that a major failure should occur, messages can still be routed to their destinations.

When a route fails, a routing management message is sent to adjacent signaling points to advise them that the originating signaling point can no longer reach a destination through its routes, and an alternative should be selected ("Do not send to me

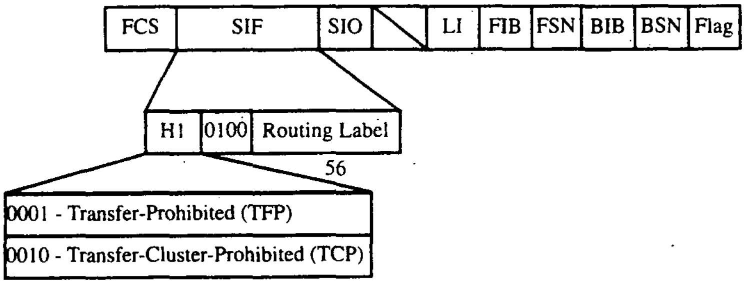

FCS
SIF
SIO
LI
FIB
FSN
BIB
BSN
Flag
H1
0100
Routing Label
56
0001 - Transfer-Prohibited (TFP)
0010 - Transfer-Cluster-Prohibited (TCP)

Figure 6.26 Both the *traffic-prohibited* (TFP) and the *transfer-cluster-prohibited* (TCP) work in much the same way. The TDP identifies a single entity, while the TCP identifies a cluster of entities.

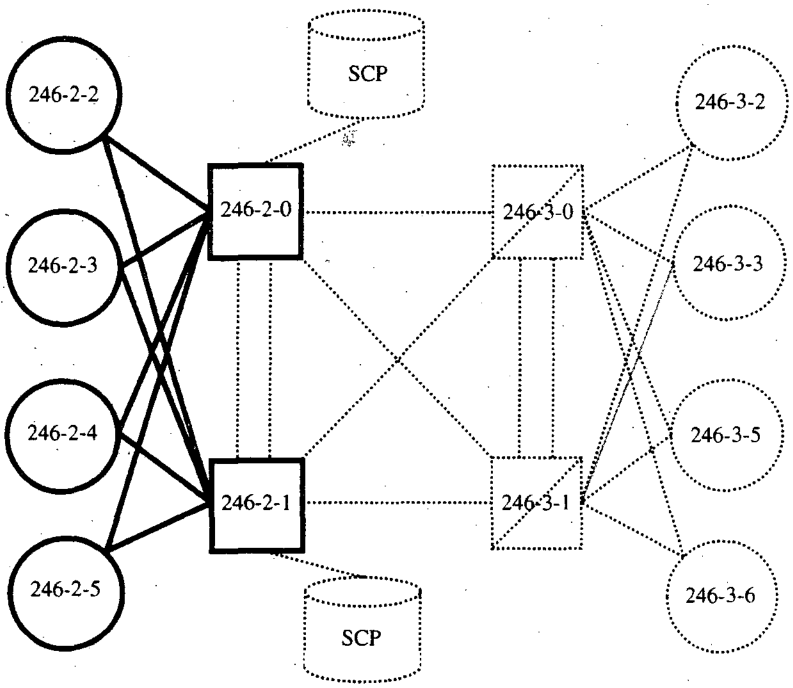


Figure 6.27 In this figure, the group of SSPs that share the same home STP have the same cluster address. Network management messages can then be sent from the home STP concerning the status of the entire cluster. Signaling traffic can also be routed by the cluster address rather than by the entire point code.

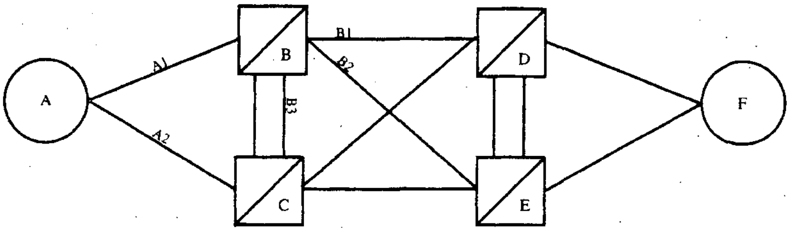


Figure 6.28 In this example, A1 is a primary route to destination F, while A2 is the alternate route. For signaling point B, B1 is the primary route to F, and B2 is the secondary route, while B3 is an alternate route.

because I can't get there"). For normal routing management, the following messages are used:

- TFP
- TFA

- *Transfer-restricted* (TFR)
- *Transfer-controlled* (TFC)
- *Signaling-route-set-test* (RST)
- *Signaling-route-set-congestion-test* (RCT)

During cluster routing management, the following messages are used:

- TCP
- *Transfer-cluster-allowed* (TCA)
- *Transfer-cluster-restricted* (TCR)
- *Cluster-route-set-test* (CRST)

These messages and their accompanying procedures are explained as follows:

TFP. The TFP message (H1 heading code 0001) is sent by a signaling point when it determines that it can no longer reach a destination (adjacent signaling point). The reason for the isolation is due to the loss of an entire route, which connects directly to the affected destination.

The message will provide the DPC for the affected signaling point. This enables adjacent signaling points to determine which alternate route to select to reach the affected destination.

In Figure 6.29, signaling point D can no longer route messages to signaling point F. A TFP is sent to signaling points B and C to advise them to select an alternate route. When signaling points B and C receive the TFP message, they stop transmission of all MSUs to the concerned signaling point (D) with the address of the affected signaling

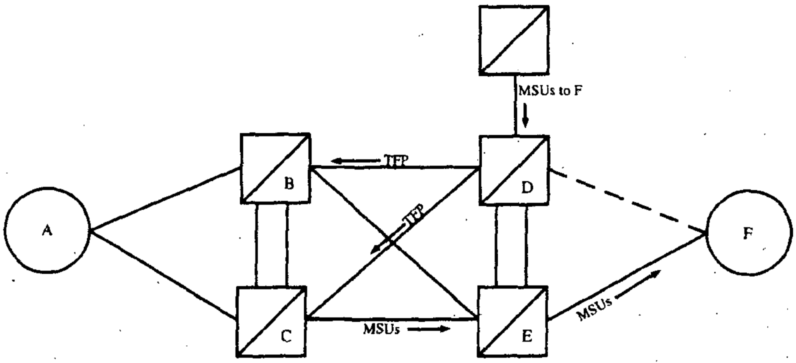


Figure 6.29 In this figure, D can no longer send MSUs to F. A TFP is sent to B and C to prevent them from sending traffic for F to D. If D gets any traffic from other signaling points, the traffic is sent to C for routing to F.

point (F) until an alternate route is determined. Once the alternate route has been determined, MSUs are transmitted via the alternate route to the affected destination (F). Any traffic generated at signaling point D destined for signaling point F is sent to signaling point B or C for routing to signaling point F.

This is a very simple example. Depending on the network configuration, this procedure can involve very little rerouting or a lot of rerouting. The objective in any routing plan is to keep all routing as direct and as simple as possible.

When the failed route becomes available again, a TFA message is sent to all adjacent signaling points by signaling point D, and normal routing is resumed.

TCP. Like the TFP message, the TCP message (H1 heading code 0010) is sent by a signaling point to a cluster of signaling points. Each STP is assigned a unique cluster address, while all signaling points that home to the STP have the same cluster address as the STP but use unique member addresses.

As seen in Figure 6.30, the home STP sends a TCP message concerning all of the signaling points that are homed to the STP. This enables one message to be sent regarding the entire cluster, rather than numerous messages sent by each signaling point in the cluster.

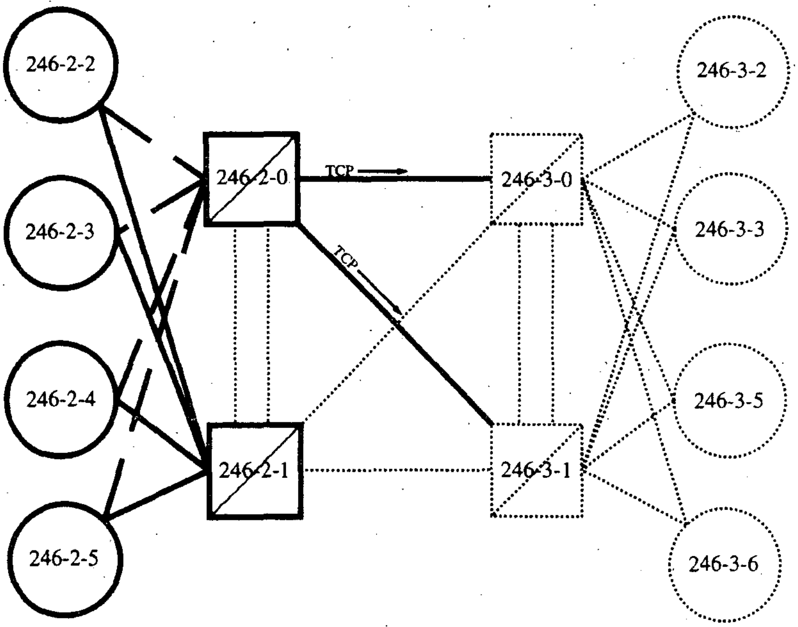


Figure 6.30 In this example, the STP 246-2-0 has failed and cannot reach any of the signaling points that home to it. A TCP message is sent to indicate this condition and cause routing to be forced to STP 246-2-1.

TFA. The TFA message (H1 heading code 0101) is used when a route becomes available again. This is sent by the originator of a TFP message to indicate that traffic may be sent once again to the affected signaling point.

The message structure is the same as the TFP. When received, the TFA should trigger a changeback to occur on the concerned link.

TCA. The TCA message (H1 heading code 0110) is used to indicate the route to the specified cluster is now available. This is sent by the originator of a TCP message to indicate that traffic may now be sent to the affected cluster point code.

TFR. The TFR message (H1 heading code 0011) is sent by an STP when it is determined that messages to a particular destination should no longer be sent to the STP for routing to the affected signaling point (see Figure 6.31). The TFR is always sent to adjacent signaling points.

In the event that a signaling link to the affected destination experiences a long-term failure (such as a processor outage), the STP receives a changeover from the affected signaling point, ordering all traffic to be diverted away from the link at fault.

The STP may then determine that it is necessary to send the TFR to all of its adjacent signaling points to prevent traffic from being addressed to the STP for routing to the affected signaling point.

The restriction does not prevent messages from being transferred from the STP entirely, however. The restriction prevents normal traffic flow and forces other signaling points to find an alternate route for traffic destined to the affected signaling point. If there are no alternate routes available, then the traffic can still be routed through the STP normally.

In this previous case, the TFR is sent using the broadcast method. The broadcast method automatically transmits the TFR when the link has been determined to have failed.

When a significant number of links in a linkset fail, a TFR is sent to adjacent signaling points using the response method. The response method sends the TFR when an MSU is received on a link for transfer to the affected signaling point.

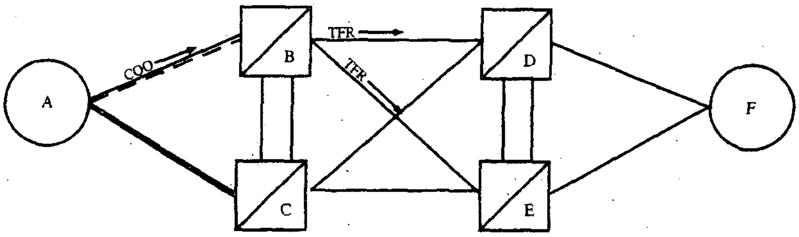


Figure 6.31 The TFR message is used to restrict traffic flow. In this example, a link from A to B has failed, leaving only one link available. A changeover has occurred, and a TFR has been sent by the STP to all of its adjacent signaling points.

The criterion for sending a TFR message under these conditions is implementation specific. The objective is to restrict traffic to the STP on the linkset that has experienced link failures before congestion occurs. Congestion can occur at both the link level and the user part level.

If a signaling point was previously prohibited and links become available, the signaling point would be considered restricted and TFR messages would be sent to all adjacent signaling points until the signaling point was 100 percent in service again. This means that all signaling links to the affected destination would have to be in service.

There are cases when traffic on an alternate route would be diverted via the controlled rerouting procedure to the restricted route. This should occur only when the restricted route has a higher priority than the alternate route. The priority of a route is set by an administration command at each signaling point.

In the event that both routes should be of equal priority and both are restricted, then load sharing should be implemented to distribute traffic evenly over both routes. This ensures that links are carrying an even amount of traffic, and one link does not become burdened with all the traffic from the failed route.

TCR. When several routes within a cluster fail or become congested, the TCR message (H1 heading code 0100) is used (see Figure 6.32). This message indicates to an adjacent STP that the concerned cluster should not be routed any messages if possible.

In Figure 6.33, the signaling points attached to STP A have the same cluster address as STP A. This means that STP A is their home STP. In the event that one or more routes within this grouping of signaling points should become unavailable, STP A would send a TCP to any adjacent STPs. This enables control over the traffic to the signaling points attached to STP A and any of its concerned signaling points. The same rules apply to the TCR as the TFR.

It should be noted that not all networks use cluster routing. Cluster routing is a network management feature and is not implemented for normal message routing. However, partial point code routing can be used for normal routing of all messages through the network. If partial point code routing is offered within a network, then, most likely, cluster routing is also implemented because the two are closely related.

SRST. This message is used to test the status of any prohibited or restricted route. When a TFP or TFR is received by a signaling point from an adjacent signaling point,

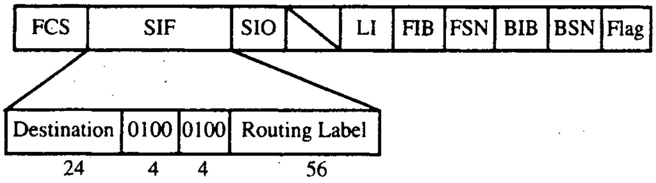


Figure 6.32 This is the message structure of a TCR message.

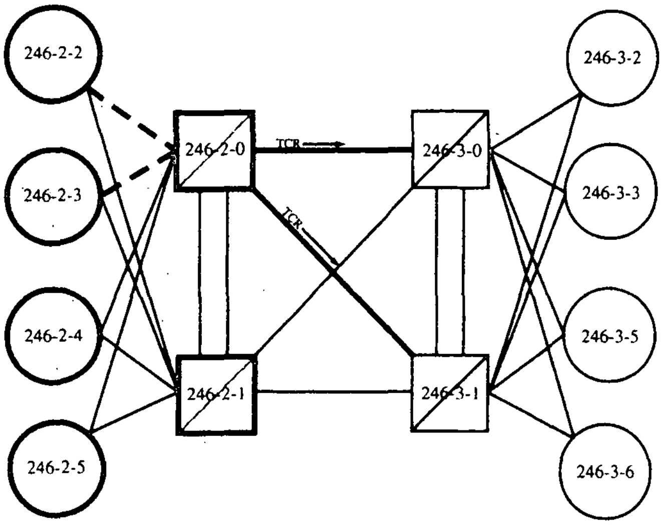


Figure 6.33 Two signaling links to STP 246-2-0 have failed. The failure of these signaling links triggered a TCR to be sent to the adjacent STPs.

a timer T10, "Waiting to repeat signaling-route-set-test message," is automatically activated (see Figure 6.34). The following H1 heading codes are used in this message:

0 0 0 1	SRST signal for prohibited destination
0 0 1 0	SRST signal for restricted destination
0 0 1 1	SRST signal for prohibited cluster
0 1 0 0	SRST signal for restricted cluster

At the expiration of timer T10, the SRST message is sent to the originator of the TFP or TFR (this also applies to TCP and TCR messages). When the SRST is sent, the timer T10 is reset.

The SRST message is retransmitted after every expiration of timer T10, until a TFA has been received by the testing signaling point. This procedure is used to ensure that a prohibited or restricted signaling point does not get stuck in that condition indefinitely. The message contains the status information (from the perspective of the originator) for the concerned signaling point as well as the heading code. No other information is necessary.

Another use for this message is when a link becomes available, but traffic is not restarting. Notification is given by level 2 (LSSU) that the link has started the proving period. The adjacent signaling point receives this status and also begins the proving period. After the proving period has ended and no other indications of failure are received, the link should be considered available.

However, only the signaling point that initiated the proving period can restart traffic on the link. The adjacent signaling point typically waits until receipt of an MSU. The adjacent signaling point should then send the SRST message to determine if the route

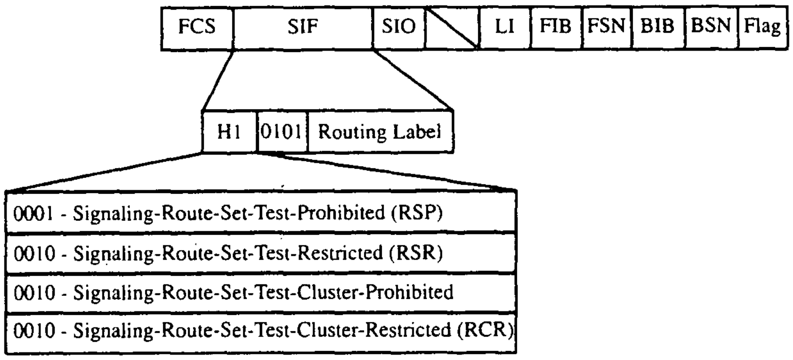

FCS
SIF
SIO
LI
FIB
FSN
BIB
BSN
Flag
H1
0101
Routing Label
0001 - Signaling-Route-Set-Test-Prohibited (RSP)
0010 - Signaling-Route-Set-Test-Restricted (RSR)
0010 - Signaling-Route-Set-Test-Cluster-Prohibited
0010 - Signaling-Route-Set-Test-Cluster-Restricted (RCR)

Figure 6.34 The message structure of SRST messages is shown in this figure.

has become available. The message would then be transmitted every T10 until either an MSU is received or the status is indicated by level 2.

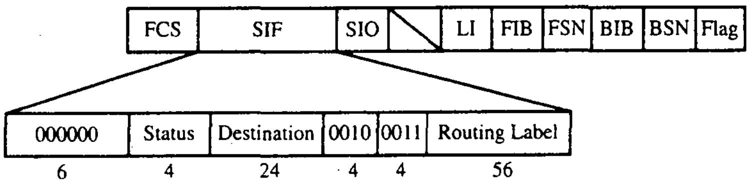
In the event that an SRST message is sent to a previously restricted cluster, the receiving STP (considered the home STP for all other signaling points in that cluster) will compare the actual status of the cluster with that indicated in the message. If the status of the cluster is not prohibited or restricted, the home STP will send a TCA back to the originator of the SRST message. The originator of the SRST then updates its status indicator for the cluster as available and allows traffic to be routed to the concerned STP. This procedure prevents a cluster from erroneously being marked by a signaling point as unavailable or restricted when a TFA or TCA message is lost.

If the cluster is not available and there are any signaling points within the cluster that are in danger of becoming congested, then the TCR message is sent to the originator of the SRST message. This prevents further congestion from forcing the signaling point into a busy condition.

TFC. The TFC message (see Figure 6.35) is sent by an STP when it receives an MSU destined for a route that has been marked by the STP as congested. The TFC is addressed back to the originator of the MSU.

In Figure 6.36, a signaling point has sent an MSU to an STP to be routed over any available route to the destination addressed in the routing label. The STP determines that the route for that destination is congested, and there are no other routes to the destination.

When the MSU is received on a signaling link, level 3 routing must determine which route to send the message out on. If the only route is congested, level 3 network management will discard the message and send the TFC back to the originator of the MSU. The TFC should be generated by the link on which the message came in, rather than using the same processor showing the congestion. In other words, if resources are showing a congestion status, sending them additional work compounds the problem. Implementation of this procedure will differ from system to system, but the overall objective is the same. Reduce the traffic to the congested route.



FCS	SIF	SIO		LI	FIB	FSN	BIB	BSN	Flag

000000	Status	Destination	0010	0011	Routing Label
6	4	24	4	4	56

Figure 6.35 The structure of the TFC message.

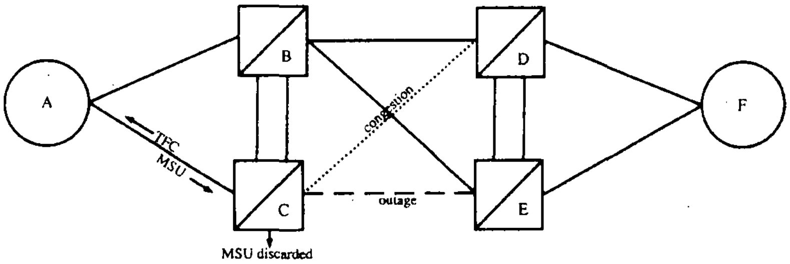


Figure 6.36 The link between C and E has failed in this example. As a result, the link from C to D has become congested. MSUs sent by A to C are discarded, and a TFC is sent by C to A.

The reduction is accomplished by returning a congestion status code in the TFC message. The congestion status indicates the priority level a message must possess before it will be routed over the congested route. Only messages of a higher priority than indicated in the congestion status field will be enabled on the congested route.

The priority is determined by the individual signaling point. Each signaling point must assign a congestion status level to a route. The status is a two-bit code, providing for a status level of 1, 2, or 3 (0 indicates no congestion). When an MSU is received by an STP and is routed to the congested route, the network indicator field of the SIO is examined to determine what priority has been assigned to the received message.

Priorities are assigned by the originating signaling point. This coding is implementation dependent. It can be an administrable value or automatically assigned based on configuration of the signaling point. Regardless, in ANSI networks, this field determines whether or not a message will be enabled to pass through a congested route, based on the congestion level of the route.

If we examine the network indicator field, there are two spare bits associated with the national network indicator. They can have values of 0, 1, 2, or 3 (which correspond with the congestion-level values). The MSU must have a priority value equal to or greater than the current congestion level of the congested route.

In addition to indicating the affected DPC (the affected point code is the one that is adjacent to the STP and the congested route) is the current congestion status of the route. This enables the signaling point that originated traffic towards the destination to determine which type of messages it can send. If it has any messages to transmit that have a priority less than the congestion status indicated in the TFC, then the messages are not sent. This prevents the STP from receiving unnecessary messages that will have to be discarded.

Messages received by the STP for a congested route are not processed and are not returned. They are simply discarded, and the TFC message is sent in the backward direction to indicate that they have been discarded. The only time the TFC is sent is when a message has been discarded because of a congested route. If the MSU has a priority equal to or higher than the congestion level, the message is enabled to pass through the route to its destination, and no TFC message is created.

It should also be noted that if a signaling point received a TFC message, and it determined that another route is available to the same destination, then it can choose another route to the destination. In this case, the signaling point would mark the route on which it received the TFC message as congested and would invoke routing management procedures.

The TFC message is sent on a regular basis to update the originating signaling point of the congestion status. If timer T15 expires within the originator of an MSU and no TFCs have been received within timer T15, the RCT procedure is invoked to determine if the route is still congested.

The RCT procedure is explained in full detail in the next section. The concept is to send a message (the test message) with a priority value one less than what the originator thinks the congestion status is. If the route is still congested, the receiving STP will send a new TFC message indicating the congestion level.

If the RCT message gets through and a timer T16 expires, then the route is considered congested but at a lower congestion level than the test message. The RCT message is sent again with a lower priority, and the procedure is repeated until the route is found to be at level 0 (no congestion).

RCT. In the event that a signaling point receives the TFC message, it needs to periodically verify that the indicated route is still under TFC procedures. There are no indications sent by the originator of the TFC message that the route is no longer under TFC procedures.

If we look at the big picture of what is happening during this procedure, we can see multiple tiers of network management. Procedures have been invoked at the link level, which only involve two signaling points directly adjacent to one another.

Traffic management is diverting traffic away from the affected signaling link, but, again, only between two adjacent signaling points. This sometimes may trigger the routing management procedure, which involves adjacent signaling points of an STP undergoing any one of the previous procedures.

The TFC procedure, on the other hand, is sent to any signaling point, which originates an MSU and is received by the concerned signaling point. Without keeping track of every TFC message sent and the destination of each message, it is not feasible for a signaling point to inform another signaling point that it is not adjacent to the changed congestion status during the TFC procedure. For this reason, the responsibility is placed on the receiver of the TFC message to continuously monitor the congestion status of a route to determine when the status has changed. This is accomplished through the use of the RCT message.

The receiver of the TFC must send out the RCT message at the expiration of level 3 timer T15. This timer is set when the TFC message is first received. When the RCT message is sent, the timer is reset.

The message is sent using the structure shown in Figure 6.37. The test message provides only the H0/H1 heading code, indicating the type of test message. The MSU, however, carries the priority level of the concerned congested link. In other words, the congested route is considered congested at a specific level, which is determined by the concerned signaling points.

This congestion level, as discussed before, corresponds to the priority of an MSU. In ANSI networks, the priority of an MSU is indicated in the SIO subservice field.

The idea is that the MSU should be sent at the same priority level as the congestion status of the concerned route. If the test message is passed through the concerned signaling point towards the affected destination, then obviously the status of the route has changed. However, there is no indication as to what the current congestion level is (see Figure 6.38).

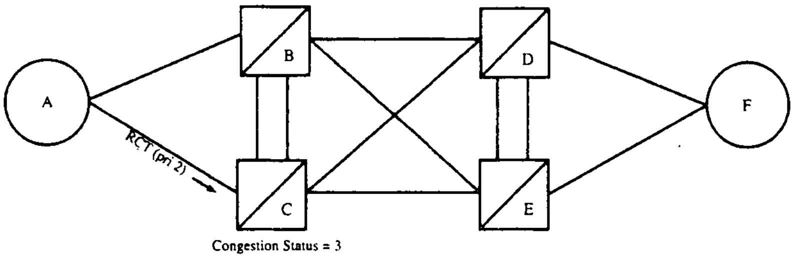


Figure 6.37 In this example, signaling point C is under congestion. Signaling point A has already been notified by a TFC message of the congestion status. The SRCT message is sent with a priority one less than the actual status of the route.

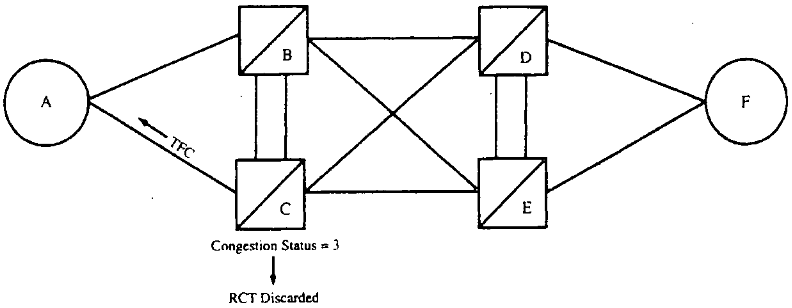


Figure 6.38 The SRCT was received, but because the congestion level has not changed, the SRCT was discarded. A TFC is returned to the originator (A) indicating the current congestion level.

If the congestion status has not changed and the MSU carrying the RCT message is received, the receiving STP will discard the message, and a TFC message will be returned to the originator of the test message indicating the present congestion status level towards the affected destination.

Remember that the MSU must have a priority equal to or more than the congestion status level. With every transmission of the RCT message, the MSU priority is set one less than the perceived value of the concerned route. For example, if the congestion status is determined to be at level 3, only MSUs with a priority of three will be passed through. The message is sent with a priority that is one less than what the originator thinks the congestion level is, based on the last TFC message received.

If the congestion level has not changed, the test message is discarded, and a TFC message is returned. If the congestion status has changed, the MSU is passed on to the destination. There is no indication or acknowledgment that the message was received and routed to the affected signaling point.

To determine when a test message has been enabled to pass through the concerned route, the originating signaling point of an RCT message sets timer T16. At the expiration of timer T16, if no TFC message has been received, it can be assumed that the test message has been passed on to the affected signaling point, and the congestion level has changed.

There is still no indication of what the new congestion level is. When the timer T16 expires, another RCT message is generated, with a priority one less than the previous test message. This new test message is sent once every T15, until the congestion status has changed, and the timer T16 expires with no receipt of a TFC message (see Figure 6.39).

This procedure continues until the congestion status has abated and the route is considered at congestion level zero. The congestion level is not defined by the standards but by each individual signaling point. This means that the congestion levels are implementation dependent, and there are no rules as to what constitutes a level 3 congestion status versus a level 1 congestion status.

The ANSI and Telcordia standards do indicate what type of messages should be allowed during the various congestion levels. The rules state that any network

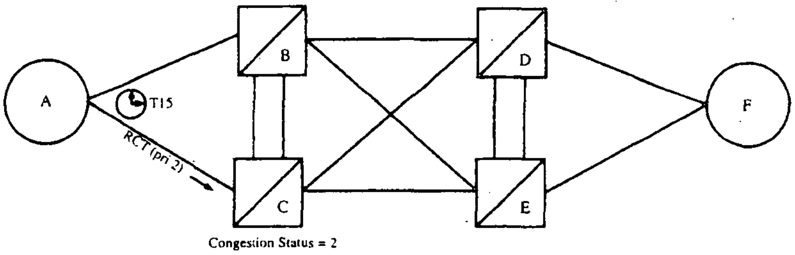


Figure 6.39 When the congestion level has decreased, eventually an SRCT will be accepted. The timer T15 triggers the transmission of SRCT messages.

management messages sent by level 3 should always have a priority of three. This, of course, is the highest priority available and ensures that network management messages always reach the affected destinations, despite the congestion status.

Any messages that are related to future connections, that is, voice circuit connections that have not yet been established, must have less priority than those messages related to existing connections. This is done to ensure that present connections are maintained properly, and any messages related to present connections can be routed through the network. Any new connections would then be based on the network's capability to route those requests through the congested route.

These messages should carry a priority of zero or one, being of a much lower priority than the existing one. This prevents new connections from being established during congestion periods, requiring more processing than what is available. This, of course, only impacts the congested route and does not prevent these connections from being established using other available routes, if any are available.

Any message sent in response to another previously received message, such as an acknowledgment, should be of the same priority as the request. For example, if a request was sent to a signaling point for information relating to an existing connection, the response should have the same priority as the request. This also ensures proper maintenance of existing connections and prevents messages from being lost and affecting these connections.

Any large messages should also be given low priorities. A large message requires additional processing resources, which adds to the congestion level of the route. This should be avoided whenever possible. What constitutes a large message is implementation specific.

The entire concept is to prevent additional processing requirements on a congested route from taking the link out of service. The route needs to be able to finish processing the connections it is already servicing and should be able to return to a normal state within a reasonable time if network management is successful in controlling the traffic flow to this route.

Network Maintenance Procedures

The procedures previously described are used by the network to maintain the reliability of the SS7 network. Usually, there is very little that can be done by service personnel during any of these procedures because the network is maintaining the status and invoking these procedures autonomously. There are occasions, however, when the network may not be successful in returning routes and links to service, requiring the intervention of service personnel.

This section will explain the big picture in network management, showing a variety of events taking place at all levels. The objective of this section is to show all the events that may take place during a failure or congestion, so that readers may have a better understanding of what is taking place and what to watch for on their own networks.

It is not the purpose of this section to prescribe any procedures that should be taken in the event that a failure or congestion is experienced because this will be dependent on your own company's maintenance procedures and the equipment used with the network. Understanding what is happening will assist you, however, in determining the next course of action.

Congestion Management

The use of various test tools can aid in troubleshooting a link and can help determine what types of tests need to be run. The types of test equipment vary, depending on the type of interface and the sophistication required. A transmission analyzer looks at the transmission from the perspective of the physical layer and does not perform any protocol analyzing.

When a high-level view is needed, a protocol analyzer or network monitoring system can aid in detecting both transmission problems and protocol problems.

A protocol analyzer will decode all layers of the protocol and allows a maintenance technician to determine at which level the error occurred. The protocol analyzer is best suited for the remote maintenance center, where messages can be monitored from and to all points on the network.

A network monitoring system usually provides protocol analysis as one of its many features, in addition to a number of other useful tools. A monitoring system collects data from all points on the network, providing a complete network-wide view of what is happening with the links, signaling points, and the protocol at all levels. A monitoring system is the most effective way to manage and protect a signaling network.

At the various exchanges, a transmission tester is best suited for troubleshooting link troubles under the direction of the remote maintenance center. When a transmission test is performed, the technician is testing the facility for its capability to send and receive bit streams to the other end of the link without error.

No addressing or other protocol functions are needed with this type of testing because the test is being performed between two entities. The purpose of such a test is to check the integrity of a signaling link. The SS7 protocol will also be checking the link integrity whenever the link is active.

In the event that the protocol takes a link out of service, maintenance personnel may be required to test the link and isolate the fault. This is not always the case, however, because most failures on digital facilities, such as DS0s, are clock oriented and can be resolved through the diagnostics of the protocol.

The procedures used for testing at this level are company dependent. The only suggestion here is to check each link before placing it into service, to alleviate any obvious problems that may occur when turning up a signaling point for the first time. Once the link has been placed into service, there should be a minimal amount of problems.

The SS7 protocol, as we have already seen, uses a hierarchical approach to network management. As we have discussed all through this chapter, there are three levels of network management:

- Link management
- Traffic management
- Route management

Even though we have already discussed these procedures in full detail, we haven't looked at the whole picture during a link failure or during congestion of a route. Let's look first at a congested route and what events could take place.

In Figure 6.40, the signaling point D has been notified of a busy condition on one or more of the links towards signaling point F. One of the links at signaling point F has experienced a busy condition. This occurs when too much traffic is sent over one link, and the processor used by that link cannot handle all of the traffic. When this occurs, the link management software initiates an LSSU with a condition of busy. The LSSU is sent by level 2 under the direction of level 3 link management. The LSSU is sent every T5 (level 2 timer), for a period of T6.

During the period in which this link is under congestion, it can still send MSUs, but it will not accept any from its adjacent signaling point. In fact, signaling point F, in this example, will hold all acknowledgments and negative acknowledgments until the congestion subsides.

Timer T6 prevents the link from remaining in a busy condition for too long. When this timer expires, the link is removed from service, and the alignment procedure is started. In this example, the link was still in congestion mode when T6 timed out. The link had failed, and the level 3 link management software began the alignment procedure (recovery).

Because there are only two links to signaling point F, the failure of one of these links has created a problem. Traffic must now be diverted away from the failed link onto the adjacent link within the same linkset. However, this link is already near capacity and adding the load of another link will possibly create a congestion condition on this link as well.

Level 3 link management places the link out of service by initiating the LSSU out of service towards signaling point D. This is done to ensure that signaling point D does not send any traffic onto the failed link. The link may be very capable of sending traffic, but signaling point F cannot process it. In fact, the link, in this case, must be able to carry traffic because the LSSU is sent on the same link for which it represents the status.

This brings up a good point. When a link has failed, the link is not always at fault. When we speak of a link failure, the link processor at either end is included as part of

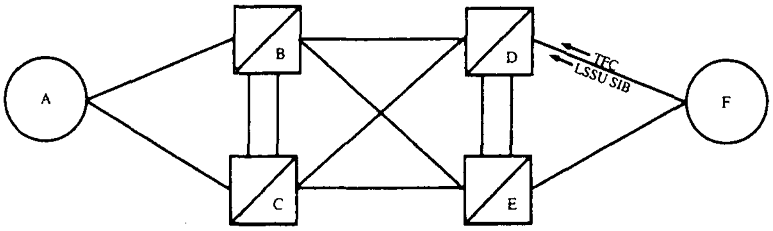


Figure 6.40 This figure shows a couple of activities. Congestion has occurred on the linkset from D to F. In this figure, the single line represents multiple links. A TFC is sent to indicate congestion. Meanwhile, one of the other links has become busy, prompting the LSSU with a *status indication of busy* (SIB).

the link. So even though the link itself is perfectly fine, the processor at either end or the interface card at either end could be at fault.

The LSSU is used at level 2 to inform the adjacent signaling point of the status of the link at the other end, so that both signaling points can be in synch during the diagnostics phase. Once the link has been removed from service (by transmission of the LSSU out of service), the signaling point F sends another LSSU with a status of out-of-alignment. This indicates that the link is no longer aligned and cannot be used to carry messages, other than level 2 messages.

The next LSSU to be sent carries the status of "normal alignment," indicating that the processor at signaling point F has begun the normal alignment procedure and signaling point D should do the same.

During the proving period, FISUs are sent from signaling point F to signaling point D. These FISUs are monitored for errors using the *alignment error rate monitor* (AERM) during the proving period, which is usually around two to three seconds.

If there are more than four errors during the proving period, the link has failed, and the alignment procedure begins all over again, beginning with the LSSU out of service. This procedure will continue until the link has been returned to service or until level 3 link management removes it from service entirely.

While the FISUs are being sent between the two adjacent signaling points, traffic must be diverted away from the failed link. This actually begins before the link begins the alignment procedure. This is initiated by level 3 traffic management. The purpose is to notify the adjacent signaling point that all traffic that was destined for the failed link must be rerouted to a new link. The traffic management message will also instruct signaling point D which link to use.

Normally, this would not be necessary. However, both ends of the link have a transmission and receiving buffer. These buffers hold transmitted and received MSUs until an acknowledgment is sent or received. In the case of the transmission buffer, these MSUs have been sent, but have not yet received any acknowledgments.

The traffic management at level 3 will first instruct signaling point F to move the contents of the failed link's transmit buffer to another link. The other link is either a link within the same linkset or of another route to the same destination. Once the MSUs have been moved to the new buffer, traffic management initiates the changeover procedure.

The COO is sent on the newly selected link, indicating the SLC of the failed link. This means that the link code table within the signaling point must also be modified. Every signaling point maintains an SLC table, which identifies the link code assignment for every link in the system. Remember that the link code is a logical assignment and does not necessarily correspond to the physical link code.

The SLC enables every link to have multiple codes. When a link is failed, the link must be removed from this table so that it is not selected by the routing function of level 3. All of the SLCs are then reassigned accordingly.

The receiver of the changeover message must mark the indicated link as failed, remove it from its own SLC table, and transfer the contents of the transmit buffer to the link on which the COO was received. When this has been accomplished, the link alignment procedure is able to start on the failed link. Signaling point D sends a changeover acknowledgment to signaling point F to indicate completion of the changeover procedure within itself.

All unacknowledged MSUs can now be resent from the transmission buffer, and newly generated MSUs can be sent via the new link as well. This will continue until the failed link has been returned to service.

At this point, network management has controlled the traffic between two adjacent signaling points and initiated a recovery procedure to return a link to service. No other signaling points have been informed of these activities. Unless the failed link creates congestion in the entire route, there is no need for further action. Let's see what would happen, however, if the condition did not change and congestion was to occur on the route.

In Figure 6.41, signaling point D has been diverting traffic under the direction from signaling point F to another link. However, signaling point D only has two links in its route to signaling point F. With one link carrying all of the traffic, signaling point D has become congested over that route. This does not mean that signaling point D is under congestion and cannot accept messages, but that messages destined for signaling point F cannot be sent through signaling point D without some method of controlling the throughput of these messages.

There are two methods used for controlling messages in this situation. Route management is primarily responsible for rerouting traffic away from a congested signaling point (or signaling point with congestion on one of its routes). At the same time, route management can also throttle the amount and type of MSUs sent to a specific destination. We will look at both of these procedures.

When signaling point D determines that the route has reached a predetermined threshold (usually determined by some configurable percentage, assigned at deployment time), the signaling point sends a TFR message to all adjacent signaling points. The TFR indicates a congestion condition to a given destination. Only the DPC is given in this message. The receiving signaling points (signaling points B and C in this example) must then determine which alternate routes they will use to route messages to destination F.

Remember that signaling point D is not the affected signaling point. Signaling point D is simply a relay station for messages destined to signaling point F. The signaling

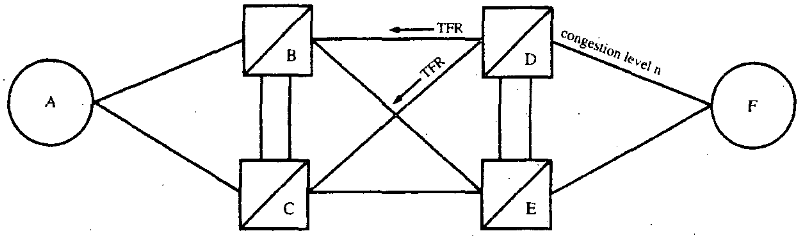


Figure 6.41 In this example, a link to signaling point F has become congested. Because this link is part of a route to F, the route congestion procedure is invoked.

point is not congested—just its route to signaling point F. Only messages destined for F are affected. All other messages can be sent to signaling point D with no impact because they are not routed over the congested route.

The TFR indicates that messages destined to signaling point F should not be routed through signaling point D. This does not stop signaling point D from receiving messages destined for F. If no other alternate routes are available, messages can still be routed through signaling point D to signaling point F. This could compound the problem of congestion if too many messages were being routed to D.

When signaling point D receives an MSU addressed to signaling point F, it must examine the congestion level of the affected link. We are no longer talking about the failed link because it is still tied up in the alignment procedure. The link that all traffic was diverted to has now become the problem. Signaling point D must determine the congestion level (0, 1, 2, or 3) for the link and notify the originator of any MSUs.

Congestion levels are also implementation dependent and can be configurable parameters based on network performance and the number of links to a given destination. In this example, we will say that the signaling link has reached a congestion level of 2. Level 3 is the most severe.

Signaling point D will use the congestion level of the link to determine which type of messages to allow through. For example, if the congestion level is level 2, only MSUs with a priority of two or above will be permitted to route through signaling point D to signaling point F. All other MSUs are discarded, and a TFC is sent to their originator.

The TFC message is sent by signaling point D to the originator of the received MSU. Only MSUs received and discarded can trigger the TFC message. The TFC message carries with it the current congestion level and the destination of signaling point F.

The receiver of this message then stops generation of all MSUs with a priority less than two. As discussed in the section about TFC procedures, there are certain types of messages that are enabled during this congestion status, and each type of message receives a particular priority. To review those priorities, refer to the discussion on the TFC procedure.

To determine when the route has become available again, the receiver of the TFC message must periodically send the RTC message. This message contains a priority of one in our example (because the TFC indicated that congestion status at level 2). This message is resent every T15, until the congestion abates.

We can now see that network management has been performed at various levels: the link level, the traffic level, and even the message-origination level. As the severity increases, so does the role of network management. In Figure 6.42, the failed link has been returned to service, and the congestion is subsiding.

Let's start by looking at the link again. The failed link has been through the alignment procedure and has successfully passed the alignment procedure. Before the link is returned to normal service, level 3 sends an SLTM. This message is also configured at deployment time and carries with it a predetermined test pattern. The purpose is to ensure the capability of the link to carry level 4 traffic and not just FISUs.

Once the SLTM has been sent and successfully received, the link is returned to service. Level 2 does not send any more LSSUs, but does enable MSUs through the link.

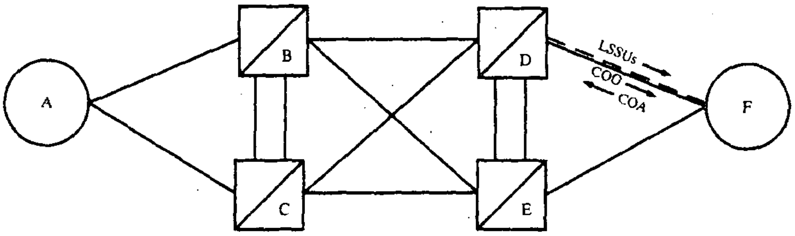


Figure 6.42 After testing a previously failed link by sending an SLTM, signaling point F sends a CBD, which is then *acknowledged* (CBA). Traffic is returned to normal.

Actually, level 3 is the driver here and initiates the transmission of MSUs again. Before this can happen, a few procedures must be initiated internally.

First, the link must be reassigned its SLC so that level 3 routing can select this link for transmission. Once the link has been restored in the link code table, level 3 traffic management sends a CBD message to signaling point D. The buffers do not need to be transferred this time because normal processing can continue on both links.

To indicate the success of the CBD, signaling point D sends a CBA on any available link. No other action is necessary. The failed link has now been restored, and traffic has been diverted back to the link. Both links are now operational. This means that the congestion condition should also be corrected.

As the congestion level subsides, the signaling link begins processing messages of a lower priority. No indication is sent by signaling point D of the new congestion level. However, any receivers of a TFC are still sending messages. This is their only means of determining the current status of the link.

When the congestion level has abated and the route is now free of congestion, the congestion level should be at level zero. Any signaling points sending messages will find they no longer receive a TFC when they send a test message with a priority of zero. They then mark the destination in their routing tables as available and begin normal transmission of signaling again.

Throughout this whole series of events, there has been no human intervention. In fact, this whole scenario may have taken only a few minutes to complete. By the time the events were detected by personnel at the remote maintenance center, the condition could have been repaired. This is what makes the SS7 network such a robust network.

Failure Management

In the preceding section, we talked about the procedures that would be used to correct a congestion condition. Now let us talk about the events that could occur during a network failure. We will start at the link level and watch as the failure migrates to an entire route.

In Figure 6.43, signaling point D has detected a failure at one of the links to signaling point F. This failure appears to be at the processor level. Level 2 is operational, level 3 is functional, but level 4 cannot be reached by level 3 message discrimination.

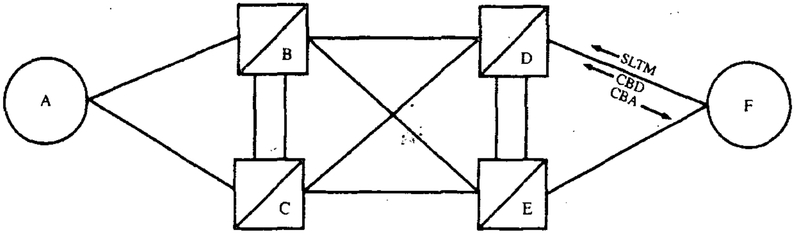


Figure 6.43 In this example, one link has failed in a two-link linkset. Level 2 LSSUs are sent on the failed link, while the changeover procedure begins on the other link.

This failure affects only one link in a two-link linkset. The first event to occur is the initiation of the LSSU by level 3 link management. Level 3 link management instructs level 2 to send the LSSU with a status of "processor outage" to signaling link F. Signaling link F thinks all is well until it receives the LSSU on the failed link. When it receives this message, it holds all MSUs to prevent transmission over the failed link. Level 2 has done its job.

Level 3 link management must now begin the recovery procedure on the link. This entails marking the link as out of service, which initiates changing the SLC and beginning the changeover procedure.

Level 2 sends the LSSU with a status of "processor outage" to signaling point F, indicating the processor failure at signaling point D. This should be followed by the LSSU of normal alignment. However, before normal alignment can begin, the traffic must be diverted away from the failed link.

The changeover procedure is invoked by level 3 traffic management. All unacknowledged messages in the transmit buffer of the link at signaling point D are transferred to the new link (in this example, the adjacent link within the same linkset).

When this procedure has been completed, the COO is sent to signaling point F providing the failed SLC. The COO, as seen in the illustration, is sent on the new link. When received, signaling point F transfers the MSUs still in the transmit buffer of the failed link to the new link. The MSUs are then retransmitted over the new link in both directions.

To acknowledge completion of the buffer transfer and receipt of the COO, signaling point F sends the changeover acknowledgment message to signaling point D. This signifies that all is completed and MSUs can now be diverted to the new link.

Now that the messages have been retransmitted and the buffers have been transferred, the link management recovery procedure can begin. This entails sending the LSSU of normal alignment over the failed link and beginning the alignment procedure. All timers and counters are set to zero, and the alignment procedure begins.

During the diverting of traffic, the processor outage problem has migrated to the new link. This means that both links to signaling point F are now inaccessible. Levels two and three are operational, but level 4 is not. This requires the link management procedures to be reported on the other signaling link.

The failure of both links has now isolated signaling point F from the rest of the network. While there is still another route available (through signaling point E), failure of E could cause a major outage.

The link management procedure on the last link of the linkset works a little differently than it did before. Now the emergency alignment procedure is used. This is virtually the same as the normal alignment procedure, except the time is much shorter (0.6 seconds), and only one error is allowed during the proving period.

Traffic management must now divert traffic away from the failed link and choose another alternate link, but this time, the links in this linkset are all out of service. This means that link management must choose a link within another linkset going to the same destination.

In this example, the other linkset is to signaling point C. The COO is sent to signaling point C on the linkset from D to C. All MSUs received by D for signaling point F will now be routed through C. Because we have already discussed what events occur during the COO, we will not go through them again here.

Traffic is now diverted away from the failed links and rerouted to the linkset from D to C. However, signaling point D is still receiving messages destined for signaling point F. To stop messages from being sent to D for F, signaling point D route management sends a TFP message to signaling points B and C. This will prevent either signaling point from sending an MSU destined for signaling point F through D.

Both signaling points B and C must now search for an alternate route. If there are no alternate routes, then the TFP message is sent to their adjacent signaling points to indicate that they can no longer reach the destination of signaling point F.

Notice that signaling point B does not need to send a COO to signaling point C. In essence, signaling point B is diverting traffic away from one route to an alternate route. This is done through routing table states.

In the event that either signaling point B or C becomes congested, they may enter into a TFC procedure. Hopefully, this will not occur, if both signaling points have ample links in their alternate routes.

Now let's see what happens when the links all return to service. The failed links between signaling points D and F are restored. This means they have successfully passed the alignment procedure, and they are capable of processing level 4 messages again. This is determined by level 3 message distribution.

The links must first be assigned the SLCs so level 3 routing may select them during the routing function. Once this has been completed, MSUs can be sent over the affected links.

The CBD message must be sent between signaling points D and C. This is to indicate to signaling point C that the traffic should now be diverted back to their old routes, and all MSUs destined for signaling point F can now be routed through signaling point D. At the same time, because the route to signaling point F through D is now accessible, a TFA message is sent by signaling point D to its adjacent signaling points (B and C) to indicate the accessibility of the route.

To prevent signaling point D from routing messages destined to signaling point F through signaling point C, both signaling points B and C may send a TFP message to signaling point D to prevent circular routing. Circular routing could occur if the route from signaling point C to F suddenly became unavailable. Messages would then be routed from D to C and then back up to signaling point B.

Once again, very little or no human intervention is required in such procedures. Remote maintenance personnel may redirect traffic through alternate networks, especially if a two-tiered network is used. In networks of this nature, the second tier enables routing through regionally located signaling points to route around clusters or regions that may be experiencing difficulty.

The rules for network outages in the RBOC networks are stringent. Any one interface should not be down more than three minutes per year. The user interface is the access from level 4 to level 3. This means that a processor outage (which indicates the failure of the interface with level 4) should not render the user part inaccessible for longer than three minutes.

A network access unit should not be down more than two minutes per year. This includes the access to signaling points from the central office, such as the SSP. These must remain accessible all the time. Failure at any of the access points (end nodes) means telephone calls cannot be made.

TCP/IP Networks

When using TCP/IP for the transport of SS7, network management functions provided by level 3 operate somewhat differently. First of all, the network devices themselves have no visibility to MTP level 3. In IP networks, these devices are routers and hubs, which use their own protocols for maintaining network reliability.

So MTP level 3 network management becomes a tool for the signaling nodes themselves to maintain virtual link reliability and to provide alternative routing schemes should the IP network fail.

At least one protocol used for the transport of SS7 over IP networks provides some level 3 functionality: the *Transport Adaptation Layer Interface* (TALI) developed by Tekelec and implemented by many vendors as a de facto standard. The purpose of this protocol is to emulate the network management functions of MTP level 3 in an IP environment, providing a much more robust alternative to IP networking than what has been proposed by other working committees (M2UA and M3UA).

Another rather unique approach of TALI is to incorporate some of the inherent features of the IP network into signaling nodes. When devices are connected to the signaling network, they automatically register with other devices on the network (provided they are running TALI as well). This registration is actually analogous, with many of the IP routing protocols that routers use to update their routing tables.

The purpose of this registration is to support network routing management in the IP environment and provide a means to route traffic around failed nodes within an IP network. The TALI protocol continually sends test messages to adjacent devices. Should an adjacent device fail to acknowledge within a specific time (determined by an administrable timer), the routing table of the node sending the test message flags the node as failed. Alternate routes are then used until the failed node becomes available again and sends a registration to all of its adjacent nodes.

The functions of MTP level 3 (and in some cases even MTP level 2) are still just as important when using IP links because IP networks do not have the same reliability and guaranteed service delivery that dedicated point-to-point channelized links provide. However, IP facilities do bring considerable advantages over channelized facilities, providing an equitable trade. As the standards evolve, the functions of MTP level 2 and 3 will be incorporated into IP protocols (TALI, M2UA, M2PA, M3UA, and SUA).

Chapter

7

General Description of SCCP Functions

The *Signaling Connection Control Part* (SCCP) is much like X.25 in the services it provides. The only really significant feature added by SCCP is global title translation. Many applications within the *Signaling System 7* (SS7) network rely on this routing feature. We will discuss global title translation in full detail a little later.

SCCP is a protocol used for accessing databases within the network. As part of level four, SCCP relies on the services of the *Message Transfer Part* (MTP) in networks based on *Time Division Multiplexing* (TDM), or the *MTP3 User Adaptation Layer* (M3UA), *MTP2 User Adaptation Layer* (M2UA), *Simple Control Transmission Protocol* (SCTP), or *Transport Adaptation Layer Interface* (TALI) in *Internet Protocol* (IP) networks. The primary difference between MTP and SCCP is in the addressing scheme and routing. SCCP also provides connection-oriented and connectionless services, whereas the MTP is strictly datagram.

The connectionless services provided by the SCCP are currently supported in *American National Standards Institute* (ANSI) networks. However, if one examines the procedures closely, the protocol uses a number of procedures and parameters to emulate connection-oriented services. In fact, SCCP has the capability to maintain a dialog with another network entity, just as if a virtual connection existed between the two resources. SCCP does not, however, establish a connection before beginning the dialog—hence, the reason it is considered connectionless. Still, the service it delivers is very much like connection-oriented service, which is why I classify it as a connection-oriented-like service.

When we look at the SS7 protocol stack, there is an indication that the *ISDN User Part* (ISUP) also uses the SCCP services to deliver messages on the network. The *International Telecommunications Union* (ITU), ANSI, and Telcordia standards all provide procedures for ISUP services over SCCP, but this has not yet been implemented. The thought is to allow SCCP to deliver end-to-end signaling capabilities for ISUP messages, rather than node-to-node via MTP.

When we examine the services provided by SCCP in contrast to MTP, we can begin to understand the fundamental differences between the two. MTP provides routing, sequencing, and flow control. However, SCCP also provides these functions. The difference lies in the user of these services.

SCCP relies on MTP to route its payload from one node to another. This means that SCCP must provide enough information to MTP that these functions can be carried out. SCCP provides the same functions, but the user is either the *Transaction Capabilities Application Part* (TCAP) or ISUP. This means that the functions used at SCCP will be somewhat different from those used by MTP. Let's look at the basic services that SCCP provides, and you can refer to Chapter 6, "Message Transfer Part (MTP) Level Three," to compare the two.

Services of SCCP

SCCP is divided into five classes of service, or protocol classes. Each protocol class defines which level of service SCCP is to provide. The five classes of service are as follows:

- **Class 0** Basic connectionless
- **Class 1** Sequenced connectionless
- **Class 2** Basic connection oriented
- **Class 3** Flow control connection oriented
- **Class 4** Error recovery and flow control connection oriented

The first two classes, class 0 and class 1, support the connectionless environment, which is all that is used in today's networks. Classes 2, 3, and 4 are used for connection-oriented services and, even though well defined, they are not used on today's network.

Class 0 services provide for the basic transport of TCAP messages. (The payload does not need to be TCAP, although this is most commonly used today.) No procedures are used to segment data or provide any sequencing of data. Typically, class 0 is used to deliver noncritical messages (such as database queries) when the guarantee of delivery is minimal.

Class 1 is used whenever more than one SCCP message exists for a transaction. A transaction is something that takes place at the TCAP level. The best way to define a transaction is as a function that is being requested by the application level.

For example, if there is a feature to be invoked at a remote switch, TCAP would send a message with several components. Each of the components would direct a particular portion of the switch to provide a function. All of the functions combined cause the feature to be invoked. The entire process is considered a transaction.

When this occurs, the TCAP message may be too big to fit in one SCCP message. SCCP must then provide segmentation of the data and possibly sequencing as well. The segmentation function divides the TCAP header and the application entity data into smaller segments, which can then easily fit within multiple SCCP messages. The TCAP header and the application-level information are left intact before the segmentation.

In other words, the TCAP header is not duplicated for each of the SCCP messages. Rather, it is segmented, as is, right into the payload portion of the first SCCP message. The sequencing function is used to ensure that the SCCP messages are received in the same order in which they were sent.

The routing function at level three examines the protocol class parameter to determine if the *signaling link selection* (SLS) field should remain the same or be rotated. If messages are part of the same transaction, the SLS is not rotated, ensuring the messages are received in sequence. If rotation of the SLS field is allowed, associated messages may be received out of sequence.

To ensure that the messages that are part of the same transaction are delivered in sequence, the routing function at level three examines the protocol class parameter to determine if the SLS field should remain the same or be rotated. If rotation is allowed, then a message may be allowed to travel through a different path, whereas if the SLS field is not rotated (bit rotation), the messages may be received out of sequence.

Protocol class 2 provides a basic connection-oriented service. A connection must first be established between the two entities and, once established, a two-way dialog may take place. In the case of SS7, the same physical link may be used to carry multiple connections at the same time. This is the same concept as multiplexing several telephone conversations onto one physical facility.

To maintain an order between the various connections, a reference number is established at both ends as an index to each of the particular connections. This allows either end to determine which virtual connection a message is destined for, much in the same way that X.25 uses virtual channels for a connection.

Protocol class 3 provides the same services as class 2, but adds the function of flow control and expedited data. Message loss and missequencing can also be detected and reported to the opposite end of the connection by way of an *SCCP management* (SCMG) message. When such an event occurs, class 3 has the capacity to reset the connection and restart transmission of the upper-level messages again.

Protocol class 4 adds error recovery to the class 3 functions. This has presently been removed from Telcordia standards, but is still defined in the ANSI standards. Error recovery supports the retransmission of errored messages.

All these protocol classes involve services of SCCP. Primitives are used to convey information between the levels, whether between the user part and SCCP, or between SCCP and MTP.

Figure 7.1 illustrates the various functions provided by SCCP. Through the use of primitives, all user parts that use the services of SCCP must discern whether they need connection-oriented or connectionless services. A separate interface is maintained for either one of these services.

The various routing functions of SCCP are handled by the *SCCP routing control* (SCRC) function, which interfaces directly with MTP. This includes any global title translation if the signaling point is equipped with that function. Remember that not all signaling points are equipped with global title translation.

The signaling point may also provide the reverse option of global title, which is translating the calling party address from point code and subsystem number into nothing

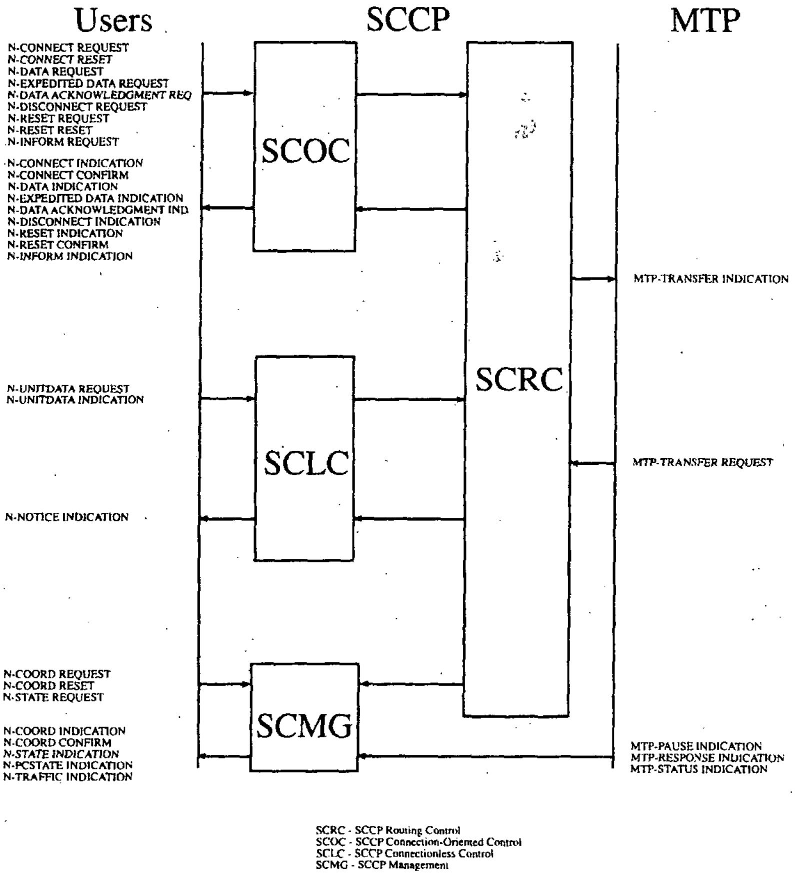


Figure 7.1 This figure illustrates the four functions defined in SCCP and the primitives used to interface to other protocols in SS7.

but global title. This is especially useful for secure networks where a message is being routed outside of the network and only the global title needs to be given to prevent outside networks from learning the point codes of internal signaling points. This is a function that would be provided by a gateway *signal transfer point* (STP).

MTP sends all received messages to the SCCP routing control for discrimination and distribution. If you remember our discussion from Chapter 6, you will remember that MTP also has a message discrimination function as well as a distribution function. The SCRC function is much like MTP in that it also must discriminate messages at the SCCP level and distribute to a higher level if addressed to the local application.

The called party address field provides the addressing information necessary for the SCRC to route a particular message to the correct destination. The SCRC is not concerned about which signaling point to route the message to because this is the function of MTP. The SCCP level is concerned only about its users, which are at the application level. Therefore, the destination point code is not used by the SCRC function. It exists simply for the sake of the MTP. The subsystem number is the only information (besides the global title, which is passed to the subsystem) that is needed at this point for routing by the SCRC.

The message distribution function routes the message to the correct user part, as defined in the called address field. Later, we will define the values of the various subsystems that have been defined by both Telcordia and ANSI and identify which applications they represent. Routing is discussed further in the next section.

The *SCCP connection-oriented control* (SCOC) and the *SCCP connectionless control* (SCLC) are both used to control functions such as segmentation and sequencing of messages. The user part must determine which service is required for the particular transmission. At the present time, only the SCLC function is used.

SCMG is notified any time a message cannot be delivered to a user part. In the case of the connectionless service, the received message is actually returned to the originator using the *Unitdata Service* (UDTS) or *Extended Unitdata Service* (XUDTS) message structure. In either one of these formats, the payload is also returned to the originator.

In the case of connection-oriented service, the connection request is denied and SCMG is notified. The data sent is not returned. SCMG may reset the connection, if one has been established, or simply return a connection reject message if no connection previously existed.

Routing Services of SCCP

As mentioned earlier, SCCP provides end-to-end routing functions to the TCAP. The TCAP protocol, in turn, provides transport services to the application entity with which it interfaces (such as the *Mobile Application Part* [MAP]).

The addressing requirements are somewhat different for TCAP than what the MTP is able to provide. MTP bases its addressing (the point code in the routing label) on information from SCCP. This information is found in a called/calling party address field in the SCCP message.

The called/calling party fields provide three forms of addresses. The first consists of digits. These digits, referred to as the global title, are usually what the subscriber (calling party) dialed (in the case of an 800 number call, for example). However, they do not have to be digits dialed. They can also be the *mobile identification number* (MIN) used in the wireless network to identify a wireless subscriber. At any rate, whatever the address is (dialed digits or some other form of digits), SCCP provides this information to MTP in the form of a primitive. This allows MTP level three to determine which signaling point to route the SCCP message to. If the destination point code is part of the SCCP called party address, this information is given to the routing MTP.

If this is the only information found in the called party address, then nothing else is included in the primitive to MTP. However, if the called party address includes a global title and subsystem number, then this information is given to MTP for transmission only.

MTP simply includes this information as part of the SCCP header and sends it to the destination. If MTP does not get this information, then these fields are filled with zeros.

Typically, when routing an ISUP message, level-three MTP knows what the end destination will be for a particular circuit. But remember that MTP needs to know only about signaling points that have a direct trunk connection to their location. This is because of the nature of the signaling taking place with ISUP services. The whole basis for this protocol is to establish a physical connection between the originating signaling point and an adjacent signaling point for a circuit connection (telephone call).

In TCAP transactions, the intent is different. We are no longer interested in circuit-related transactions. The purpose of TCAP is to provide a means for information exchange between two network entities, whether they are a database or another switch. This requires different routing procedures.

Global Title Translation

Network databases are typically centrally located within a network (wireline network). In order to route messages to these databases from switches, every switch in the network would have to know the routing address of the databases. This is not practical and is difficult to maintain (especially when the databases are in another company network). To simplify routing tables throughout the network, global title translation is used as an alternative.

Not every STP in the network needs to have this capability. In fact, global title translation is usually a function that is centralized within the network. The capability to provide this type of routing at the SCCP level allows MTP routing tables within individual signaling points to remain condensed, without the need to know every point code on the network.

When using the global title translation feature, messages from a *service switching point* (SSP)—which is always the originator of such messages—are routed to an STP, which must then translate the SCCP address fields into the point code/subsystem number combination. The message is then given back to the MTP with the additional address information so that it may be routed by MTP level three to its final destination.

This method of routing also allows networks to accept messages from other networks without disclosing their own internal point codes. For example, a company providing a database feature to other carriers may provide the point code of a gateway STP. The purpose of the gateway STP is to control who has access to the network (using a security feature called gateway screening) and also to provide global title translation.

The carriers route SCCP messages to the database service using only global title digits. It is then up to the database service to determine which of their databases will receive the messages by performing global title translation on the global title digits and routing the message through their own network using the point code/subsystem number combination.

I should mention here, by the way, that the point code in the called party address field is that of the *service control point* (SCP), which is providing an interface to the actual database. Although it is possible to have a database resident within an SCP, these devices are usually front-end database servers used to route SCCP messages to actual database systems located on an X.25 network or a *Transmission Control Protocol* (TCP)/IP network.

The subsystem number is actually the address of the database itself and is necessary for getting the TCAP message to the database. Each SCP may interface to a number of databases, each with its own subsystem number.

We can now see the fundamental differences between SCCP routing and MTP routing. SCCP is providing a more flexible routing scheme and actually provides three addresses: the global title, the point code, and the subsystem number. MTP routes to the point code of a signaling point, which has a direct voice facility connection or signaling points within a cluster. It may also address signaling points that provide centralized services, such as global title translation.

Flow Control

Another fundamental difference between MTP and SCCP is flow control. The MTP network management procedures of level three provide flow control to a signaling point. In the event that a signaling point experiences a failure at the link or processor level, level-three management provides the procedures for routing around the failure.

SCCP also provides flow control, but at a different level. As previously mentioned, SCCP interfaces to the TCAP and ISUP protocols, which, in turn, provide services to an application entity.

The flow control in the SCMG procedures provides management of message flow to a user part rather than a signaling point. In the event that a particular user part (today it is only TCAP) becomes congested, the SCMG function throttles the traffic destined to that user part. This has no effect on the traffic destined to the signaling point, unless configuration of the signaling point warrants rerouting.

Certain procedures allow for a database function or application entity function to be duplicated or replicated at another mated signaling point. If this is the case, then through configuration of the signaling point, the messages destined for a congested user part may be redirected to another signaling point.

This is not necessarily a feature of the protocol, other than an indication as to how the message should be handled. (The indication is through the multiplicity indicator, located in the SCMG messages.)

The multiplicity indicator tells whether there are duplicated subsystems or not. There are no handling procedures other than the indicator. If the indicator says that the subsystem is solitary, then messages to a congested user part can be returned to the sender by SCCP management. If the subsystem is duplicated, then the signaling point configuration can determine the method to be used for handling messages. They may be rerouted to the duplicate subsystem, or they may be returned. This, of course, depends on the implementation at the signaling point.

In *Regional Bell Operating Company* (RBOC) networks, messages routed to a congested or failed user part must be rerouted to the duplicate subsystem only if the duplicate subsystem is available and not under congestion itself. Otherwise, the message is discarded and a UDTS or XUDTS message with a return cause is sent to the message originator.

Message rerouting is also supported in the ANSI standards, but has not made its way into the standards publications as of the date of this publication. There is mention of the requirements, however, in Telcordia STP Generic Requirements TA-NWT-000082, Issue 5, May 1992 Revision 1, July 1992. This document cites the need for rerouting SCCP messages per the ANSI T1S1.3 standards group, but indicates that the requirement has not yet been added to the other standards publications and has therefore been included in the STP generic requirements publication.

Flow Control Procedures

Flow control is provided only in connection-oriented procedures. The purpose is to manage the number of data units sent on a particular connection. The unique thing about SCCP connection-oriented services is the capability to use one transaction to manage several connection sections.

The credit parameter in the connection request and connection confirm is used to establish the window size for a connection section. There may be multiple connection sections within one signaling point at any given time. The connection section is the equivalent of a virtual connection within an entity.

During the connection phase, the connection request message sends a credit parameter with the requested window size. This is based on the number of the messages that need to be sent. The credit may be negotiated between the two entities.

When the connection confirmation is returned, the credit parameter indicates the window size that was granted, based on the available resources of the destination signaling point. The actual credit granted may be larger or smaller than what was originally requested.

The sequence numbering then determines when additional messages may be sent. If the window size has already been met, then no other messages may be sent until the sent messages can be acknowledged and processed. Once they have been removed from the receiving signaling points buffer, additional messages may be sent.

The sequence numbers in the SCCP messages are used to notify both entities when messages have been acknowledged. This works the same way as message sequence numbering at the MTP level.

Another unique point about flow control compared to MTP is the fact that flow control at SCCP is used to control the flow of messages to a user part and not a signaling point. MTP flow control controls the traffic destined to a signaling point, whereas SCCP controls traffic to an application.

If the resources available to a particular user part (which serves as the communications interface to applications) become congested, then SCCP flow control is used to throttle the messages to the affected subsystem.

Connection-Oriented Services

SCCP supports connection-oriented services for TCAP and the ISUP. However, none of these uses connection-oriented services in networks today. It is important to remember that the protocol defines the procedures and functionality for a great many services, but only a fraction are actually implemented to date. For the sake of this book, we consider the possibility of all these procedures and functions being used at some point in time.

This is a particularly interesting development because the mechanisms used in the connectionless service emulate a connection-oriented protocol. Part of the reason for using connectionless versus connection-oriented services lies in the resources required to support hundreds of connections at any given time.

Because of the nature of the SS7 network, establishing a virtual connection with an application entity for every transaction that takes place in the network and maintaining that connection through the entirety of the transaction would require far too many resources. If the same information could be sent to the application without having to establish a connection, then the results would be the same, but with fewer resources.

Nevertheless, connection-oriented services are defined and described here because there may come a time when these services will be needed. Besides, no discussion could be complete without discussing all the capabilities of this network, instead of what is implemented.

Connection-oriented services provide for two types of connections: permanent and temporary. Permanent connections are established for operations, maintenance, and administration functions. These connections must be maintained permanently so that continuous real-time information can be exchanged between the network entities and the operations centers. In today's networks, this is not used because connection-oriented services are not supported.

Temporary connections are used for all other services and, as the name implies, are established on a temporary or as-needed basis. These must be established when a data transfer is requested and be released when a data transfer is complete. This works just like a telephone call in the *Public Switched Telephone Network* (PSTN).

For example, in the case of remote control of another telephone switch, there may be a need to establish a temporary connection before transmitting the actual data. This requires the services of connection-oriented SCCP, which will send a connection request to the distant end and establish a connection with the resources required at the remote switch. Once the connection has been established, the actual data, or control information, is sent over the established connection (which is actually a virtual connection, or session, if you think in mainframe terms).

When the transaction is complete and there is no other control information to be sent, the connection can be released. This means that whatever resources that were required for the transaction are now made available for other entities. Temporary connections, if used today, would be the most commonly used connection.

As mentioned already, connection-oriented services are not presently supported in ANSI networks, nor in any of the ITU networks known by this author. This may change in light of the many new *Advanced Intelligent Network* (AIN) features being deployed throughout the world. To date, there have not been any features defined that require connection-oriented services.

Even though connection-oriented services are not supported, the connectionless services do emulate many of the features of a connection-oriented protocol. Index numbers are used throughout the message in the TCAP portion to provide references back to previous transmissions, much in the way that virtual circuits are used in X.25. These index numbers are of no significance to the SCCP protocol. It is important to understand how these work and how TCAP works in general, so that one may understand the services of SCCP. You may want to read Chapter 8, "Overview of TCAP," first or review this chapter after reading about TCAP.

Connection-oriented services consist of several phases, which represent the various activities that take place during transmission of data between two entities. Some of these phases are represented during connectionless services as well.

The main disadvantage of connection-oriented services is the number of resources required for a connection-oriented transaction. For example, for two entities to exchange data regarding a mobile telephone subscriber (such as location data), a connection would be established first, virtually dedicating resources on both ends for the transaction. The data could then be transferred and the resources released.

The problem comes when many of these transactions are occurring throughout the network. In wireless networks, a mobile subscriber's phone may send a location update message every three to five minutes. You can imagine the number of such messages that must travel through a wireless network supporting 10,000 to 15,000 subscribers! This is one of the reasons connectionless is so widely used in SS7 networks today. Nevertheless, let's look at the procedures of the connection-oriented protocol and discuss the functions.

Connection-Oriented Procedures

As mentioned earlier, several phases take place during a connection-oriented transmission. The first phase is the connection establishment phase. This phase is when the resources of both entities, the receiver and the originator, are dedicated to the transaction.

This is then followed by the data transfer phase, which is when the data is actually being exchanged between the two entities. During this phase, no other entities are allowed to send anything to the resource that has been assigned to this transaction. Notice we said "resource." Switches and computers have multiple processors and are capable of handling many transactions at once. This is due to the usage of distributed processing.

Following the data transfer phase is the release phase. This consists of a volley of messages exchanged between the two entities, releasing the resources that were dedicated for the particular transaction.

To better understand the various phases and the exchange of messages that takes place, let's look a little closer at each of the three phases.

Connection Establishment. To understand the events that take place, we must first understand the routing of SCCP messages. A discussion of SCCP routing can be found in previous sections; here we need to talk about the differences between routing in connection-oriented and connectionless procedures.

In connection-oriented messages, the addressing is somewhat different than that of connectionless. The address information for all SCCP messages is located in two fields: the called party address and the calling party address.

In connectionless services, the called and calling party addresses represent the originating and destination point codes for the message. However, in connection-oriented services, each node involved in the connection establishes a connection between itself and the next intermediate node.

A good example of how this works is the voice network. A central office usually does not have a direct trunk connection to the final destination of the call, unless it is a local call. To reach the final destination, it must route through any number of intermediate switches.

During the call setup, a message is sent from the originator to the next intermediate node involved with the connection. The connection is then established between these two entities. Any messages regarding that connection are directed to the intermediate node and not to the final destination.

The intermediate node is then responsible for establishing a connection between itself and another intermediate node or the final destination if possible. This requires messages that are originated from the intermediate node to the next node in the connection. This same scheme is used for SCCP routing when connection-oriented services are to be used.

However, with connectionless services, the message can be addressed directly to the destination and, regardless of the number of intermediate nodes in the path, the message is routed according to the final address. No connection is established and the message addressing consists of the message originator and the final destination of the message.

In connection-oriented services, each node will have to establish two connections. The first connection is for the incoming message. The second connection is for the outgoing message. Both of these connections must be correlated to one another. This is referred to as a connection section in the protocol. The entity that establishes the connections must be responsible for maintaining an association between the incoming and the outgoing ports used for this connection.

Other messages can be received over these facilities and be addressed to other entities. This makes this connection section a logical, or virtual, connection, rather than a physical connection.

To establish a connection between two entities, the originator must send a *connection request* (CR) message. This is sent to the first intermediate node or the final destination if there is a direct connection available (the destination is an adjacent node to the originator). The CR contains the information necessary to define the parameters of the connection. This includes the *Quality of Service* (QoS)—which is defined by the protocol class—and the addresses.

The data parameter may contain bearer information to be exchanged with the application with which a connection must be established. A maximum of 130 octets may be sent in the data field.

To ensure minimal delay caused by routing through an excessive number of intermediate nodes, a hop counter is maintained. This hop counter is configured per network and decrements as it passes through each node. When the counter reaches zero, the message is in error, the connection is aborted, and an error message is returned with a cause code to the originator.

When the connection request is received by the destination, a *connection confirmed* (CC) message is returned. The CC is an acknowledgment that the resources necessary to maintain such a connection are available and are now reserved for the transaction. In the event that the resources are not available, a *connection refusal* (CREF) message is returned, along with a refusal cause code.

When a connection has been established, a local reference number is assigned to that connection (see Figure 7.2). This works much the same way as a virtual channel in the X.25 protocol. Other messages may be received over the physical link, even though this link has been established for a connection, but any messages bearing the same address and any previously established reference numbers will be handled in the same way as if the link was a permanent physical connection. You may think of this as a switched virtual circuit, which will be released upon termination of the transaction.

The reference number is assigned to both the incoming connection and the outgoing connection, but both independent of one another. They are referred to as the source local reference number (incoming) and the destination local reference number (outgoing).

All messages that come in to the signaling point on a connection are then given the source local reference number of that signaling point. When a message is returned to that entity, it may provide the reference number so that the signaling point then knows to which connection the message is relevant.

Reference numbers are of local significance only. In other words, if the reference number of a connection at the originating signaling point were sent to the destination signaling point, the destination would not know what the number was, because it was created and corresponds to a connection back at the originating signaling point.

The local reference number is used by the remote node as an address. It becomes the destination reference number for all subsequent messages. This address is not found in

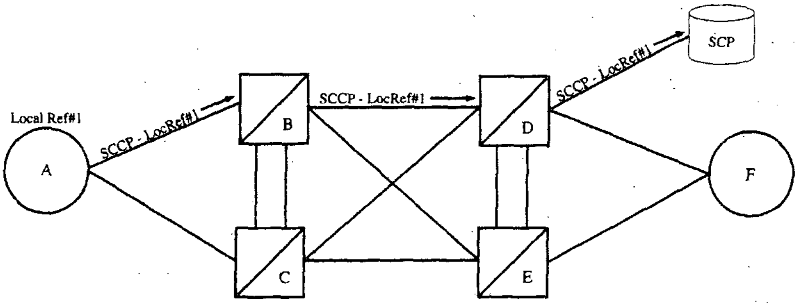


Figure 7.2 This figure illustrates an SCCP message being sent to an SCP with a logical reference value of 1. The local reference number is only of significance to A, which uses this number much in the same way as logical channels are used in X.25. Any responses to this SCCP message will carry the same local reference number.

the connection establishment phase because it has not yet been created. During the data transfer phase, all data transfer messages will contain a destination local reference number, which is assigned by the remote entity.

When a connection has been released, the reference number assigned to the connections section related to that connection is frozen. This means that the number may not be reassigned for a certain time. A timer is set to determine when the reference number may be unfrozen and available for reassignment to a new connection. This procedure is necessary to prevent an incoming message associated with a previous connection section from being routed to an incorrect reference number.

Also a part of the connection establishment phase is the negotiation for the QoS. The receiver may choose a different QoS based on present conditions within that node and the resources available. The QoS is directly related to the protocol class parameter.

If we review for a moment the valid protocol classes, this means that an originator may propose protocol class 3, which calls for connection-oriented services with sequenced delivery and flow control. The receiver may instead assign protocol class 2, which provides the sequenced delivery without flow control. The CC message is used to indicate which protocol class will be used.

If flow control is provided, the window size of the receiver may also be negotiated. This is accomplished using the credit field in the CR message and the CC message.

When the CR is sent, the credit field is sent with the proposed window size for the connection. The receiver may then assign a different (lower) window size if it wishes. This is sent using the connection confirm message credit parameter. The credit parameter is also related to the QoS of a message. This is assigned to a connection throughout the lifetime of the connection.

Data Transfer Phase. When a message containing data is sent over the established connection, the called party address is used by the SCCP routing control function to determine which connection section to route the message over. All subsequent data messages with this called party address will be routed over the same connection section.

In the event that there is an incoming and outgoing connection section (represented by the source local reference number and the destination local reference number), then the SCCP routing control must determine which of the outgoing ports is associated with the source local reference number and the SCCP routing control routes the message to that destination.

The message, when received by the final destination, is given by MTP to the SCCP routing control to determine the address of the message. If it is determined that the address is of the signaling point that has received it, then the data is given to the user part (TCAP or ISUP) via a primitive. Remember that primitives are used whenever a lower-level function is passing information up to a higher-level function. The primitive is the interface through software that communicates with the other signaling point entities.

Sequence numbering is provided (if the protocol class warrants) by using the sequence number parameters in the *data form 1* (DT1) and the *data form 2* (DT2) messages. The sequence numbers act as an acknowledgment for previously received messages as well. This works much the same way as in other protocols and as sequencing works at the MTP level.

The originator of data may send up to 127 data messages through a connections section in one direction. Another 127 messages may be sent through the same connection section in the opposite direction. At any time, the receiver may change the size of the window, depending on resources and other parameters, by changing the value of the credit parameter. The credit parameter indicates the number of messages that may be sent in any one direction.

For example, if the credit parameter has a value of 7, only 7 messages may be sent in one direction (towards the sender of the credit parameter of 7). When the receiver of this credit has sent 7 messages, it must wait until it receives a message acknowledging receipt of the previously sent 7 messages. It may then continue transmission.

If the credit size has been changed, then the sender is limited to the number of messages it may send or, if the number has been increased, the sender may be able to send more messages. This dynamic window size allows the protocol to control the number of messages sent to an entity according to real-time events, rather than using a fixed parameter that is not sensitive to events that may take place during transmission (such as congestion).

Data acknowledgment messages may be sent even when there is no data to be transmitted. This allows for the acknowledgment of received messages and allows transmission to continue, even when it is only in one direction.

Some messages may exceed the capacity of the SCCP envelope. When this occurs, the data must be segmented into multiple packets before transmission. Keep in mind that the level above SCCP (the user part) includes header information. This header information is not included in every single data segment. Rather, the data and the header information are sent in their entirety to SCCP routing and, when segmented, the first set of bits is encapsulated and transmitted.

The remaining bits are then encapsulated and transmitted, with no knowledge of header information. The header information of the user part will then be received along with some of the data in the first segment. The header information will be in its entirety. The remaining data will be received in subsequent SCCP messages. Segmentation is not necessary if the message is equal to or less than 255 octets.

To ensure that data has been received, an expedited data message may be sent instead. The expedited data message allows up to 32 octets of user data to be sent to the destination. No further data may be sent, however, until an acknowledgment has been received. Once an acknowledgment has been received, then additional expedited data messages may be sent.

This is only true on one connection section. Multiple expedited data messages may be sent to a node (signaling point) on different connection sections, but only one at a time may be sent on any one connection section. This ensures that the transmitted data is indeed received without error. Flow control can be accomplished by withholding the expedited data message acknowledgment. Expedited data only applies to protocol classes 3 and 4.

In the event that two entities are no longer in sync with each other, a reset can be initiated. The reset changes the sequence numbering back to 0 at both ends of the connection, and it changes the window back to its initial size when the connection was originally established. The credit field is also reset to 0.

When a reset message is received, the receiver also resets its sequence numbers back to 0. Any data messages received after the reset request are discarded. A confirmation must be sent to confirm that the reset has been initiated before data can begin transmitting again. Once the confirmation has been sent, the originator of the reset begins sending data using sequence numbers beginning with 1.

Release Phase. Once the data transmission has been completed, and no further transmissions are necessary, a release may be initiated. The release may be initiated by either node at any time during the life of the connection. How ever, there are measures that are taken at the user part to ensure that a connection is not prematurely released before an entity has completed its transactions.

The TCAP uses a permission parameter, which is inherent within specific message types, indicating whether or not an entity has permission to disconnect a connection or not. Permission is not granted when there is additional data to be sent in association with a particular transaction.

As previously mentioned, the release may be initiated by either node and by the user part or SCCP. When SCCP requests a release, it usually indicates a problem with the connection. SCCP may request a release or a pause on the connection, or it may also request a release without permission to reestablish the connection.

The release cause parameter indicates the reason for the release and will also implicate the originator of the release. A release must be acknowledged by a release complete before the connection is considered available for another transmission.

Connectionless Services

A connectionless transfer of data using the services of SCCP requires the use of the unitdata and the extended unitdata message structures. These message structures provide all the information necessary for data to be transferred to and processed by a remote entity.

Two protocol classes support connectionless services: protocol class 0 and protocol class 1. When protocol class 0 is used, there is no guarantee that the subsequent data will arrive in the same order in which it was transmitted. That is because the signaling link selection field in the routing label of the MTP header may be rotated at each node. When this occurs, a message may travel a different route than its associated messages.

Due to cross delay, messages are received out of sequence when this occurs. To prevent this from happening, protocol class 1 can be specified. Any message with a protocol class of 1 indicates that the signaling link selection field should not be rotated, and any other messages received for the same destination are transmitted over the same SLS as the previous associated messages. This ensures that messages that are associated with one another follow the same path and do not get delivered out of sequence.

There is really only one phase during connectionless procedures: data transfer. There is no connection establishment because a connection is not necessary. Data is encapsulated into a message envelope with all of the information necessary for the receiving entity to process the information as it was received.

Subsequent data may be sent in the same fashion as long as there is enough information for the receiving entity to process the data. Obviously, connectionless services do not require the same resources as connection-oriented services and thus have found more favor in SS7 networks today. Let's look a little closer at the procedures used to transfer data with a connectionless protocol.

Connectionless Procedures

The application service element wishing to send data to a remote entity requests connectionless services from SCCP. The application service element can be TCAP or some other transport mechanism used by an application entity. The application entity could be the *Operations, Maintenance, and Administration Part* (OMAP) or MAP.

The method used to transport data between two entities is transparent to the application entity and is the responsibility of the application service element, such as TCAP. All routing is handled by SCRC, which provides routing parameters to MTP.

In large networks, it is not feasible for every node to know the address of every other node. For this reason, the SCCP routing control function can provide additional translation services. This is known as global title translation and is explained in detail in the preceding section on SCCP routing.

When there is too much data to fit into one SCCP envelope, or when it is determined that the message should be segmented (this can be determined by network management based on network conditions), the *extended unitdata* (XUDT) is used. The data is then divided into equal-length segments. The rule is that the first segment should be sized in such a way that the total message length is less than or equal to the size of the first segment multiplied by the number of segments being sent. This is to prevent buffers from becoming unmanageable.

All subsequent messages are configured with the same address information, and protocol class 1 is selected to ensure the in-sequence delivery of all messages. The segmentation parameter in the XUDT message is set to indicate that additional segments are to be sent, and the segment number field indicates how many additional segments are yet to be transmitted.

In the event that a message is received in error, an error message is returned to the originator of the message. The error message is in the form of the unitdata service or extended unitdata service messages. The data is also returned, along with a return cause code, which indicates why the message is being returned.

Unlike connection-oriented services, connectionless services do not assign any logical reference numbers to transmissions because there is no connection to be established. Therefore, tracking is not important. The only objective is to get the data to its destination. At the user part level, many schemes are used to ensure that messages are received without error and, at least in the case of TCAP, indexing schemes keep track of associated data.

SCCP Management (SCMG)

SCMG is used to maintain the integrity of SCCP services. Although MTP maintains link integrity and alerts adjacent signaling points of congestion at another signaling point, SCMG is concerned about the status of a subsystem or application entity.

To accomplish this, SCMG is divided into three tasks: signaling point status, subsystem status, and traffic management. Management messages use the unitdata message structure found in connectionless SCCP.

To maintain the status of the signaling point, SCMG relies on information from the MTP that is sent to SCMG through primitives. These primitives consist of the MTP-Pause, MTP-Resume, and MTP-Status primitives and their parameters. Subsystem information is gathered by SCMG through primitives from the subsystem directly to SCMG.

Probably the most advantageous feature of SCMG is the capability to route messages away from a failed or congested subsystem to a mated subsystem in another location. This ensures that services are not lost when a subsystem fails, and it guards against failures due to subsystem congestion.

This requires that subsystems be replicated or duplicated and placed in different geographical locations. When this has been adhered to, the diversity of the network increases, and the reliability increases. The protocol can now control message flow to the replicated databases.

A database can play several roles within the protocol structure. One is as the dominant subsystem. The dominant subsystem will hold a higher priority than its replicated subsystems. All replicated subsystems possess the same subsystem number, so it is up to the signaling points to determine how to route to the subsystems. This is decided at configuration time, and each signaling point is configured to handle SCCP traffic according to these rules.

Each replicated subsystem may have a priority, with the subsystem that has the highest priority receiving the bulk of the traffic. The assignment of the dominant subsystem may change dynamically or may be a fixed configuration, depending on the network. One thought is to allow the priority to change based on the current load. This provides for a more dynamic routing scheme, but could prove difficult to implement.

A solitary subsystem is one that is not replicated and must therefore handle all traffic. In the event that this subsystem fails, traffic may be routed to another network or stop altogether. This is the least favorable situation on any network, because it increases the single point of failure and decreases reliability on the network.

Another scheme is to have a primary subsystem that receives all traffic until a failure occurs, in which case, all traffic is routed to an alternate. The alternate would then handle all SCCP traffic until a failure occurs, in which case it would be marked as inaccessible and all traffic would be routed back to the other alternate.

This uses a standard master/slave relationship, but it is not favorable because there is no use of the other subsystem until a failure occurs. This means that the other subsystem is sitting idle and not being utilized in any capacity. If something is wrong with this subsystem (in terms of being able to handle messages), you will not likely find out until it goes online and begins handling messages, which is a little too late.

SCMG also uses a concept referred to as the concerned point code. There are really two point codes that are affected by SCCP management. One is the affected point code, which is the failed or congested entity. The other is the point code that utilizes the service of the affected point code. The concerned point code must be notified when there is a status change at the affected point code, so that it knows how to route SCCP messages. The concerned point code is updated about an affected point code's status using SCMG messages and connectionless services of SCCP.

SCMG messages are sent to adjacent signaling points (adjacent in the logical sense only) to alter the translation functions located within those signaling points. By altering the translation function, when a protocol class 0 message is received, messages can be routed to other replicated subsystems. The action to be taken depends on the type of SCMG intervention.

Signaling Point Status Management

Signaling point status management is concerned with the status of an SCP. If the SCP becomes congested or should fail, then the subsystems adjacent to the SCP cannot be reached. Traffic must then be diverted to replicated subsystems.

This requires a series of management messages that provide the status of a signaling point (SCP) and subsystem combination. These messages are not to be confused with the messages used by network management at level three, although there are some similarities. Level three is more concerned with all the signaling points within the network, rather than just a select type.

A signaling point prohibited procedure indicates that the affected signaling point has been prohibited and cannot receive any traffic. The signaling point will be an SCP, rather than an SSP or STP, because this is the only entity SCMG is really concerned about.

When a prohibited message has been received, the receiving node changes its translations to route traffic to the replicated subsystems, if any exist, according to the configuration of the replicated subsystems (dominant role, alternate role, or solitary).

When the signaling point (SCP) is considered to be "allowed," the translation tables are once again modified according to the roles of the replicated subsystems. Traffic is then allowed to be directed towards the affected signaling point, and subsystem status tests may be invoked.

All translation changes are made at adjacent nodes. Adjacency refers to a logical adjacency. This means that a path exists from one signaling point to the affected signaling point. An example would be an STP that provides global title translation. This entity needs to be kept apprised of the signaling point status because it will have to change its routing tables and translation tables based on the status of the SCPs and their subsystems.

Subsystem Status Management

The purpose of subsystem status management is to monitor the status of individual subsystems within a signaling point. An SCP may have multiple subsystems. If the signaling point becomes congested or fails, then none of those subsystems can be reached (signaling point status management). If only one of the subsystems becomes congested or fails, then subsystem status management redirects traffic from that one subsystem to other replicated subsystems.

This should point out the fundamental difference between signaling point status management and subsystem status management. One is concerned with the status of the SCP while the other is concerned with the status of the subsystems located within or adjacent to the SCP.

Translation tables must also be changed to allow routing to be diverted away from the failed or congested subsystem and be routed to replicated subsystems, depending on their roles in the network (solitary, dominant, or alternate).

The same status is provided for subsystems as it is for signaling points. A subsystem is either prohibited or allowed. There is no restricted mode as used by level-three management. If a subsystem is unable to handle traffic due to congestion or failure, then traffic is immediately diverted to replicated systems. The throttling of traffic cannot be tolerated, due to the nature of the transactions taking place at a subsystem.

When a subsystem is marked as prohibited, a status test is used to audit the prohibited subsystem and ensure the status is correct. The test is invoked by the SCP so that it may keep track of the status of all of its subsystems. If, for any reason, the subsystem status has changed to allowed and the subsequent management messages are not received, a subsystem could remain prohibited in the status tables of the adjacent SCP, while the actual status of the subsystem is allowed.

In addition to testing the status of a subsystem, an SCP may also invoke a broadcast that allows the SCP to inform other local subsystems (concerned subsystems) of the status of other signaling points or subsystems. This is reserved for local subsystems, which means they have a direct adjacency to the SCP and can be reached through the use of primitives, rather than protocol messages.

Other signaling points can be notified through the use of a broadcast procedure for signaling points. This procedure is used to inform concerned signaling points of status changes concerning subsystems. Only concerned signaling points are sent status updates. A concerned signaling point is that which regularly routes messages to the SCP/subsystem combination. Usually, this is an STP, which provides the global title translation function for the rest of the network.

Now you can see one of the distinct advantages behind using a centralized STP with global title translation, rather than spreading the routing function through all of the nodes. SCMG, as well as routing, can be simplified if only a few of these entities provide this functionality.

There is also a procedure that allows for the calculation of traffic mixes, although much of this is still under study. Traffic mix information can be provided as an option in ANSI networks and can prove useful to some databases for the purpose of network management and network monitoring.

The traffic mix indication informs end databases as to the type of SCCP traffic being routed: normal SCCP traffic or backup. Normal SCCP traffic is what would normally be routed to the subsystem without network management intervention. Backup traffic consists of all messages routed to the subsystem as the result of an SCCP management function. This indicates that the receiving subsystem is a replicate subsystem and is receiving traffic from another subsystem that is prohibited.

As mentioned earlier, a subsystem is either prohibited or allowed. No procedures are currently defined for flow control to a subsystem. Because of the nature of the transactions that take place at a subsystem, flow control may not be a viable option.

SCCP Message Structure

SCCP is divided into several sections, as we saw earlier. In this section, we will define the various fields and values for these fields, and identify where they can be found in the SCCP message. These fields are defined according to their location in the SCCP message, which has three parts: the mandatory fixed part, mandatory variable part, and optional part (see Figure 7.3).

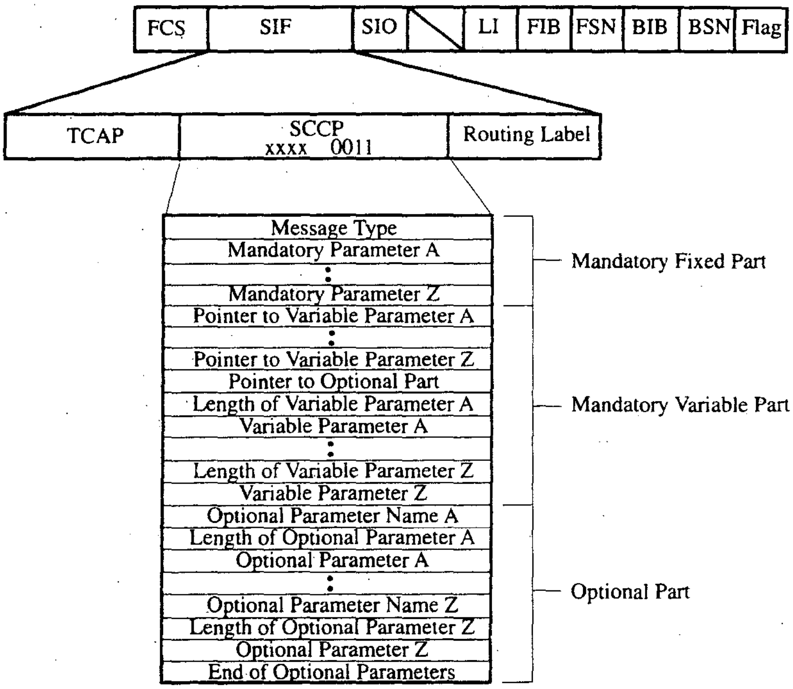


Figure 7.3 This figure identifies all the fields in an SCCP message and its location in relation to the rest of the protocol.

The mandatory fixed part consists of those parameters that are mandatory for the particular message. Each message will have different parameters that will be fixed in length (according to the message type). The message type may have one octet or may have several octets of parameters. The field will not vary, however, and because of the message type, it can be determined how large these parameters will be.

The mandatory variable part consists of those parameters that are required of a particular message type, but are not of a fixed length. A good example of a mandatory variable part is a called or calling party address. This field uses length indicators to identify the length of each individual parameter field and the beginning of the parameter. Also in the mandatory variable part is a pointer that identifies the beginning of the optional part. This field consists of the binary representation of the binary offset. An offset is the octet count from the beginning of the pointer to the optional part indicator.

Each of the parameters found in the mandatory variable part is preceded by a length indicator and the parameter indicator. A pointer at the beginning of the mandatory variable part points to the octet of each of the individual parameters.

The optional part also uses length indicators after every parameter. The parameter is identified by a one-octet indicator, which provides a unique pattern for every parameter type.

Following the optional parameter is the length indicator, which provides the length for the entire parameter, excluding the parameter indicator. The length indicator is then followed by the parameter itself.

Not all SCCP messages will use all of these fields. Some SCCP messages will only use the mandatory fixed part; others will only use the mandatory fixed and variable parts. The following are the definitions for the various parameters found in the SCCP message structure.

Mandatory Fixed Part

The mandatory fixed part succeeds the message type code, which identifies which type of SCCP message is being sent. As discussed in the introduction, two types of services are provided by SCCP: connection oriented and connectionless. The connection-oriented classes of service require several different types of SCCP messages, while the connectionless requires only one. In the United States, connection-oriented classes of service are not supported. We will define them here anyway because there may be a use for these services in the future.

Even though there are no present functions for connection-oriented services using SCCP, many functions of SCCP emulate connection-oriented services. We will look at the various parameters that are used for connection-oriented emulation when we examine the mandatory variable fields and the optional parameters.

Mandatory Variable Part

The parameters found in this field will depend on the type of message being sent. Each message has different requirements and may or may not require variable parameters. Not all SCCP messages will use the mandatory variable part.

The "mandatory" indicates that this field may be required for specific message types. The "variable" indicates that the length of these parameters is not fixed in nature and will be variable. This includes addresses and other parameters, which will vary from message to message.

Optional Part

The optional part is always a variable-length field and may or may not be used with specific message types. "Optional" indicates that the field is not required for a specific parameter, but may be used to provide additional information relating to a transaction.

Length indicators are used before every parameter in this field to delineate between the various parameters. By providing these length indicators, the receiving signaling point can determine where the beginning and the end of a parameter are without the use of pointers.

Message Types

The first field of the mandatory fixed part is the message type. This field is found in all SCCP messages. The message type will determine which parameters will be used in the mandatory variable part and the optional part.

The mandatory fixed part will be followed by variable and optional fields in some situations. This depends on the message type. The various parameters will provide additional information, again depending on the message type (the parameters are not shown in their entirety in this section).

This section describes the message types that are supported for connection-oriented and connectionless services. These message types and their parameters are shown as follows with explanations for the message types and their functions. The parameters are explained in their entirety in the next section.

End of Optional	Hop Counter	Data	Clg Party	Credit	Cld Party	Proto Class	Src Loc Ref	Msg Type
8	24	24 to 3120	32+	24	24+	8	24	8

Connection Request (CR)	**0 0 0 0**	**0 0 0 1**
Mandatory Fixed Part		
Source Local Reference	0 0 0 0	0 0 1 0
Protocol Class	0 0 0 0	0 1 0 1
Mandatory Variable Part		
Called Party Address	0 0 0 0	0 0 1 1
Optional Part		
Credit	0 0 0 0	1 0 0 1
Calling Party Address	0 0 0 0	0 1 0 0
Data	0 0 0 0	1 1 1 1
SCCP Hop Counter	0 0 0 1	0 0 0 1
End of Optional Parameters	0 0 0 0	0 0 0 0

End of Optional	Data	Cld Party	Credit	Proto Class	Src Loc Ref	Dst Loc Ref	Msg Type
8	24 to 3120	32+	24	8	24	24	8

Connection Confirm (CC)	**0 0 0 0**	**0 0 1 0**
Mandatory Fixed Part		
Destination Local Reference	0 0 0 0	0 0 0 1
Source Local Reference	0 0 0 0	0 0 1 0
Protocol Class	0 0 0 0	0 1 0 1
Optional Part		
Credit	0 0 0 0	1 0 0 1
Called Party Address	0 0 0 0	0 0 1 1
Data	0 0 0 0	1 1 1 1
End of Optional Parameters	0 0 0 0	0 0 0 0

End of Optional	Data	Cld Party	Cause	Dst Loc Ref	Msg Type
8	24 to 3120		8	24	8

Connection Refused (CREF)	**0 0 0 0**	**0 0 1 1**
Mandatory Fixed Part		
Destination Local Reference	0 0 0 0	0 0 0 1
Refusal Cause	0 0 0 0	1 1 1 0
Optional Part		
Called Party Address	0 0 0 0	0 0 1 1
Data	0 0 0 0	1 1 1 1
End of Optional Parameters	0 0 0 0	0 0 0 0

End of Optional	Data	Cause	Src Loc Ref	Dst Loc Ref	Msg Type
8	24 to 3120	8	24	24	8

Released (RLSD)	**0 0 0 0**	**0 1 0 1**
Mandatory Fixed Part		
Destination Local Reference	0 0 0 0	0 0 0 1
Source Local Reference	0 0 0 0	0 0 1 0
Release Cause	0 0 0 0	1 0 1 0
Optional Part		
Data	0 0 0 0	1 1 1 1
End of Optional Parameters	0 0 0 0	0 0 0 0

Src Loc Ref	Dst Loc Ref	Msg Type
24	24	8

Release Complete (RLC)	**0 0 0 0**	**0 1 0 1**
Mandatory Fixed Part		
Destination Local Reference	0 0 0 0	0 0 0 1
Source Local Reference	0 0 0 0	0 0 1 0

Data	Seg/Reassembly	Dst Loc Ref	Msg Type
16 to 2048	8	24	8

Data Form 1 (DT1)	**0 0 0 0**	**0 1 1 0**
Mandatory Fixed Part		
Destination Local Reference	0 0 0 0	0 0 0 1
Sequencing/Reassembling	0 0 0 0	0 1 1 0
Mandatory Variable Part		
Data	0 0 0 0	1 1 1 1

Data	Seq/Segment	Dst Loc Ref	Msg Type
16 to 2048	16	24	8

Data Form 2 (DT2)	**0 0 0 0**	**0 1 1 1**
Mandatory Fixed Part		
Destination Local Reference	0 0 0 0	0 0 0 1
Sequencing/Segmenting	0 0 0 0	1 0 0 0
Mandatory Variable Part		
Data	0 0 0 0	1 1 1 1

Credit	Rcv Seq #	Dst Loc Ref	Msg Type
8	8	24	8

Data Acknowledgment (AK)	**0 0 0 0**	**1 0 0 0**
Mandatory Fixed Part		
Destination Local Reference	0 0 0 0	0 0 0 1
Receive Sequence Number	0 0 0 0	0 1 1 1
Credit	0 0 0 0	1 0 0 1

Data	Clg Party	Cld Party	Proto Class	Msg Type
16 to 2032	2+	3+	8	8

Unitdata (UDT)	**0 0 0 0**	**1 0 0 1**
Mandatory Fixed Part		
Protocol Class	0 0 0 0	0 1 0 1
Mandatory Variable Part		
Called Party Address	0 0 0 0	0 0 1 1
Calling Party Address	0 0 0 0	0 1 0 0
Data	0 0 0 0	1 1 1 1

End of Optional	ISNI	Segment	Data	Clg Party	Cld Party	Hop Cntr	Proto Class	Msg Type
8	24 to 144	48	16 to 2032	2+	3+	8	8	8

Extended Unitdata (XUDT)	**0 0 0 1**	**0 0 0 1**
Fixed Mandatory Part		
Protocol Class	0 0 0 0	0 1 0 1
SCCP Hop Counter	0 0 0 1	0 0 0 1
Mandatory Variable Part		
Called Party Address	0 0 0 0	0 0 1 1
Calling Party Address	0 0 0 0	0 1 0 0
Data	0 0 0 0	1 1 1 1
Optional Part		
Intermediate Signaling Network Identification (ISNI)	1 1 1 1	1 0 1 0
Segmentation	0 0 0 1	0 0 0 0
End of Optional Parameters	0 0 0 0	0 0 0 0

Data	Clg Party	Cld Party	Return Cause	Msg Type
16 to 2032	2+	3+	8	8

Unitdata Service Message (UDTS)	**0 0 0 0**	**1 0 1 0**
Mandatory Fixed Part		
Return Cause	0 0 0 0	1 0 1 1
Mandatory Variable Part		
Called Party Address	0 0 0 0	0 0 1 1
Calling Party Address	0 0 0 0	0 1 0 0
Data	0 0 0 0	1 1 1 1

End of Optional	ISNI	Segment	Data	Clg Party	Cld Party	Hop Cntr	Rtn Cause	Msg Type
8	24 to 144	48	16 to 2032	2+	3+	8	8	8

Extended Unitdata Service Message (XUDTS)	**0 0 0 1**	**0 0 1 0**
Mandatory Fixed Part		
Return Cause	0 0 0 0	1 0 1 1
SCCP Hop Counter	0 0 0 1	0 0 0 1
Mandatory Variable Part		
Called Party Address	0 0 0 0	0 0 1 1
Calling Party Address	0 0 0 0	0 1 0 0
Data	0 0 0 0	1 1 1 1
Mandatory Variable Part		
Intermediate Signaling Network Identification (ISNI)	1 1 1 1	1 0 1 0
Segmentation	0 0 0 1	0 0 0 0
End of Optional Parameters	0 0 0 0	0 0 0 0

Data	Dst Loc Ref	Msg Type
16 to 264	24	8

Expedited Data Message (ED)	**0 0 0 0**	**1 0 1 1**
Mandatory Fixed Part		
Destination Local Reference	0 0 0 0	0 0 0 1
Data	0 0 0 0	1 1 1 1

Dst Loc Ref	Msg Type
24	8

Expedited Data Acknowledgment Message (EA)	**0 0 0 0**	**1 1 0 0**
Mandatory Fixed Part		
Destination Local Reference	0 0 0 0	0 0 0 1

Reset Cause	Src Loc Ref	Dst Loc Ref	Msg Type
8	24	24	8

Reset Request Message (RSR)	**0 0 0 0**	**1 1 0 1**
Mandatory Fixed Part		
Destination Local Reference	0 0 0 0	0 0 0 1
Source Local Reference	0 0 0 0	0 0 1 0
Reset Cause	0 0 0 0	1 1 0 0

Src Loc Ref	Dst Loc Ref	Msg Type
24	24	8

Reset Confirmation Message (RSC)	**0 0 0 0**	**1 1 1 0**
Mandatory Fixed Part		
Destination Local Reference	0 0 0 0	0 0 0 1
Source Local Reference	0 0 0 0	0 0 1 0

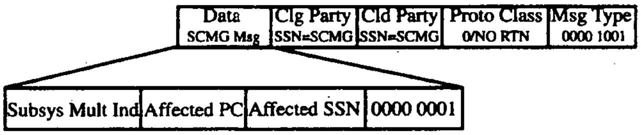
Err Cause	Dst Loc Ref	Msg Type
8	24	8

Error Message (ERR)	**0 0 0 0**	**1 1 1 1**
Mandatory Fixed Part		
Destination Local Reference	0 0 0 0	0 0 0 1
Error Cause	0 0 0 0	1 1 0 1

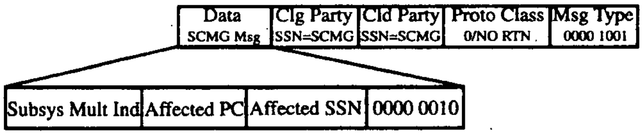
Credit	Seq/Segment	Proto Class	Src Loc Ref	Dst Loc Ref	Msg Type
8	16	8	24	24	8

Inactivity Test Message (IT)	**0 0 0 1**	**0 0 0 0**
Mandatory Fixed Part		
Destination Local Reference	0 0 0 0	0 0 0 1
Source Local Reference	0 0 0 0	0 0 1 0
Protocol Class	0 0 0 0	0 1 0 1
Sequencing/Segmenting	0 0 0 0	1 0 0 0
Credit	0 0 0 0	1 0 0 1

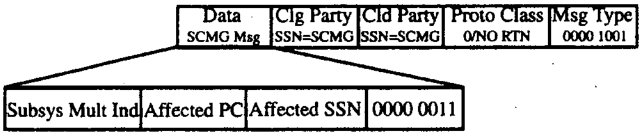
Data SCMG Msg	Clg Party SSN=SCMG	Cld Party SSN=SCMG	Proto Class 0/NO RTN	Msg Type 0000 1001
56	24	24	8	8



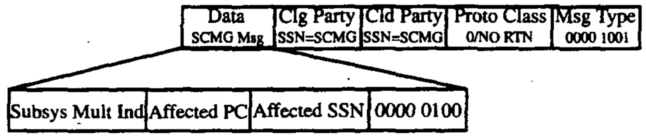
Subsystem-Allowed (SSA)	**0 0 0 0**	**0 0 0 1**
Affected Subsystem Number	0 0 0 0	0 0 0 1
Affected Point Code	0 0 0 0	0 0 1 0
Subsystem Multiplicity Indicator	0 0 0 0	0 0 1 1



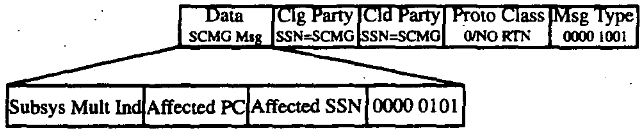
Subsystem-Prohibited (SSP)	**0 0 0 0**	**0 0 1 0**
Affected Subsystem Number	0 0 0 0	0 0 0 1
Affected Point Code	0 0 0 0	0 0 1 0
Subsystem Multiplicity Indicator	0 0 0 0	0 0 1 1



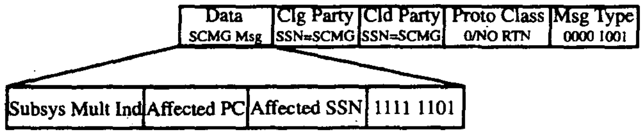
Subsystem-Status-Test (SST)	**0 0 0 0**	**0 0 1 1**
Affected Subsystem Number	0 0 0 0	0 0 0 1
Affected Point Code	0 0 0 0	0 0 1 0
Subsystem Multiplicity Indicator	0 0 0 0	0 0 1 1



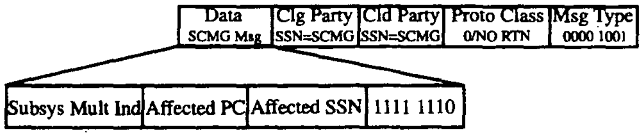
Subsystem-Out-of-Service-Request (SOR)	**0 0 0 0**	**0 1 0 0**
Affected Subsystem Number	0 0 0 0	0 0 0 1
Affected Point Code	0 0 0 0	0 0 1 0
Subsystem Multiplicity Indicator	0 0 0 0	0 0 1 1



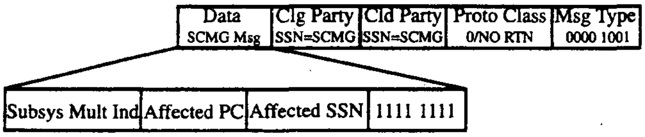
Subsystem-Out-of-Service-Grant (SOG)	**0 0 0 0**	**0 1 0 1**
Affected Subsystem Number	0 0 0 0	0 0 0 1
Affected Point Code	0 0 0 0	0 0 1 0
Subsystem Multiplicity Indicator	0 0 0 0	0 0 1 1



Subsystem-Backup-Routing (SBR)	**1 1 1 1**	**1 1 0 1**
Affected Subsystem Number	0 0 0 0	0 0 0 1
Affected Point Code	0 0 0 0	0 0 1 0
Subsystem Multiplicity Indicator	0 0 0 0	0 0 1 1



Subsystem-Normal-Routing (SNR)	**1 1 1 1**	**1 1 1 0**
Affected Subsystem Number	0 0 0 0	0 0 0 1
Affected Point Code	0 0 0 0	0 0 1 0
Subsystem Multiplicity Indicator	0 0 0 0	0 0 1 1



Subsystem-Routing-Status-Test (SRT)	**1**	**1**	**1**	**1**	**1**	**1**	**1**	**1**
Affected Subsystem Number	0	0	0	0	0	0	0	1
Affected Point Code	0	0	0	0	0	0	1	0
Subsystem Multiplicity Indicator	0	0	0	0	0	0	1	1

SCCP Parameters

The previous section only identified the parameters that are used by the various message types. The actual data found in each parameter was not indicated. In this section, we will look at each one of the individual parameters and explain the various data values possible within each particular parameter.

The parameters differ, depending on the message type they are used with. Some parameters are common and can be found in several message types. Others are specific to a particular message type and contain information relevant only to the specific message type.

This section will not attempt to identify which message types use each of these parameters because that has already been discussed. This section will explain the parameters in detail and provide the bit values of each one.

Called/Calling party address	**0**	**0**	**0**	**0**		**0**	**0**	**1**	**1**
Octet One									
Address Indicator	**H**	**G**	**F**	**E**		**D**	**C**	**B**	**A**
Subsystem Indicator									
Address contains subsystem number									1
Address does not contain subsystem number									0
Point Code Indicator									
Address contains point code								1	
Address does not contain point code								0	
Global Title Indicator	**H**	**G**	**F**	**E**		**D**	**C**	**B**	**A**
No global title included			0	0		0	1		
Global title includes translation type, numbering plan, and encoding			0	0		1	0		
Global title includes translation type only			0	0		1	0		
Not assigned in U.S. networks			0	0		1	1		
					to				
			1	0		0	0		
Spare			1	0		0	1		
					to				
			1	1		1	0		
Reserved for extension			1	1		1	1		

Routing Indicator	**H**	**G**	**F**	**E**	**D**	**C**	**B**	**A**
Route using global title only		0						
Route using point code/subsystem number		1						
National/International Indicator								
Address indicator coded as international	0							
Address indicator coded as national	1							

The subsystem indicator and the point code indicator are used to indicate whether or not one of these two entities is found in the address. In fact, this entire octet is used to indicate which elements of the address are found and which elements are to be used by routing.

The routing indicator is used to instruct the routing function as to which element of the address to use for routing the message. If the routing indicator is equal to 0, this indicates that global title translation is necessary. The receiving signaling transfer point, if it has global title capability, should then provide global title translation. If the STP does not have this functionality, it should route the message to its destination. Usually, the end STP will provide the global title translation before routing to an adjacent SCP for routing to the actual database or application.

Octet Two

Subsystem Number	**H**	**G**	**F**	**E**	**D**	**C**	**B**	**A**
Subsystem number unknown/not used	0	0	0	0	0	0	0	0
SCCP Management	0	0	0	0	0	0	0	1
Reserved	0	0	0	0	0	0	1	0
ISDN User Part	0	0	0	0	0	0	1	1
Operations, Maintenance, and Administration Part (OMAP)	0	0	0	0	0	1	0	0
Mobile Application Part (MAP)*	0	0	0	0	0	1	0	1
Home Location Register (HLR)*	0	0	0	0	0	1	1	0
Visited Location Register (VLR)*	0	0	0	0	0	1	1	1
Mobile Switching Center (MSC)*	0	0	0	0	1	0	0	0
Equipment Identification Register (EIR)*	0	0	0	0	1	0	0	1
Authentication Center*	0	0	0	0	1	0	1	1
Spare	0	0	0	0	1	0	1	1
					to			
	1	1	1	1	1	1	1	0
Reserved for expansion	1	1	1	1	1	1	1	1

The subsystem number identifies the application to which the message is being addressed or from where the message originated. New subsystems have been added recently in the Telcordia recommendations to include the various database functions found on the wireless network. The MAP is actually an application entity and uses the services of the TCAP and SCCP protocols to deliver control and signaling information through the network. MAP is used today in conjunction with IS-41 for seamless roaming and hand-off procedures within the wireless network.

*These are implemented in RBOC networks for the use of cellular internetworking.

The registers are actually databases that store information regarding wireless subscribers. The *home location register* (HLR) stores information regarding subscribers within the provider's calling area. The *visited location register* (VLR) provides information regarding those subscribers outside of their home calling area using roaming numbers. The cell sites constantly update the VLR as the wireless phone broadcasts location signals and, in turn, updates the HLR using SCCP and TCAP.

Octet Three

Point Code	**H**	**G**	**F**	**E**		**D**	**C**	**B**	**A**
Member Number	0	0	0	0		0	0	0	0
					to				
	1	1	1	1		1	1	1	1

Octet Four

Point Code	**H**	**G**	**F**	**E**		**D**	**C**	**B**	**A**
Cluster Number	0	0	0	0		0	0	0	0
					to				
	1	1	1	1		1	1	1	1

Octet Five

Point Code	**H**	**G**	**F**	**E**		**D**	**C**	**B**	**A**
Network Identification	0	0	0	0		0	0	0	0
					to				
	1	1	1	1		1	1	1	1

The point code is represented in the same format as the destination point code and the origination point code address found in the routing label. The same rules apply here as to the routing label (regarding the ranges allowed for point codes).

Octet Six

If the global title indicator in the address indicator is equal to 0001, the following format is used for the global title parameter:

Translation Type	**H**	**G**	**F**	**E**		**D**	**C**	**B**	**A**
Reserved	0	0	0	0		0	0	0	0
891 Telecommunications Credit Cards	0	0	0	0		0	0	0	1
14-Digit Calling Card	0	0	0	0		0	0	1	0
Cellular Nationwide Roaming Service	0	0	0	0		0	0	1	1
Global Title = Point Code	0	0	0	0		0	1	0	0
Calling Name Delivery	0	0	0	0		0	1	0	1
Call Management Application	0	0	0	0		0	1	1	0
Message Waiting Application	0	0	0	0		0	1	1	1
Internetwork Applications	0	0	0	0		1	0	0	0
					to				
	0	0	0	1		1	1	1	1
Network-Specific Applications	1	1	0	0		0	0	0	0
					to				
	1	1	1	1		1	0	0	0
Message Waiting Application	1	1	1	1		1	0	0	1

Network-Specific Applications	1	1	1	1	1	0	1	0
Call Management Application	1	1	1	1	1	0	1	1
14-Digit Calling Card Application	1	1	1	1	1	1	0	1
800 Number LIDB Application*	1	1	1	1	1	1	1	0

Translation types help route messages between networks to the proper functio within a signaling point. They are optional and network dependent. The Telcordia recommendations provide several predefined codes (shown previously), but every networ can assign its own translation types. The only rule here is that the translation typ name and the translation type number be used consistently in any one signaling poin and across the network.

Octet Seven

Encoding Scheme	**H**	**G**	**F**	**E**	**D**	**C**	**B**	**A**
Unknown					0	0	0	0
Binary Coded Decimal, odd number of digits					0	0	0	1
Binary Coded Decimal, even number of digits					0	0	1	0
Spare					0	0	1	1
						to		
					1	1	1	1

Numbering Plan	**H**	**G**	**F**	**E**	**D**	**C**	**B**	**A**
Unknown	0	0	0	0				
ISDN/Telephony Numbering Plan	0	0	0	1				
Reserved	0	0	1	0				
Data Numbering Plan	0	0	1	1				
Telex Numbering Plan	0	1	0	0				
Maritime Numbering Plan	0	1	0	1				
Land Mobile Numbering Plan	0	1	1	0				
ISDN/Mobile Numbering Plan	0	1	1	1				

The numbering plan identifies the format used for the global title. For example, i the United States, telephone numbers use the formula stipulated by the Nort American Numbering Plan. This is classified as the ISDN/Telephony numbering plar Wireless networks use the land mobile numbering plan.

The encoding scheme identifies the format used for the digits. Digits are always displayed in BCD format (unless the value is unknown). This parameter must always b of even length (eight-bit multiples), so an indication of whether the parameter represents an even or an odd number of digits is provided. In the event that an odd number of digits is represented, then the last four bits (bits EFGH) are padded with a zeros.

*This translation type has already been defined in many networks for 800 number translations. However, this number can be used for other network-specific applications. In the event that this translation type is being used for something other than 800 number translations, consideration towards internetworking should be taken.

Octet Eight and beyond
The actual address (which can be dialed digits or any other number) is divided into four-bit segments using BCD numbering. If the address includes an odd number of digits, the last four bits of the octet are set to all zeros, and the encoding scheme specifies an odd number of digits.

Address Signal	**H/D**	**G/C**	**F/B**	**E/A**
Digit 0	0	0	0	0
Digit 1	0	0	0	1
Digit 2	0	0	1	0
Digit 3	0	0	1	1
Digit 4	0	1	0	0
Digit 5	0	1	0	1
Digit 6	0	1	1	0
Digit 7	0	1	1	1
Digit 8	1	0	0	0
Digit 9	1	0	0	1
Spare	1	0	1	0
Code 11*	1	0	1	1
Code 12*	1	1	0	0
Spare	1	1	0	1
Spare	1	1	1	0
ST	1	1	1	1

If the global title indicator in the address indicator is equal to 0010, the format is the same as previously, except for the absence of the numbering plan and encoding scheme parameters.

The address signal provides the first digit in bits ABCD, the second digit in bits EFGH, and the third digit in the ABCD bits of the next octet. This pattern is repeated for every digit.

The called and calling party addresses provide adequate information for receivers of all SCCP messages. The originator of a message, providing enough information that a response can be returned to the correct signaling point, generates the calling party address.

The calling party address should include at least the point code and subsystem number of the originator or the global title if one exists. The objective is to provide enough information that will identify the originator even after global title translation. If an SCCP message is global title translated, then the origination point code may change to that of the signaling point providing the global title. But if the subsystem number of the originator and the global title of the originator are provided, then the responses can be returned to the true originator of the message.

Credit **0 0 0 0 1 0 0 1**

This parameter is a one-octet field, not counting the indicator. Following the indicator is a one-octet field indicating the number of messages that may be sent without

*Use of these codes is not fully defined to date and is under further study.

acknowledgment. This parameter is used only with connection-oriented services to allow for a sliding window size (flow control).

When an acknowledgment is sent, the sender of the acknowledgment will set the credit field to a particular value indicating the number of messages that the originator may send. This is based on the current status of the signaling point sending the acknowledgment.

The originator of the connection may then negotiate for a higher window size if it deems it necessary (based on the number of messages it has to send). If no negotiation is necessary, then the credit parameter is accepted and the originator begins sending data. An acknowledgment is then expected within the given window size.

The purpose of this parameter is to allow for more flexible flow control. In most protocols, the window size is a preset configuration, which may not be changed dynamically according to the current status of the node. With this parameter, the protocol may adjust its window size using this credit parameter when traffic increases and the signaling point needs more control over its resources.

Data 0 0 0 0 1 1 1 1

The data that is actually being carried by SCCP follows this indicator. The data may be TCAP, or it may be some other protocol. TCAP may be carrying actual application data, as is the case with MAP or IS-41. This means that there may be another layer of protocol before the actual data.

This should be taken into account when transmitting through the network using the services of SCCP and other transport protocols, such as TCAP, because each additional protocol will require some of the space in the data field. The maximum size for this field is 272 octets.

SCCP management also uses this field to transport management messages. SCCP management will always use the unitdata format.

Credit 0 0 0 0 0 0 0 1

Octets One, Two, and Three

This three-octet field (four, counting the indicator) consists of the indicator value and the three-octet number assigned by the destination. This is different from the source local reference, which is assigned by the local originator. The purpose of this parameter is to identify a connection within a signaling point.

Each entity in a connection assigns a number for reference. There should be a source (inbound) and destination (outbound) local reference number. This number is then used during connection-oriented calls for establishing, maintaining, and releasing connections.

The numbers are of local significance only; in other words, they have no meaning to other entities other than as the originator of the number. They are included in the message so that return messages can reference responses to the numbers.

End of Optional Parameters 0 0 0 0 0 0 0 0

This parameter is found at the end of every SCCP message that has optional parameters. If no optional parameters are used in the message, it is not used.

Error Cause	**0**	**0**	**0**	**0**		**1**	**1**	**0**	**1**
Local ref. a mismatch—unassigned destination LRN	0	0	0	0		0	0	0	0
Local ref. a mismatch—inconsistent source LRN	0	0	0	0		0	0	0	1
Point code mismatch	0	0	0	0		0	0	1	0
Service class mismatch	0	0	0	0		0	0	1	1
Unqualified	0	0	0	0		0	1	0	0
Spare	0	0	0	0		0	1	0	1
					to				
	1	1	1	1		1	1	1	1

The error cause parameter is used only in the error message. The error message is used with connection-oriented services. This is returned to the originator of a message that was received in error. The error listed is not caused by transmission problems, but by the originator and represents a protocol error rather than a data error.

Protocol Class	**0**	**0**	**0**	**0**	**0**	**1**	**0**	**1**
Class Indicator								
Class 0					0	0	0	0
Class 1					0	0	0	1
Class 2					0	0	1	0
Class 3					0	0	1	1
Message Handling (Classes 0 and 1 only)								
Discard message on error	0	0	0	0				
Spare	0	0	0	1				
		to						
	0	1	1	1				
Return message on error	1	0	0	0				
Spare	1	0	0	1				
		to						
	1	1	1	1				

The class indicator identifies the types of services to be provided by SCCP. Class 0 is basic connectionless service. In basic connectionless service, messages can be delivered out of sequence. Most messages use class 0 services on today's networks.

Class 1 provides sequenced connectionless services. The sequence numbering is provided through the sequencing/segmenting parameter. MTP has the ultimate responsibility of guaranteeing in-sequence delivery. Using the same route for all sequenced messages accomplishes this, while not using the bit rotation scheme for link load sharing. This ensures that all SCCP messages travel the same path, thus guaranteeing in-sequence delivery.

Messages that must be broken down into smaller messages are segmented and sent in multiple SCCP messages. These are then sent using class 1 SCCP. The messages are divided into equal lengths so that all SCCP segments are equal in size, or as close as possible to equal in size.

Class 2 services provide a basic connection-oriented delivery of messages. Basic connection-oriented services do not guarantee any specific level of service, other than establishing a connection and delivering data through the established connection. Once the data transmission is complete, the connection is released.

Class 3 services add flow control to the connection-oriented function. Flow control allows the starting and stopping of data flow according to resources and their congestion status (level-four resources, not level three).

Class 4 adds error recovery as well as flow control. Error recovery consists of the retransmission of SCCP messages that are received in error. This is also a connection-oriented service.

U.S. networks do not use any of the connection-oriented services of SCCP. In fact, this author does not know of any networks currently using connection-oriented services of SCCP.

Receive Sequence Number	**0**	**0**	**0**	**0**	**0**	**1**	**1**	**1**
Spare								0
Sequence Number	0	0	0	0	0	0	0	
				to				
	1	1	1	1	1	1	1	

This parameter indicates the next expected sequence to be received. It is sent in the backwards direction to indicate an acknowledgment of received messages. Unlike MTP, which uses the backward sequence number as an acknowledgment, this indicates the next sequence number that SCCP expects to receive. MTP indicates the last received sequence.

Refusal Cause	**0**	**0**	**0**	**0**	**1**	**1**	**1**	**0**
End user originated	0	0	0	0	0	0	0	0
End user congestion	0	0	0	0	0	0	0	1
End user failure	0	0	0	0	0	0	1	0
SCCP user originated	0	0	0	0	0	0	1	1
Destination address unknown	0	0	0	0	0	1	0	0
Destination inaccessible	0	0	0	0	0	1	0	1
Network resource—QoS available/permanent	0	0	0	0	0	1	1	0
Network resource—QoS not available/transient	0	0	0	0	0	1	1	1
Access failure	0	0	0	0	1	0	0	0
Access congestion	0	0	0	0	1	0	0	1
Subsystem failure	0	0	0	0	1	0	1	0
Subsystem congestion	0	0	0	0	1	0	1	1
Expiration of the connection establishment timer	0	0	0	0	1	1	1	0
Inconsistent user data	0	0	0	0	1	1	0	1
Not obtainable	0	0	0	0	1	1	1	0
Unqualified	0	0	0	0	1	1	1	1
Spare	0	0	0	1	0	0	0	0
				to				
	1	1	1	1	1	1	1	1

The refusal cause parameter is found in the connection refused message. It indicates the reason that a connection request has been denied. This is used only with the connection-oriented class of messages.

The value "network resource—QoS not available" indicates that the requested QoS could not be provided either on a permanent or temporary condition.

Release Cause	**0**	**0**	**0**	**0**		**1**	**0**	**1**	**0**
End user originated	0	0	0	0		0	0	0	0
End user busy	0	0	0	0		0	0	0	1
End user failure	0	0	0	0		0	0	1	0
SCCP user originated	0	0	0	0		0	0	1	1
Remote procedure error	0	0	0	0		0	1	0	0
Inconsistent connection data	0	0	0	0		0	1	0	1
Access failure	0	0	0	0		0	1	1	0
Access congestion	0	0	0	0		0	1	1	1
Subsystem failure	0	0	0	0		1	0	0	0
Subsystem congestion	0	0	0	0		1	0	0	1
Network failure	0	0	0	0		1	0	1	0
Network congestion	0	0	0	0		1	0	1	1
Expiration of reset timer	0	0	0	0		1	1	0	0
Expiration of receive inactivity timer	0	0	0	0		1	1	0	1
Not obtainable	0	0	0	0		1	1	1	0
Unqualified	0	0	0	0		1	1	1	1
Spare	0	0	0	1		0	0	0	0
					to				
	1	1	1	1		1	1	1	1

The release cause parameter is found in the released message and is used to indicate the reason for releasing a specific connection. Keep in mind that these are logical connections and have nothing to do with any voice circuits or any other physical connections within the PSTN.

The release cause is found only when using connection-oriented services and is only used in the released message. All other release or return messages use other specific parameters. Although there may be a lot of similarities between these various parameters, they serve a significantly different purpose.

The user indicated in the first few cause codes refers to the application entity, which may be TCAP, or a higher-level user, such as MAP.

Reset Cause	**0**	**0**	**0**	**0**		**1**	**1**	**0**	**0**
End user originated	0	0	0	0		0	0	0	0
SCCP user originated	0	0	0	0		0	0	0	1
Message out of order—incorrect P(s)	0	0	0	0		0	0	1	0
Message out of order—incorrect P(r)	0	0	0	0		0	0	1	1
Remote procedure error—message out of window	0	0	0	0		0	1	0	0
Remote procedure—incorrect P(s) after reinit	0	0	0	0		0	1	0	1
Remote procedure error—general	0	0	0	0		0	1	1	0
Remote end user operational	0	0	0	0		0	1	1	1
Network operational	0	0	0	0		1	0	0	0
Access operational	0	0	0	0		1	0	0	1
Network congestion	0	0	0	0		1	0	1	0
Not obtainable	0	0	0	0		1	0	1	1
Unqualified	0	0	0	0		1	1	0	0
Spare	0	0	0	0		1	1	0	1
					to				
	1	1	1	1		1	1	1	1

This parameter is found only in the reset request message, which is used for connection-oriented services. The parameter provides the reason for requesting the reset of a virtual connection. The end user is the application entity, such as the MAP that is using the services of SCCP. If the request is successful, the connection is reset and communications are reestablished.

Return Cause	**0**	**0**	**0**	**0**		**1**	**0**	**1**	**1**
No translation for an address of such nature	0	0	0	0		0	0	0	0
No translation for this specific address	0	0	0	0		0	0	0	1
Subsystem congestion	0	0	0	0		0	0	1	0
Subsystem failure	0	0	0	0		0	0	1	1
Unequipped user	0	0	0	0		0	1	0	0
Network failure	0	0	0	0		0	1	0	1
Network congestion	0	0	0	0		0	1	1	0
Unqualified	0	0	0	0		0	1	1	1
SCCP hop counter violation	0	0	0	0		1	0	0	0
*Error in message transport	0	0	0	0		1	0	0	1
*Error in local processing	0	0	0	0		1	0	1	0
*Destination cannot perform reassembly	0	0	0	0		1	0	1	1
Spare	0	0	0	0		1	1	0	0
					to				
	1	1	1	1		0	1	1	1
Invalid ISNI routing request	1	1	1	1		0	1	1	1
Invalid ISNI routing request	1	1	1	1		1	0	0	0
Unauthorized message	1	1	1	1		1	0	1	0
*Message incompatibility	1	1	1	1		1	0	1	1
*Cannot perform ISNI constrained routing	1	1	1	1		1	1	0	0
*Redundant ISNI constrained routing information	1	1	1	1		1	1	0	1
*Unable to perform ISNI identification	1	1	1	1		1	1	1	0
Reserved for extension	1	1	1	1		1	1	1	1

This parameter is found only in class 0 and class 1 messages when connectionless services are provided. The purpose is to identify the reason that a particular SCCP message was returned to the originator.

When an SCCP message is received in error, or any one of the previous events occurs, the original SCCP message is returned with its data to the originator. The UDTS or XUDTS message structure is used to return the data and the return cause to the message originator. It is then up to the originator to determine a plan of action: either retransmit or abandon the transaction.

Segmentation	**0**	**0**	**0**	**1**	**0**	**0**	**0**	**0**
Octet One								
Remaining segments	**H**	**G**	**F**	**E**	**D**	**C**	**B**	**A**
Last segment					0	0	0	0
One segment left					0	0	0	1

*Applies only to an XUDTS message.

Two segments left					0	0	1	0
Three segments left					0	0	1	1
Four segments left					0	1	0	0
Five segments left					0	1	0	1
Six segments left					0	1	1	0
Seven segments left					0	1	1	1
Eight segments left					1	0	0	0
Nine segments left					1	0	0	1
Ten segments left					1	0	1	0
Eleven segments left					1	0	1	1
Twelve segments left					1	1	0	0
Thirteen segments left					1	1	0	1
Fourteen segments left					1	1	1	0
Fifteen segments left					1	1	1	1
Spare			0	0				

In-Sequence Delivery Option (ISDO)	**H**	**G**	**F**	**E**	**D**	**C**	**B**	**A**
In-sequence delivery		1						
Not in-sequence delivery		0						
First bit								
First segment	1							
All other segments	0							

The remaining segments parameter identifies how many segments are to follow that are associated with this message. This is used whenever the data to be sent by SCCP is larger than the 255 octets, and the data is broken into smaller segments for transmission. The segments are as close to equal length as possible.

In addition to the remaining segments parameter, an indicator is provided that instructs the originator and any intermediate signaling points whether or not in-sequence delivery is to be used. The first segment is always sent with the first bit parameter set to 1.

In-sequence delivery is used with both connection-oriented and connectionless services. This parameter is used only with class 1 messages and is part of the extended unitdata and the unitdata SCCP messages.

Segmenting/Reassembling	**0**	**0**	**0**	**0**	**0**	**1**	**1**	**0**
No more data								0
More data								1
Spare	0	0	0	0	0	0	0	

This parameter only shows whether additional data or segments are to follow. No other information is provided. It is found in the DT1 message only, which is used in a connection-oriented data transfer. The remaining seven bits in this parameter are currently reserved for future implementation.

Sequencing/Segmenting	**0**	**0**	**0**	**0**	**1**	**0**	**0**	**0**
Octet One								
Spare								0

Sending Sequence Number—P(s)	0	0	0	0		0	0	0	
					to				
	1	1	1	1		1	1	1	
Octet Two									
More Data Indicator	**H**	**G**	**F**	**E**		**D**	**C**	**B**	**A**
No more data									0
More data									1
Received Sequence Number—P(r)									
	0	0	0	0		0	0	0	
					to				
	1	1	1	1		1	1	1	

The first octet of this parameter is used to indicate the sending sequence number. This is not the same as what we discussed with the sequence number parameter earlier. This parameter is not used with unitdata messages (connectionless). This parameter is used with connection-oriented classes, DT1, and DT2 messages.

The second octet provides the received sequence number, which is the same as an acknowledgment. If there is to be another SCCP message carrying additional data associated with this particular segment, then the more data indicator is used. This is only needed when the data has been divided among several SCCP segments.

	H	**G**	**F**	**E**	**D**	**C**	**B**	**A**
Intermediate Network Selection	**1**	**1**	**1**	**1**	**1**	**0**	**0**	**1**
Octet One								
Mark for Identification Indicator								
Do not identify network								0
Identify network								1
ISNI Routing Indicator								
Neither constrained nor suggested ISNI routing						0	0	
Constrained ISNI routing						0	1	
Reserved for suggested routing						1	0	
Spare						1	1	
Reserved for expansion of the IRI field					X			
Type Indicator								
Type zero ISNI parameter format				0				
Type one ISNI parameter format				1				
Counter	X	X	X					

The *Intermediate Network Selection* (INS) parameter can be used for the routing of SCCP messages between specific networks. The network identifier fields are used to indicate the network ID and cluster ID as pointers for routing instructions. The SCCP message can be directed to these addresses for further addressing and routing. For small networks, the network cluster field can be defaulted to all zeroes.

Chapter

8

Overview of TCAP

The *Transaction Capabilities Application Part* (TCAP) is designed for noncircuit-related messages. These messages are destined for database entities as well as actual end office switches. The TCAP protocol provides a means for the reliable transfer of information from one application at a switch location to another application within another network entity. To understand this, it is probably best to look at some of the actual applications and problems that we face in the telephone network today.

The first usage of the TCAP protocol was 800 number translation. An 800 number cannot be routed through the telephone network, because the area code 800 does not specify any particular exchange. To overcome this problem, the number must be converted into a routable number. This requires a database.

The database for 800 numbers provides a routing number that the local office can then use to route the call through the *Public Switched Telephone Network* (PSTN). This database is usually centrally located within the service provider's network. It does not make good sense to place this database in too many multiple locations, since that would make maintenance of the database more difficult.

The problem with centralized databases is providing access to them. All switches in the network must be able to access the database and retrieve the routing number for the 800 numbers used on their network. To compound the problem, in today's network, all 800 numbers must be routable by all carriers. This means that no matter which telephone company "owns" the 800 number, all other telephone companies must be able to access the proper database and retrieve the routing number for that 800 number. This is known as a transportable 800 number.

Making 800 numbers transportable also allows subscribers to keep the 800 numbers they have had provided by another carrier, even when they change carriers. Previous to the transportability ruling, if subscribers changed carriers, they had to surrender their 800 number and obtain a new number from the new carrier. This is no longer the case.

The TCAP protocol provides the parameters and services to maintain a dialog with a database. The protocol contains message types that are used by the *Service Control Point* (SCP) to query a database for specific information. This information is then carried back to the requesting central office switch using the same TCAP protocol. In short, TCAP messages are designed for accessing either a database or another switch, and either retrieving information or invoking features using its parameters and message types.

Wireless networks have the same types of needs, since they are very dependent on databases and the remote control of switch features. The wireless networks have previously used proprietary networks, prohibiting the capability to access remote databases in other networks. It is for this reason that the wireless industry has begun deploying *Signaling System 7* (SS7).

TCAP provides the mechanism for transferring information from one switch to another, even if they are a substantial distance apart. The information is not related to any one circuit (such as in an *ISDN User Part* [ISUP] message), and the information must be transferred through the network using end-to-end signaling.

ISUP messages do not use end-to-end signaling and must follow the same path used to establish the circuit connection. This means the message must be passed along from one exchange to another, with intermediate *Signaling Transfer Points* (STPs) through-switching the ISUP messages to the next exchange. This is one of the fundamental differences between TCAP and ISUP.

Another difference between the two protocols is the transport used. ISUP uses the *Message Transfer Part* (MTP) for routing the message from one exchange to another. The MTP protocol does not support end-to-end signaling, so TCAP must use an additional protocol as a transport. The *Signaling Connection Control Part* (SCCP) protocol is used with MTP to route messages end to end.

The SCCP protocol provides the additional controls needed when passing messages from end to end. However, MTP must also be used to provide the routing functions from one node to the next. The MTP also provides the basic error detection and correction needed for a reliable message transfer.

Now that we have identified the mechanism used for through-switching informational messages from one exchange to another, we can begin looking at specific applications. Many applications exist within the telephone company network. Many of these applications have not even been developed yet. We will talk about both present and future applications.

We have already discussed the use of TCAP for accessing a database (using the 800 number scenario). Many other databases are used in the telephone network besides the 800 database. Every telephone number has records associated with it. These records identify who the subscriber is and what types of services they have subscribed to.

These *Line Information Databases* (LIDBs) belong to the individual telephone companies that provide services to their subscribers. For this discussion, we will use the call forwarding feature as an example. Call forwarding enables subscribers to forward their phone calls to other subscriber telephone numbers until the call forwarding is canceled. All calls to the subscriber telephone number are then redirected to the forwarded number.

This feature requires the use of the LIDBs to verify that the subscriber is allowed to use this feature. When a subscriber invokes the forwarding feature, the information about where the call is forwarded to is stored in the end switch. The database is used when the feature is accessed (by dialing some sort of feature code). The database verifies that the subscriber is allowed to use this feature.

This is probably the simplest of applications. Let's look at a more sophisticated function of TCAP. The TCAP protocol also provides the mechanism to access other remote switches and activate features within that switch. The switch must have the feature capability already; TCAP only invokes the feature remotely.

Later, we talk about a scenario using another feature called *automatic callback*. In this feature, when a subscriber dials a number that is busy, the subscriber can enter a feature code and hang up. When the dialed number becomes available, the local exchange notifies the caller's local switch by sending a TCAP message. This TCAP message enables the local switch to ring the phone of the caller. The distant switch has reserved the called party's line so no other phone calls can be routed to it.

When the calling party answers the phone, normal call setup procedures are used to establish the connection between the two exchanges. TCAP, in this case, serves as an alerting mechanism, sending an informational message (not circuit-related) to another entity within the network. This can be extended to many other types of applications where remote invocation is an option.

In the wireless network, TCAP has become the solution to roaming. Prior to the deployment of SS7 in the wireless network, when wireless subscribers took their phones to other areas serviced by other wireless providers, they would have to call ahead and obtain roaming numbers. The roaming number was only good for a specific geographic area (a *regional service area* [RSA]).

When the roaming number was dialed, the wireless network immediately knew how to route the call, because the roaming number could be routed like any other *Plain Old Telephone Service* (POTS) number. The problem was that roaming was not seamless and required user intervention.

Some wireless subscribers found themselves having to have two and three numbers for their wireless phone, depending on where they were. This defeats the purpose of having a mobile telephone and was the reason for the seamless roaming.

The missing element was the capability to update the network databases with the current location of the wireless telephone subscriber. This information is updated every few minutes by the cell site sending a message to the *mobile switching center* (MSC) identifying the mobile subscriber, and the cell site reporting the subscriber's presence.

An application entity called IS-41 provides procedures for updating databases on the status of wireless subscribers. Every wireless subscriber has a home database, called the *Home Location Register* (HLR), which keeps a record on where the wireless subscriber is located. This record is what gets updated every few minutes.

TCAP is used to carry these update messages from one database (the *Visitor Location Register* [VLR]) to the subscriber's HLR. When a call comes in for the subscriber, the call is routed to the home regional service area, which must then look at the HLR to determine how to connect the call to the subscriber. The HLR then provides the information (location and status) so the PSTN knows how to connect the call.

When the subscriber moves to another area, the TCAP protocol is used once again to update the HLR on the new location and cancel the subscriber's registration in the previous regional service area. This is completely transparent to the subscriber and allows wireless subscribers the freedom to move around the network without ever having to register with other service providers. The network keeps track of their locations the entire time the wireless phone is activated.

In the *Intelligent Network* (IN), TCAP is the protocol that will be used to invoke features in remote switches. As we discussed earlier with the automatic callback feature, TCAP enables features to be activated and deactivated remotely. In the IN, services will be activated and deactivated the same as features.

Services include high-speed data transmission or video circuit connections. TCAP will provide access to the database, which will activate these services for the subscriber. As we discussed in the first chapter, subscribers will activate and deactivate these services through a terminal located on their premises and connected to the SS7 network via a data link.

TCAP Functionality

The TCAP provides a way for end users in the SS7 network to access other end users on a peer-to-peer level. In the SS7 network, end users are seen as applications within the network entities.

The SS7 protocol provides a means for database access between signaling points as well as access to remote operations. The latter function is somewhat newer to the SS7 protocol stack and is slowly finding applications on the network. The *Advanced Intelligent Network* (AIN) will depend heavily on the capability of a signaling point to access another signaling point remotely and to invoke an operation or feature within the remote signaling point.

The wireless network also depends today on remote access to other signaling nodes. For example, when a wireless switching center needs to hand over control of a call to another switching center, signaling information regarding that call must be transferred to the remote switching center. SS7 provides the means to accomplish this task.

In the IN, features such as automatic callback require the capability of an end office switch to send a message to another end office switch concerning the status of a previously called number. SS7 also provides this capability.

The protocol used for these types of transactions is the TCAP. The TCAP protocol originally provided database access, yet the intention has always been there to provide the facilities within the protocol to invoke remote features and access remote applications within the signaling network.

Description of TCAP

An application uses an application process to communicate with the other entities in the network. The application process then communicates with the TCAP and other protocols to transfer the information across the network.

The function that enables applications to communicate with one another is a communications function called the *application service element* (ASE). The TCAP and the *Mobile Application Part* (MAP) are both examples of ASEs (see Figure 8.1).

An ASE provides the communications services for applications within any signaling point. These communications are peer-level communications. An application process acts as the coordinator of network services, such as ISDN call setup and mobile services. The application process can be found below the function of applications, but above the ASE.

An application can use more than one application service element for any one transaction. This is necessary to provide flexibility in the communications function, allowing the addressing of multiple entities within one message.

Another method of reaching several entities within one transaction is the use of subsystem numbers for addressing. The subsystem numbers found in the called and calling party addresses in SCCP are the addresses of applications. By using a subsystem number rather than a point code, the transaction message can be sent to multiple signaling points or can even be rerouted by network management in the event of a failure.

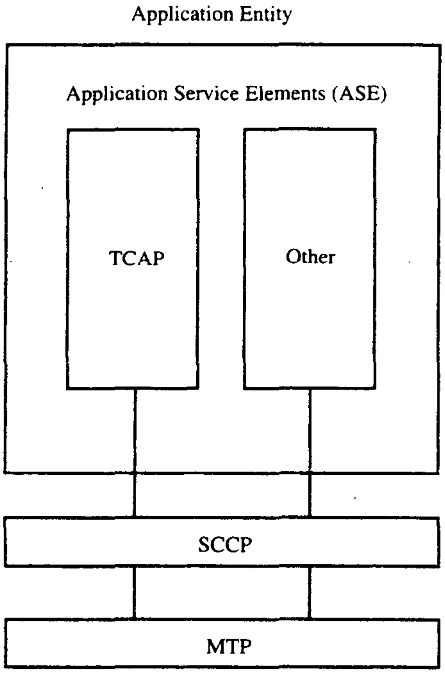


Figure 8.1 The ASE provides a communication interface to *application entities* (AEs) such as the MAP and the *Operations, Maintenance, and Administration Part* (OMAP).

One of the newest protocols to join TCAP is the MAP. This protocol is found in IS-41 wireless networks and is used for transporting roaming information and other signaling from one wireless network to another.

The TCAP and the MAP protocols are capable of transferring information and invoking operations within other entities. This is probably the most important aspect to understanding that these protocols not only transfer information from one entity to another, but also Invoke operations (or tasks) within a remote entity. An operation may be a particular function within the switching equipment or a database located at an end node.

The TCAP protocol is capable of both invoking an operation and returning the results of that operation using the services of the SCCP protocol. SCCP provides the network services, while the MTP provides the physical layer and data link layer functionality.

An application process is considered a function above layer seven of the *Open Systems Interconnection* (OSI) model. An example of an application may be the LIDB or OMAP. These applications are found in various signaling points, depending on the location of the signaling point in the network and its function within the network (STP, gateway STP, or translation STP).

In order for an application to send information and communicate to another application at a remote signaling point, a communications protocol must be used. This protocol should be able to address the application and invoke an operation or task within the application. For example, if a node needs to invoke a CLASS feature at a distant node for a call in progress, a message would need to be sent to the distant signaling point informing it of the operation needed. The receiving signaling point should be given enough information about the call to be able to invoke the specified CLASS feature.

The address of the message sent would be of the application providing the operation. This address is not an address used within the rest of the network, but references only the various applications supported within a network. The addressing scheme used for these applications is the subsystem number. The subsystem number does not have to be known at every node within the network. A global title can be used to route a message to the adjacent node of the application, at which point global title translation must provide the subsystem number and the point code of the application.

Today only connectionless services (datagrams) are supported. In some cases, a connection with a peer application may be necessary, in which case connection-oriented services must be employed to establish a connection with the application and maintain the connection until the originating signaling point has completed its transactions.

In the United States, support of connection-oriented services is not yet endorsed. Both *American National Standards Institute* (ANSI) and Bellcore standards mention connection-oriented services and even define their functions. Yet no application requiring connection-oriented services has emerged. Connectionless services are supported and are what U.S. networks use for communicating between application processes.

Because datagrams are used for information transfers between applications, some sort of reference must be used to associate multiple messages to a single transaction and multiple components to a single operation. Two forms of referencing are used: the transaction ID and the invoke ID.

A transaction ID is used as a reference within a dialog so the receiver can associate the received message with a transaction in progress. This transaction ID is significant only to the local receiver, but is sent to the remote as a reference to be used in any responses.

An invoke ID is used to identify Invoke components, which are used to invoke an operation within the entity. One transaction may consist of several invokes. The receiver of an Invoke is typically expected to respond with a Return Result, in which case a correlation ID is used to identify which invoke component the result is referencing. The correlation ID is a mirrored image of the invoke ID.

ASP Services

The *Application Service Part* (ASP) consists of the layers above the SCCP and below the TCAP. It provides the functions of layers four through six of the OSI model. These functions are not presently required in the SS7 network and are under further study; however, the *International Telecommunications Union—Telecommunication Standardization Sector* (ITU-TS) and ANSI standards do reference these as viable functions.

The lack of connection-oriented services in today's network is why the ASP is not currently needed. However, as the network matures and new technologies emerge, connection-oriented services will become a necessity for certain applications. This will force the need for the functions of these middle layers.

TCAP Message Structure

The message units within TCAP are partitioned into three portions: the *transaction portion*, the *component portion*, and the *dialog portion*. The transaction portion provides the information necessary for the signaling point to route the component information to its destination. Included in the transaction portion is the transaction ID, which is used as a reference for tracking all TCAP messages.

The component portion gets its information from the *operation protocol data unit* (OPDU) received from the application. The OPDU contains the primitives and parameters necessary to invoke an operation or request services from another entity (such as a database query).

The dialog portion is used to identify the version of the transaction being used. It also provides security information if encryption is used on the transaction. The dialog portion is an optional field within TCAP.

To fully understand the structure of these primitives, review the section in Chapter 7, "General Description of SCCP Functions" titled "Primitives." This section discusses the use of primitives and interfaces between SCCP and MTP. The structure of all primitives is the same throughout the layers of SS7. The interface between TCAP and the lower layers of the stack looks like this:

P-UNITDATA Generic Name Specific Name Parameter

The P- indicates the location of the interface. These primitives consist of the same generic names described in Chapter 7. The specific names used are described in the following paragraphs.

The *operation protocol data unit* (ODPU) consists of several types of data units. Each type indicates the type of service or operation to be invoked by the receiver. These OPDUs are then carried through the SS7 network in the form of a TCAP message. The component portion of the TCAP message is used to transport these OPDUs to their destinations. The following OPDUs are found in TCAP:

Invoke

Return Result

Return Error

Reject

The *Invoke* OPDU is used to request an operation from another application. For example, for the MAP to be able to notify a remote MSC to take control of a wireless call in progress, an Invoke OPDU must be created from the originating MSC. The Invoke OPDU is then used to form a TCAP message that, in turn, is transferred to the destination MSC through the SS7 network.

Upon receipt of the *Invoke* message, the remote MSC then initiates the operations required using the parameters provided in the component portion of the TCAP message. The Invoke message may require a return message to indicate successful completion of an operation. The *Return Result* OPDU is used by the destination application to indicate successful completion of an operation. This, in turn, creates the appropriate TCAP message, with the component portion carrying the Return Result and its parameters.

If the operation was unsuccessful, a *Return Error* or *Reject* OPDU is created at the destination MSC and is used to create the TCAP message to be returned to the originator of the Invoke message.

TCAP messages contain information elements that are used to convey the information being transported to the remote application. These information elements are always structured the same, regardless of their contents. The first field of every information element is the tag (referred to as an *identifier* in ANSI).

The tag indicates the type of information element being sent and, in doing so, enables the receiver to determine how the contents will be interpreted. The tag is one octet and is coded to describe the handling of the contents (see Figure 8.2).

Following the tag is the length field. The length field indicates the number of octets to be found in the contents field. The contents field is the location of the information be-

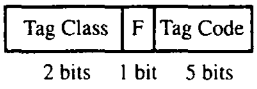


Figure 8.2 Tag class from ITU Q.733.

ing conveyed. This is not a fixed field, but a variable, depending on the type of components used.

The contents field consists of one or more components. In the case when more than one information element may exist in the contents field, the same structure as previously described is used (that is, tag, length, and contents) for each information element. However, the length field indicates the length of the individual information element, rather than the whole information element and all its components.

The tag is coded with a class, form, and tag code. The tag code may exceed the one-octet length in some cases, but is usually maintained within the first octet. The structure of a tag is shown in Figure 8.2.

Tag Class

The tag class is used to indicate whether this particular information element uses a common structure or if the contents are proprietary. This information is used by the receiver to determine how the message is to be interpreted and handled. As shown in Figure 8.2, the tag class uses bits HG. There are four values in this field:

Universal	0 0
Application-wide	0 1
Context-specific	1 0
Private use	1 1

The universal tag is one that is compliant with ITU-TS Recommendation X.208 and is used for all types of application entities. Universal tags are compatible with other recommendations and can be used with X.400 MHS as well as with other ITU-TS standards.

Application-wide indicates that the tag can be used with all applications within the SS7 standard. These tags do not conform to any other recommendations. Application-wide tags in the United States refer to the international standardized TCAP.

Context-specific indicates that the context of the received information is determined by the preceding component (information element). When more than one component is necessary, the first component received indicates how subsequent components are to be interpreted (if they carry a tag class of context-specific). The same component in a different message may be interpreted differently, depending on its usage within that message. Context-specific is used with components that are part of a series of components. These are referred to as *constructors*, meaning that each component builds off another.

The private use class indicates that the component is specific to a national standard (such as ANSI) or a private proprietary standard. ITU-TS recommendations indicate that this field is used for national or private use and does not define any of the components that use this tag class. The ANSI standards define specific national TCAP components not found in ITU-TS that are coded as private class.

Form

An information element may require subsequent information elements or may be a single value. The F bit is used to code each information element as either a primitive (single value) or constructor (multiple information elements). The values are shown as follows:

Primitive	0
Constructor	1

Tag Code

The code indicates the type of information element being sent. This field is expandable beyond the five-bit structure by using extension bits. In the event that an extension is needed, the first five bits (EDCBA) of the first octet in the identifier are set to 11111. This indicates that the tag code is found in the second and subsequent octets. If the second octet is extended, bit H is set to 1, indicating an extension. In each subsequent octet, bit H indicates whether there is an extension (1) or no extension (0).

In the following discussions, each information element and its contents will be described. The coding (national, private, primitive, and so on) will be provided within the description.

The TCAP message (see Figure 8.3) consists of three parts, as mentioned previously. The first part is labeled the transaction portion and provides the information necessary to identify the nature of the transaction. The transaction portion is a mandatory field for all TCAP messages.

The second part is labeled the dialog portion and is used to identify the version of the transaction as well as security information (in the event the transaction is encrypted). This part was recently added to TCAP.

The third part is labeled the component portion and is the part that contains the contents of the primitives sent down from the various applications. The component portion also contains the parameters that identify the specific details of the TCAP message. A dialog is maintained by sending a series of components in one or more TCAP messages and correlating those that are associated with a specific transaction and operation.

The protocol provides various parameters within the component portion that enable it to emulate a logical connection (hence, fulfilling the need for connection-oriented services without the need for connection-oriented SCCP).

Following is a description of all the fields within the transaction portion.

Package Type Identifiers

The package type identifier describes the type of transaction being sent (see Figure 8.4). A transaction may be a one-way transmittal, or it may require a two-way dialog. The package type determines these requirements. The following descriptions describe the function of each of these package types in a connectionless environment. Connection-

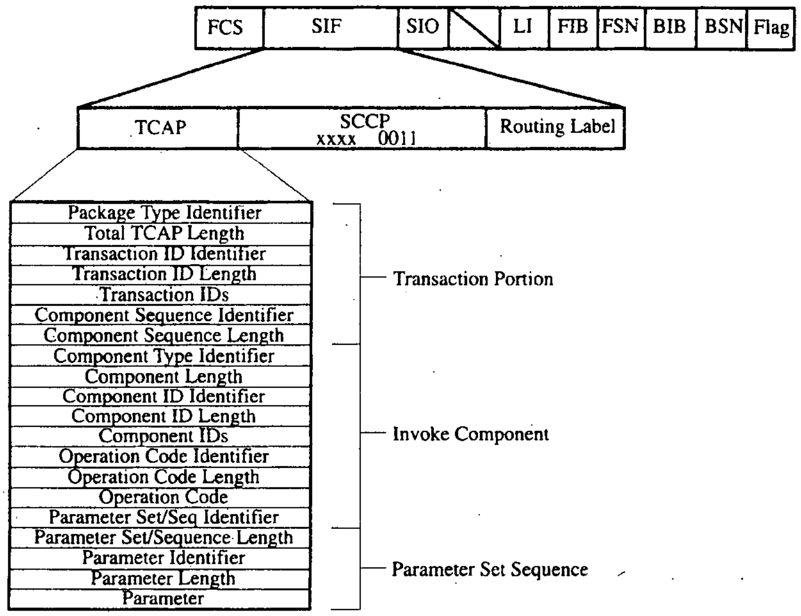


Figure 8.3 This figure shows the components of a TCAP message.

Package Type	H G F E	D C B A
Unidirectional	1 1 1 0	0 0 0 1
Query w/Permission	1 1 1 0	0 0 1 0
Query w/out Permission	1 1 1 0	0 0 1 1
Response	1 1 1 0	0 1 0 0
Conversation w/Permission	1 1 1 0	0 1 0 1
Conversation w/out Permission	1 1 1 0	0 1 1 0
Abort	1 1 0 1	0 1 1 0

Figure 8.4 These are the package types used in TCAP.

oriented services are discussed separately, since these services are not yet implemented in U.S. networks.

The following is a description of each package type identifier and its mandatory values. An asterisk in front of a parameter indicates that the field is mandatory but may be empty (set to all zeros).

Unidirectional

This type of TCAP is sent in one direction only and does not require a return message (or response). No transaction identifier is required for *unidirectional* messages.

Unidirectional messages are used for sending information to an application when a transaction does not need to be established. No correlation between multiple components is necessary.

The unidirectional TCAP consists of the following fields (an M indicates a mandatory field, while O indicates an optional field):

Package type identifier (M)

Total TCAP message length (M)

*Transaction identifier (null) (M)

*Transaction ID length (null) (M)

Dialog portion (O)

Component sequence identifier (M)

Component sequence length (M)

Query with Permission

A *Query* is used to access information stored within a database. It is also used to initiate the transaction and triggers the assignment of a transaction identifier. If a dialog is to take place, a Query will be followed by a *Conversation*. The dialog allows for information to be exchanged between the two applications without impediment.

The receiving signaling point of a *Query with Permission* is granted permission to end a transaction if it deems it necessary. Upon receipt of a Query with Permission, the receiving application must decide whether it wishes to establish a transaction and maintain a dialog (using Conversation components). In the event that it does not wish to establish a transaction, the *Response* component is sent. This releases the application from any transaction. No transaction ID is established.

If the receiving application does wish to establish a transaction, the Conversation component is sent (either with or without permission), and a dialog can be maintained between the two applications.

When an application needs to enter into a transaction with a remote application, and it does not anticipate sending additional components for the transaction, a Query with Permission is generated.

The Query with Permission consists of the following fields (an M indicates a mandatory field, while O indicates an optional field):

Package type identifier (M)

Total TCAP message length (M)

Transaction ID identifier (M)

Transaction ID length (M)

Originating transaction ID (M)

Dialog portion (O)

Component sequence identifier (M)

Comment sequence length (M)

Component sequence (M)

Query Without Permission

The *Query without Permission* is identical to the Query with Permission except for the permission granted. The receiver of this message is not granted permission to end the transaction. This, of course, does not include termination for network management reasons. As mentioned in the preceding paragraph, ending a transaction is accomplished by releasing the transaction ID.

When an application needs to enter into a transaction with a remote application, and the originator of the transaction anticipates additional components related to the transaction to be sent, a Query with Permission is generated. This prevents the remote application from ending the transaction before all the components have been received.

The Query Without Permission consists of the following mandatory parameters (an M indicates a mandatory field, while O indicates an optional field):

Package type identifier (M)

Total TCAP message length (M)

Transaction ID identifier (M)

Transaction ID length (M)

Originating transaction ID (M)

Dialog portion (O)

Component sequence length (M)

Component sequence identifier (M)

Component sequence (M)

Response

The *Response* is used to end a TCAP transaction. In the case of a Query, the Response is used to return the requested data. In the case of a dialog between two applications, the Response is the last transmission sent.

The Response consists of the following parameters (an M indicates a mandatory field, while O indicates an optional field):

Package type identifier (M)

Total TCAP message length (M)

Transaction ID identifier (M)

Transaction ID length (M)

Responding transaction ID (M)

Dialog portion (O)

Component sequence identifier (M)

Component sequence length (M)

Component sequence (M)

Conversation with Permission

This TCAP message is sent after a Query and is used to carry out a dialog between two applications. The package type Conversation continues between the two entities until the transaction is complete, at which point a Response is sent by either party.

When the Conversation component is sent, the transaction ID received from the originating node is duplicated and placed in the originating transaction ID field. Besides the originating transaction ID, the receiving node also creates a transaction ID corresponding to the transaction established locally. This transaction ID is significant only to the remote application and is placed in the responding transaction ID field.

Once the Conversation component is sent, all subsequent messages must contain one of the Conversation components (with or without permission). This component is then used to maintain a dialog between the two applications until the transaction is finished and one of the applications sends a Response component.

The Permission indicator grants the receiver of this message the ability to end the transaction (by releasing the transaction ID). Either party can end the transaction.

Permission is granted when an application has responded to all received components and does not anticipate the need to send further components related to a transaction. In essence, the application uses the permission indicator to ensure that the transaction is maintained until all the components related to a specific transaction can be sent. This prevents the remote application from ending the transaction prematurely.

If an application during a dialog determines the need to gain control over the release of a transaction (even if it previously relinquished control), then it may send a *Conversation Without Permission* even when it had previously relinquished control.

A transaction is ended by an application sending a Response component. The transaction can be ended even when permission is not granted in special circumstances (resource management intervention, for example).

The *Conversation with Permission* consists of the following parameters (an M indicates a mandatory field, while O indicates an optional field):

Package type identifier (M)

Total TCAP message length (M)

Transaction ID identifier (M)

Transaction ID length (M)

Originating transaction ID (M)

Responding transaction ID (M)

Dialog portion (O)

Component sequence identifier (M)

Component sequence length (M)

Component sequence (M)

Conversation Without Permission

This message is the same as the Conversation with Permission except for being able to end a transaction. As previously described, this component is sent to prevent the remote application from ending a transaction before the originating application has completed the transmission of all its components. It consists of the following parameters (an M indicates a mandatory field, while O indicates an optional field):

Package type identifier (M)

Total TCAP message length (M)

Transaction ID identifier (M)

Transaction ID length (M)

Originating transaction ID (M)

Responding transaction ID (M)

Dialog portion (O)

Component sequence identifier (M)

Component sequence length (M)

Component sequence (M)

P-Abort

A *P-Abort* is used when the originating entity must end a transaction. It is important to remember that when a transaction is aborted or ended under normal conditions, the application is responsible for the action. A transaction can be ended by the protocol or lower layers of the stack in a number of ways. At this level, the TCAP identifies why the abort was necessary (as determined by the application) and forwards the reason or causes to the remote entity.

The P-Abort consists of the following parameters (an M indicates a mandatory field, while O indicates an optional field):

Package type identifier (M)

Total TCAP message length (M)

Transaction ID identifier (M)

Transaction ID length (M)

Responding transaction ID (M)

Dialog portion (O)

P-Abort cause identifier (M)

P-Abort cause length (M)

P-Abort cause (M)

U-Abort

The *user abort* (U-Abort) is used when the user of TCAP aborts a transaction. It is used in the same way as the P-Abort, but contains slightly different information. The U-Abort consists of the following parameters (an M indicates a mandatory field, while O indicates an optional field):

Package type identifier (M)

Total TCAP message length (M)

Transaction ID identifier (M)

Transaction ID length (M)

Responding transaction ID (M)

Dialog portion (O)

U-Abort information identifier (M)

U-Abort information length (M)

U-Abort information (M)

Each of the package types identified here contains a number of additional parameters (as noted in the preceding descriptions). These parameters contain the specific information regarding the transaction taking place.

These package types are identified in the transaction portion. Following the transaction portion is the dialog portion. The dialog portion contains information regarding the version of the transaction and security information if any of the transaction is encrypted. The dialog portion is an optional part of TCAP. Following is a listing of each of the dialog portion fields:

Protocol version identifier

Protocol version length (O)

Application context identifier (O)

Application context length (O)

Application context name (O)

User information identifier (O)

User information length (O)

User information (O)

Security context identifier (O)

Security context length (O)

Security context (O)

Confidentiality identifier (O)

Confidentiality length (O)

Confidentiality information (O)

The application context fields are only used in unidirectional, query, user abort, and the first backward conversation or response TCAP messages.

Following the dialog portion is the component portion, which consists of one or more components. The component structures were described earlier as "information elements." Each information element may be a *primitive* (single value) or a *constructor* (multiple components).

The component types listed consist of the information elements described earlier. An identifier is used as a tag for an information element. The parameters for each of these fields are described later in the chapter. They are listed here as an introduction to the structure of a TCAP message. The following is a listing of each of the component types and the fields that are sent with each component.

All component fields are mandatory. Those with an asterisk indicate mandatory fields that can be empty (all bits set to zero, or null). The length indicators provide the number of octets for the fields immediately following, not including the length field itself. This and all other fields are explained further later in this chapter.

Invoke component

Component type identifier

Component length

*Component ID identifier

*Component ID length

*Component IDs

Operation code identifier

Operation code length

Operation code

*Parameter set/sequence identifier

*Parameter set/sequence length

*Parameter set/sequence

Return Result component

Component type identifier

Component length

*Component ID identifier

*Component ID length

*Component IDs

*Parameter set/sequence identifier

*Parameter set/sequence length

*Parameter set/sequence

Return Error component

Component type identifier

Component length

*Component ID identifier

*Component ID length

*Component IDs

Error code identifier

Error code length

Error code

*Parameter set/sequence identifier

*Parameter set/sequence length

*Parameter set/sequence

Reject component

Component type identifier

Component length

*Component ID identifier

*Component ID length

*Component IDs

Problem code identifier

Problem code length

Problem code

*Parameter set/sequence identifier

*Parameter set/sequence length

*Parameter set/sequence

Again, the two portions of the TCAP message are used in different ways. The transaction portion is used by the receiver of the message to determine which transaction this message is associated with (if one is in progress) or to begin a new transaction. The component portion is used by the receiver to invoke an operation at a remote application. The component portion may consist of one or more information elements providing a sequence of operations to be performed. Each of these information elements can be correlated through a correlation ID, which is used to associate multiple components to an operation.

Connectionless TCAP Functionality

The connectionless services deployed in today's networks provide for some connection-oriented emulation. However, there have been procedures defined for connection-oriented services in both the ANSI standards and Bellcore standards. This indicates that there may come a day when TCAP and SCCP will need to support full connection-oriented services.

Three levels of identification are provided for transactions and their operations. The first and highest level of reference is the transaction ID. The transaction ID is used when multiple transactions are sent to an application to correlate the received transaction with other transactions already in progress. But within a transaction may be components related to multiple operations.

The correlation ID is used to correlate multiple components to a component already received. The receiver of a TCAP transaction associates the transaction ID to an earlier transaction and then correlates the various components to operations already in progress based on the correlation ID. It is also the correlation ID that keeps multiple components associated with one another.

When an Invoke component is sent, it may or may not require a response. The response sent must reference the Invoke it is associated with. This is done through the invoke ID, which is then mirrored in the Return Result component's correlation ID. When a component is responding to an Invoke component, it must also be determined whether this is the last component or if additional components will be sent as a response. This is determined by the TCAP function using the Invoke (last/not last) and Return Result (last/not last) components.

You cannot have multiple operations using the same invoke ID. The invoke ID cannot be reassigned until all the components expected have been received and all Return Results have been sent. The application will then inform the application process to end the transaction, which releases all transaction IDs, correlation IDs, and invoke IDs associated with the transaction.

The application process provides the necessary data to TCAP for invoking an operation. When an Invoke is requested, the components and their parameters are provided to TCAP for inclusion in an Invoke message. There may be more than one transaction at one time, and each transaction may have multiple operations running concurrently. TCAP must be able to address these operations individually and as a group.

TCAP can send multiple components and address multiple operations in one TCAP transaction. This means that a single TCAP message may contain information for several operations that are not associated. The use of the correlation ID and transaction ID keeps the various parameters straight within a TCAP transaction.

Handover Procedures

An application can request the application process to send a component or multiple components to another remote application entity for processing. When this occurs, a *handover* must be generated by TCAP. The handover may be temporary or permanent.

The purpose of the handover is to send the information required of the remote application in order to perform the operations being requested of it. A single component may be all that is required, or several components may be required. In the case of a temporary handover, any responses by the new application are sent to the application that initiated the handover.

When a permanent handover occurs, the new application can be directed to send all responses directly to the originating application. In essence, the application is directed to assume control of the transactions and operations being requested. The application does not have any knowledge of the application that requested the handover. All transactions are treated as if they were addressed directly to the application.

The information required of the application is provided in an Invoke component, which possesses the handover operation parameter. For example, if a dialog is in progress between applications A and B, with B being the remote application and A being the originating application, B would send an Invoke component to application C with the temporary handover operation. Besides the temporary handover operation, the transaction ID of B, the SCCP calling party address of B, and the package type that application C should use in its messages to application A are all provided as parameters of the Invoke component.

Recovery Procedures

In the event that an error occurs, TCAP invokes one of three levels of recovery procedures. The three levels of recovery correspond directly with the three types of errors that could occur. Errors can be classified as protocol errors, application errors, or end-user errors. Application errors are detected first, then application errors, and then end-user errors.

TCAP defines only the procedures used to recover from errors between TCAP and the application process. Any error procedures within the application process or the application are implementation dependent and beyond the scope of the SS7 standards. TCAP is responsible for reporting the errors to the application process, which, in turn, will report it to the application that determines the correct recovery procedure if the error is at the application level.

Protocol Errors

Protocol errors involve incorrect TCAP messages. Incorrect means the TCAP message contained package types or components that are invalid (or unrecognized), called for an operation that the application process did not recognize, or referenced a transaction ID or correlation ID that was not in progress.

Protocol errors can be detected by TCAP or the application process. It is reported to the remote application process using the *Reject* component. The Reject component must also provide the type of error (cause). This is sent to the remote application process that is then responsible for recovering from the error.

These types of errors are different from other protocol errors in the way recovery is invoked. In most protocols, when an error is detected by a remote entity, the remote entity discards the erred packet and requests a retransmission. In TCAP, the erred packet is still discarded, but a Return Result is sent back to the originator to inform them why the packet was rejected. It is then up to the originator of the erred message to determine if retransmission is necessary. If the application process decides to retransmit, the message is resent as if for the first time.

Application Errors

Application errors are errors involving the application process and indicate a violation of the application process procedures. These errors can also indicate common resources (such as recordings) that are unavailable.

An unexpected sequence of components and an unexpected data value are also types of errors that occur at the application process level. These errors mean that the application process was expecting components in an order different from how they were received (in comparison to a script or some other program). An unexpected data value means that data was received that does not match what should have been received for the received component.

A missing customer record and overdue reply are also listed as possible causes of application errors. These errors are reported using the *Return Error* component.

End-User Abnormalities

These errors are found last by the application process and indicate an error caused by the end user of the application. Two causes cited in the ANSI standard are caller abandonment and improper call response.

Caller abandonment indicates that a caller hung up before the transaction could be completed. This does not necessarily mean that an error occurred, but that the transaction could not be completed as normal.

Improper caller response indicates that a caller did not dial the proper information during a call where callers are asked by a recording to input some form of information (such as the calling card number when billing a call to a calling card).

Reject Component

The Reject component reports many of these errors. In the Reject component, the type of problem is identified and the problem itself is also identified. The type of problem is divided into categories that correspond to the various portions of the TCAP message structure. Errors are identified as transaction portion, general, Invoke component, Return Result component, or Return Error component.

Errors in the transaction portion are identity errors occurring within the transaction portion of the TCAP message. These include the package type and transaction ID. The errors are reported using the Reject component, and they report errors in the transaction portion of the message.

The types of errors that can occur in the transaction portion include an unrecognized package type, a badly structured transaction portion, an unrecognized transaction ID, a permission release problem, or an unavailable resource.

Unknown package types indicate that a package type was indicated but is not defined as received in the signaling network. In the case of an international network, this would indicate that a package type is not defined in the ITU-TS standards. The package type is the parameter that identifies what type of TCAP message is being sent and is used by the receiver to determine how to handle the received message.

A badly structured transaction portion indicates a problem with the encoding of the message. For example, the length may not be as indicated or may be in conflict with the expected length for the indicated package type. The *total TCAP message length* is used to indicate the length of the entire TCAP message.

An unrecognized transaction ID indicates that the transaction ID indicated in the message is not currently in progress. Either the transaction has been ended and the transaction ID released or the message has an incorrect transaction ID. The receiver of this message is responding to a previous Invoke or Conversation component that provided the transaction ID of the originator. This message could also indicate that the originating Invoke component may have had an incorrect originating transaction ID, which is mirrored and returned with any responses.

A *permission to release* problem is an error currently under study and is not presently defined. The only mention of this error is in the ANSI T1.114 standard. It does not appear in the Bellcore TR-NWT-000246 publication.

When applications require other resources, such as recordings, there may come an instance when those resources are not available. When this happens, the operation being requested is rejected. The originating application can then determine whether to try again later.

General problems are related to problems recognizing the component portion, or they indicate some problem with the component portion. This includes an unrecognized component type, an incorrect component portion, or a badly structured component portion. In all these cases, the component portion cannot be recognized and is rejected. These errors are not related to problems with the various components inside the component portion. These errors indicate a problem with the entire component portion itself.

Along with problems in the component portion, errors may occur within each component. The protocol is capable of reporting errors related to specific component types. Error reports for Invoke components, Return Result components, and Return Error components can be generated when a problem is detected with the component itself. A problem with one component does not reflect a problem with the entire transaction, only with the specified component. Therefore, the rest of the components that may exist within the component portion may be processed without error, unless they are all associated with the same operation, in which case the whole transaction is affected.

Invoke component errors include the receipt of duplicate invoke IDs, unrecognized operation codes, incorrect parameters, or unrecognized correlation IDs. Duplicate invoke IDs and correlation IDs indicate that these numbers have already been assigned for an operation or transaction already in progress and cannot be reused until they are released. An unrecognized operation code indicates that the received operation code is not presently defined, and an incorrect parameter indicates that a parameter other than what was expected was received.

Three types of errors are related to the Return Result component. When a component is received with a correlation ID that is not recognized (because it does not match any transactions in progress), an error is returned as an *unrecognized correlation ID.* An Invoke component that was not successful may result in the return of a result that is not expected (other than a success indication). In this event, an *unexpected return result* error is generated. If a parameter is undefined or unexpected (it does not match what should have been received for the type of component), an error of *incorrect parameter* is returned.

Errors related to the Return Error component include unrecognized correlation IDs, unexpected return errors, unrecognized errors, unexpected errors, and incorrect parameters. An unrecognized correlation ID indicates that the correlation ID received does not match any operation presently in progress.

An unexpected return error occurs when a Return Error component is received that does not report a failure of the invoked operation. A return error may also contain an unrecognized error, which is one not defined by the application process. If the returned error is not applicable to the invoked operation, then it is marked as unexpected. An incorrect parameter or unexpected parameter will also generate an error.

Return Error Component

This component is used to report a failure of an operation. It is also possible for this component to report the success or the failure of an operation. At any rate, the operation fails.

Included in this component are the error that occurred and the application process that was in error. If an invoke ID or a correlation ID was provided in either the Invoke component or the Return Result component, then it also is reflected in the Return Error component.

Return Result Component

This component reports the success of an operation. It also reports any end-user errors that may have occurred. The successful operation is identified, as well as parameters that identify the end-user error.

An end-user error does not necessarily cause an operation to fail. The invoked operation may be successful, but may have caused a problem within the application entity. This would cause the Return Result to report the end-user failure.

Problems relating to the transaction portion of the TCAP message follow a different procedure. Since the transaction portion is what the receiver uses to determine the handling of a received TCAP, any errors result in the application process not knowing how to handle the message. The usual procedure in this case is to discard the message. There are instances, however, when this is not favorable. Whenever enough information can be obtained from the message, some sort of error report should be returned to the user, indicating the type of error. The *Abort* component is used to report such errors.

The Abort component is used whenever any type of component is received where either the originating and/or responding transaction ID can be derived from the message. In all other instances, the TCAP is discarded, with no report to the user. Without the transaction ID, any report to the user would be fruitless.

Definition of TCAP Parameters

The preceding section described the various functions of the TCAP message and its components. This section identifies the values of these components and their parameters. The values given are derived from the Bellcore TR-NWT-000246 publication, Issue 2, Revision 2, December 1992.

Transaction Portion

The transaction portion (see Figure 8.5) identifies whether or not the component portion consists of a single transaction or multiple transactions, and it alerts the receiving application as to how to handle the message. Important to the receiver is the type of structure used within TCAP. This is determined by the package type. Also significant is the length of the TCAP message in its entirety.

The transaction portion provides the necessary data for the receiver to be able to determine the nature of the message and its relation to any existing transactions in progress. The following are the package types and their bit values used in the transaction portion to identify the type of TCAP message being presented.

Package Type Identifiers	**H**	**G**	**F**	**E**	**D**	**C**	**B**	**A**
Unidirectional	1	1	1	0	0	0	0	1
Query with Permission	1	1	1	0	0	0	1	0
Query Without Permission	1	1	1	0	0	0	1	1
Response	1	1	1	0	0	1	0	0
Conversation with Permission	1	1	1	0	0	1	0	1
Conversation Without Permission	1	1	1	0	0	1	1	0
Abort	1	1	0	1	0	1	1	0

Total TCAP Message Length. This indicates the total length of the TCAP message (including all components and parameters). Within each component in the component message is a length indicator to indicate the length of that individual component. All variable parameters also include length indicators.

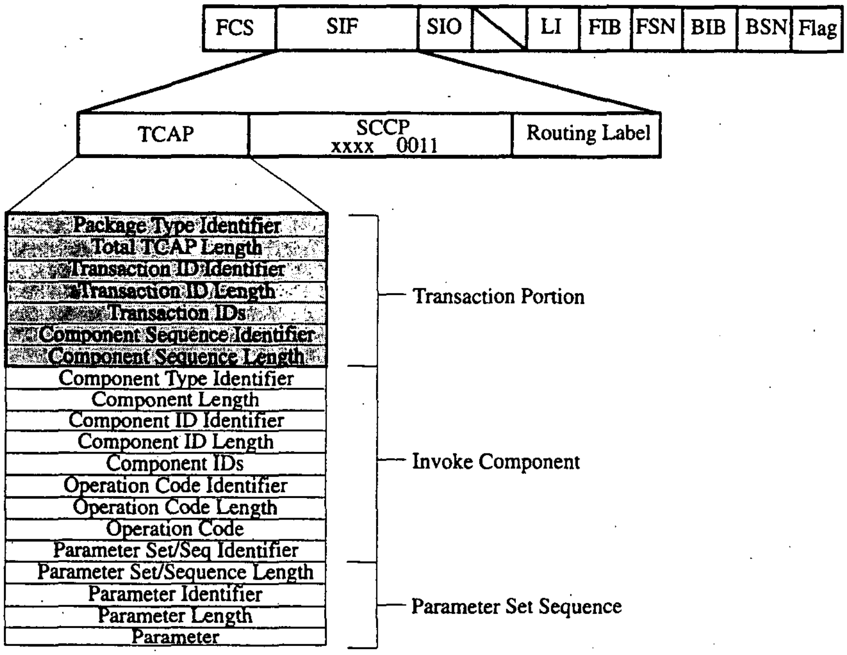


Figure 8.5 This figure identifies the components of the transaction portion.

Transaction ID Identifier 1 1 0 0 0 1 1 1. This indicates that a transaction ID is present and will follow this field. It is coded as national and primitive (refer to the discussion in the previous section describing the encoding used for national and international messages). As mentioned in the discussion at the beginning of this chapter, the transaction ID is of local significance only. This identifier only suggests the presence of the transaction ID and does not represent the ID itself.

Transaction ID Length. This indicates the length of the transaction ID, including both the originating and responding transaction ID, if applicable. If the package type is unidirectional, the length will be equal to zero. Only the unidirectional package type will have a length of zero, because a transaction ID is not assigned to this package type.

If the package type is one that contains only the originating transaction ID, the length will be equal to four octets. The originating ID is sent to the remote application to be used as a reference in any response.

If the remote application is sending a component that will require a response of some nature, then the originating transaction ID and a response ID will be sent. The originating transaction ID is sent to associate the response to the transaction of the originator. The response transaction ID is sent to be used by the originator when it sends a response back to the remote application. These dual transaction IDs occur when there is a dialog between two applications, and Invoke components are sent from one entity to another.

Transaction IDs. The number of transaction IDs given depends on the package type. The following indicates when multiple transaction IDs are provided.

Package Type	Originating ID	Responding ID
Unidirectional	No	No
Query with Permission	Yes	No
Query Without Permission	Yes	No
Response	No	Yes
Conversation with Permission	Yes	Yes
Conversation Without Permission	Yes	Yes
Abort	No	Yes

Originating Transaction ID. This identifies the transaction ID assigned by the originator of the message. The length is four octets, and it is the first field when multiple transaction IDs are presented. As previously described, this is sent by the originator to the remote application to be used as a reference in any responses.

Responding Transaction ID. This identifies the transaction ID generated by the responding application. This is independent of the originating transaction ID and is significant to the responding application only. This field is used when the responding application expects a response from the originator of an Invoke operation component. This is a four-octet field and follows the originating transaction ID.

P-Abort Cause Identifier 1 1 0 1 0 1 1 1. This identifier indicates that the Abort cause value follows. This identifier is coded as national and primitive. An Abort cause is sent when a transaction is aborted before it had time to get completed. An Abort, as described earlier, indicates a problem within the application entity rather than a protocol error.

P-Abort Cause Length. The length does not include this field and the identifier field; it includes only the cause field. The cause field is always one octet in length; thus, this field will always carry a value of one octet.

P-Abort Cause	H	G	F	E	D	C	B	A
Unrecognized package type	0	0	0	0	0	0	0	0
Incorrect transaction portion	0	0	0	0	0	0	0	1
Badly structured transaction portion	0	0	0	0	0	0	1	0
Unrecognized transaction ID	0	0	0	0	0	0	1	1
Permission to release problem	0	0	0	0	0	1	0	0
Resource unavailable	0	0	0	0	0	1	0	1

The cause code informs the receiver of an abort and what caused the application process at the remote end to suddenly abort a transaction in progress. An unrecognized package type indicates that the package type received is not currently defined at that application entity.

An incorrect transaction portion indicates that an identifier in the transaction portion was unexpected (not what should have been received in comparison with the rest of the transaction portion and the transaction in progress).

A badly structured transaction portion indicates that there is a problem with the encoding of the transaction portion. This could mean that the transaction portion is not of the correct length for the indicated identifiers, or that there are missing fields.

An unrecognized transaction ID occurs when a TCAP is received with a transaction ID that does not reflect a transaction currently in progress. The receiving application does not know to which transaction and operations to associate the message; therefore, it will abort the transaction and return the cause to the sender.

The cause, a *permission to release* problem, is not currently defined and can be used for further study. The last cause code, *resource unavailable*, is returned when a transaction is aborted because the resources necessary to perform the operations specified (such as recordings or databases) are not available.

User Abort Information Identifier 1 1 0 1 1 0 0 0. The identifier indicates that the user abort information field is to follow. This identifier is coded as national and primitive. A user can initiate an abort when an unexpected event occurs, such as when the caller hangs up before the transaction can be completed. The user in this message is not the calling or called party, but an application entity.

User Abort Information Length. The length does not include this field and the identifier field; it includes only the user abort information field. The information field is a variable field.

User Abort Information. This variable field is used by TCAP to provide information regarding a user abort. The user abort indicates the reason why a user (application) aborted a transaction before it could be completed. The abort codes are not defined in ANSI or Bellcore, because these are implementation-specific codes defined by the various manufacturers.

Component Sequence Identifier 1 1 1 0 1 0 0 0. This field is used to notify the receiver that a sequence of components will follow this field. This identifier is coded as national and constructor. The component sequence identifier does not identify how many components are to follow, only that they will follow. This is really used as a header to the component portion, and all TCAP messages will contain this identifier.

Component Sequence Length. The length of the entire component portion is indicated in this field. The length does not include the length field itself or the identifier before it.

Dialog Portion

The dialog portion is an optional part of TCAP. It contains information regarding security, encryption, and the protocol version of the data contained in the user part of TCAP. The following are the parameters of the dialog portion:

Dialog Portion	H	G	F	E	D	C	B	A
Dialog portion	1	1	1	1	1	0	0	1
Protocol version	1	1	0	1	1	0	1	0
Integer application context	1	1	0	1	1	0	1	1
Object identifier application context	1	1	0	1	1	1	0	0
User information	1	1	1	0	1	0	0	0
Integer security context	1	0	0	0	0	0	0	0
Object identifier security context	1	0	0	0	0	0	0	1
Confidentiality	1	0	1	0	0	0	1	0
Integer confidentiality algorithm	1	0	0	0	0	0	0	0
Object identifier confidentiality algorithm	1	0	0	0	1	0	0	1

The *dialog portion* identifier indicates that a dialog portion is included in the TCAP message. This is an optional field used when the user data contained in the TCAP message is secure data. Following the dialog portion identifier is the dialog portion length field. This provides the length (in octets) of the entire dialog portion of the TCAP message, not including this length field or the dialog portion identifier.

The *protocol version identifier* indicates that the protocol version follows. The protocol version length indicates the length of just the protocol version part of the dialog portion. The protocol version field indicates which version of the T1.114 protocol is being sent in the TCAP message. This enables nodes to determine which TCAP version of the T1.114 protocol is being sent in the TCAP message. This also enables nodes to determine which TCAP version (for example, ANSI T1.114, 1997) is being used to encode the SS7 message.

The *application context identifier* indicates that an application context name follows. This is used by the applications at the receiving nodes. It is accompanied by user data in the user information fields of the dialog portion. The identifier is followed by the application context length, which provides the length of the application context name only.

The *user information identifier* indicates that user information is provided in the dialog portion. The user information length follows, indicating the length of the user information portion only, not counting the user information identifier or the length fields.

The *security context identifier* indicates that security information is contained in the dialog portion. It is followed by the security context length field. The security context provides information regarding the encryption of user information when encryption is used. This enables the secure transmission of user information when TCAP is being used to access databases that may have sensitive data.

The last part of the dialog portion is the confidentiality fields. The *confidentiality identifier* indicates that a confidentiality algorithm follows, after the *confidentiality length* field.

Component Portion

The component portion contains one or more components and their associated parameters. These components are used to invoke operations and return results to the invoking application.

A component may consist of the component itself, which indicates how the receiver will respond (if any response is necessary), the operation to be performed (optional), and any required parameters to be used when invoking the specified operation.

Components will always have an identifier, length field, and contents, which specify what the component is asking of the receiver. Their values and descriptions are shown as follows.

Component Type Identifier	**H**	**G**	**F**	**E**	**D**	**C**	**B**	**A**
Invoke (last)	1	1	1	0	1	0	0	1
Return Result (last)	1	1	1	0	1	0	1	0
Return Error	1	1	1	0	1	0	1	1
Reject	1	1	1	0	1	1	0	0
Invoke (not last)	1	1	1	0	1	1	0	1
Return Result (not last)	1	1	1	0	1	1	1	0

An Invoke (last) is used to invoke an operation at a remote application entity. For example, if a signaling point wanted to invoke some feature (operation) at another remote signaling point, a TCAP with an Invoke component would be sent to the remote signaling point, which would then act upon the Invoke request accordingly. The indication of "last" means that there are no more Invoke components for this transaction.

The Invoke (not last) is the same as just described, with the exception of the "not last" indicator, which alerts the receiver that additional Invokes are to be sent in relation to this transaction.

The Return Result (last) is used to return the results of an invoked operation. Not all operations require the Return Result to be sent. The indicator of "last" means that there will be additional results sent in response to the specified Invoke and transaction. These are usually accompanied by a correlation ID, which is used to associate the Return Result with an earlier Invoke component.

Return Error components are used to return an error code in the event of an operational error. An operational error is one that occurs at the application level and does not necessarily indicate a protocol problem.

The Reject component indicates a problem with protocol and means that the component was rejected because of an incorrect component portion in an earlier Invoke.

Component Length. The length field identifies how many octets are left in the component. This field and the identifier are not included in the length. There may be multiple components in one component portion; each component includes a component length field. This is not the length of the entire component portion (as in the transaction portion). This is only the length for this component (in which it is carried).

Component ID Identifier 1 1 0 0 1 1 1 1. This field indicates that a component has an invoke ID and, possibly, a correlation ID. As mentioned earlier, these IDs are of local significance only and are sent so that Return Results can be correlated with an earlier Invoke request. The invoke ID is assigned by the originator of an Invoke and requires that any Return Result will require a correlation ID. The correlation ID is the same ID as the invoke ID and is returned within the Return Result so the originator of an Invoke can correlate returns with their respective Invokes.

Component ID Length. The length field indicates the total length of the component ID field only. The ID length may be zero (unidirectional only), four (invoke ID only), or eight octets in length (both invoke ID and correlation ID provided). A Return Result can have only a correlation ID; therefore, the length for a Return Result will always be four octets. The same is true for Return Error and Reject. As shown in the following table, only the Invoke can carry both an invoke ID and a correlation ID.

Component IDs. The component may have an invoke ID and a correlation ID based on the following criteria. These IDs are used to associate multiple components with operations and transactions already in progress.

Component Type	Invoke ID	Correlation ID
Invoke	Optional	Yes
Return Result	No	Yes
Return Error	No	Yes
Reject	No	Yes

Invoke ID. The invoke ID is a one-octet identifier assigned to a component that is invoking an operation. This is an optional field and is of local significance only.

Correlation ID. The correlation ID is used whenever a component is responding to another component. If responding to a component that contained an invoke ID, the correlation ID is mandatory. The correlation ID is a mirror image of an invoke ID and is used as a reference for associated invoke IDs with previous invokes.

Operation Code Identifier	H	G	F	E	D	C	B	A
National TCAP	1	1	0	1	0	0	0	0
Private TCAP	1	1	0	1	0	0	0	1

National TCAP operation codes are defined in both ANSI and Bellcore standards. These are TCAP messages that must be common in all systems interworking with *Bell Operating Company* (BOC) networks and other ANSI networks.

Private TCAP operation codes are of significance only to private networks and are not compatible with any networks that interwork. Private networks cannot be connected to the *Public Switched Telephone Network* (PSTN) because the message types will conflict with national standards.

Operation Code Length. The length field identifies the length of the operation code field only and does not include itself or the identifier field. If the operation code is national

TCAP, the length will be two octets. If the operation code is private TCAP, there is no limitation on the length.

Operation Codes

The operation codes are implementation specific and are used by the application entities as instructions on how to carry out the component action. TCAP as a protocol does not interact with the operations; it only delivers them to the appropriate application process.

Operation codes may vary from network to network, because they are very network dependent. This section identifies those operation codes used within the Bellcore networks as identified in the Bellcore publication, TR-NWT-000246, T1.114.5, Issue 2, Revision 3, December 1992. The EIA/TIA has also specified operation codes for usage in the wireless network. The operation codes used in a wireless network provide a means for mobile switching centers to pass information from one to another as well as to invoke operations at remote MSCs.

The operation code is a two-octet field divided into the operation family and the operation specifier. The operation family is a seven-bit field (bits A–G) and identifies the group or category related to this operation. Bit H indicates whether or not a response is expected.

The second octet of the operation code consists of the operation specifier, which is used to identify the operation being requested. The specifier is the specific instruction being asked of the application.

Operation Families	**G**	**F**	**E**	**D**	**C**	**B**	**A**
Parameter	0	0	0	0	0	0	1
Charging	0	0	0	0	0	1	0
Provide Instructions	0	0	0	0	0	1	1
Connection Control	0	0	0	0	1	0	0
Caller Interaction	0	0	0	0	1	0	1
Send Notification	0	0	0	0	1	1	0
Network Management	0	0	0	0	1	1	1
Procedural	0	0	0	1	0	0	0
Operation Control	0	0	0	1	0	0	1
Report Event	0	0	0	1	0	1	0
Miscellaneous	1	1	1	1	1	1	0

The operation family does not provide enough information for the application to process the operation. This only serves as a category for the operation code. As you will notice in the rest of this section, the operation codes themselves are not all unique codes. They each use an ascending order beginning with the binary value of 0000 0001. The operation family field then becomes the delimiter between the various operation codes.

Parameter	**H**	**G**	**F**	**E**	**D**	**C**	**B**	**A**
Provide value	0	0	0	0	0	0	0	1
Set value	0	0	0	0	0	0	1	0

The parameter family provides instructions on how to use the parameters accompanying this parameter. There are two options: *provide value* and *set value*. The provide value indicator instructs the receiver to provide the requested value for the given parameter. The parameter portion of the TCAP message contains the actual parameter to which this operation refers.

The option *set value* instructs the receiver to set the value for the given parameter. This is for further study and has not yet been implemented. One example of how this operation code may be used is in the case where a signaling point has control of a call, but does not have the necessary resources available to continue processing (such as recordings). A temporary handover procedure is invoked, and the *provide value indicator* shown here could be included in the parameters sent to the remote application to indicate the need for resources.

Charging	**H**	**G**	**F**	**E**	**D**	**C**	**B**	**A**
Bill call	0	0	0	0	0	0	0	1

Presently, there is only one option for this operation code. The *bill call* option is used to notify the receiving application that a billing record is to be created for the calling party indicated in the parameters.

Provide Instructions	**H**	**G**	**F**	**E**	**D**	**C**	**B**	**A**
Start	0	0	0	0	0	0	0	1
Assist	0	0	0	0	0	0	1	0

This operation code is used to request instructions during an assist procedure. The *start* option is for further study, while the *assist* option indicates that the assist procedure has been requested and the receiver of an assist is asking the sender for instructions.

Connection Control	**H**	**G**	**F**	**E**	**D**	**C**	**B**	**A**
Connect	0	0	0	0	0	0	0	1
Temporary connect	0	0	0	0	0	0	1	0
Disconnect	0	0	0	0	0	0	1	1
Forward disconnect	0	0	0	0	0	1	0	0

The four codes used in this operation are grouped into associated pairs. The *connect* and *disconnect* are related and will require further study. When a connect is issued, the disconnect is used to terminate the connection.

The *temporary connect* is used when an error is encountered and a connection to another database for the completion of processing is required. The temporary connect operation must also be accompanied by parameters providing the subsystem number of the database, the routing number of the exchange to which a connection is being requested, and a reference number for the correlation of transactions between the database and the requesting exchange.

When the temporary connect is received, the receiving entity knows that the *forward disconnect* will follow. The forward disconnect is used to terminate a temporary connection.

Caller Interaction	**H**	**G**	**F**	**E**	**D**	**C**	**B**	**A**
Play announcement	0	0	0	0	0	0	0	1
Play announcement and collect digits	0	0	0	0	0	0	1	0
Indicate information waiting	0	0	0	0	0	0	1	1
Indicate information provided	0	0	0	0	0	1	0	0

The caller interaction family of operations enables announcements to be specified and provides a mechanism for application processes to communicate with one another regarding the state of expected information. The first two options, *play announcement* and *play announcement and collect digits*, are identical, with the exception of the latter, which waits and collects dialed digits from the user. The dialed digits can then be routed to a voice response unit for processing.

The indicators for information waiting enable one application process to inform another that there is information waiting and, when the information has been transferred, that the information has been provided.

Send Notification	**H**	**G**	**F**	**E**	**D**	**C**	**B**	**A**
When party free	0	0	0	0	0	0	0	1

The operation in this operation family is used for certain CLASS features, such as automatic callback where a caller reaches a busy signal and requests notification from the network when the party becomes available. When the called party becomes available by placing the receiver back on-hook, the remote exchange is notified via TCAP that the called party is available. ISUP call setup is then used to set the call up as if it were a normal call.

Network Management	**H**	**G**	**F**	**E**	**D**	**C**	**B**	**A**
Automatic code gap	0	0	0	0	0	0	0	1

The *automatic code gap* is used by network management to temporarily inhibit specified codes for the specified time. Additional parameters indicate the time the codes are to be inhibited and the duration between operations.

Procedural	**H**	**G**	**F**	**E**	**D**	**C**	**B**	**A**
Temporary handover	0	0	0	0	0	0	0	1
Report assist termination	0	0	0	0	0	0	1	0
Security	0	0	0	0	0	0	1	1

This operation family is used to control procedural operations. The *temporary handover* operation indicates that a temporary handover is presently in progress. When the temporary handover procedure has been completed, the receiver of this message will then release all resources dedicated to this operation.

The *report assist termination* is used to end an assist procedure.

Operation Control	**H**	**G**	**F**	**E**	**D**	**C**	**B**	**A**
Cancel	0	0	0	0	0	0	0	1

The *cancel* operation is used with the send notification operation to cancel a *when party free* operation. This is presently the only operation that can be canceled. The *service key* parameter accompanies this operation and provides the called party number.

Report Event	**H**	**G**	**F**	**E**	**D**	**C**	**B**	**A**
Voice message available	0	0	0	0	0	0	0	1
Voice message retrieved	0	0	0	0	0	0	1	0

This operation family is used with the *Voice Message Storage Retrieval* (VMSR) systems. When a subscriber uses such a service, and the VMSR system is located at another exchange (other than the subscriber's), the *voice message available* option is used to alert the subscriber's exchange of a message. The calling number of the party that left the message can be included, as well as the time the message was left. A Return Result is sent when the operation has been successful.

The *voice message retrieved* operation is used to remove the message available indicator from a subscriber's VMSR. Both the subscriber's number and the identification of the VMSR system used by the subscriber are provided with this operation code.

Miscellaneous	**H**	**G**	**F**	**E**	**D**	**C**	**B**	**A**
Queue call	0	0	0	0	0	0	0	1
Dequeue call	0	0	0	0	0	0	1	0

Error Codes

These error codes are used to indicate the reason for an unsuccessful completion of an operation. The error code is sent in a Return Error component to the originator of the operation request. The error codes are coded as either national or private. The error codes identified as follows are national error codes as defined in Bellcore publication TR-NWT-000246, Issue 2, Revision 2, December 1992:

Error Code	**H**	**G**	**F**	**E**	**D**	**C**	**B**	**A**
Unexpected component sequence	0	0	0	0	0	0	0	1
Unexpected data value	0	0	0	0	0	0	1	0
Unavailable resource	0	0	0	0	0	0	1	1
Missing customer record	0	0	0	0	0	1	0	0
Spare	0	0	0	0	0	1	0	1
Data unavailable	0	0	0	0	0	1	1	0
Task refused	0	0	0	0	0	1	1	1
Queue full	0	0	0	0	1	0	0	0
No queue	0	0	0	0	1	0	0	1
Timer expired	0	0	0	0	1	0	1	0
Data already exists	0	0	0	0	1	0	1	1
Unauthorized request	0	0	0	0	1	1	0	0
Not queued	0	0	0	0	1	1	0	1
Unassigned DN	0	0	0	0	1	1	1	0
Spare	0	0	0	0	1	1	1	1
Notification unavailable to destination DN	0	0	0	1	0	0	0	0
VMSR system ID did not match user profile	0	0	0	1	0	0	0	1

An unexpected component sequence indicates that one or more components were received that did not match what should have been received (was expected), considering the previous components received from the same originator. Likewise, an unexpected data value indicates that data received within a component was not what should have been sent, considering the type of component.

When resources required to carry out an Invoke are not available, the receiving application will return an error of unavailable resources. The originator must then determine if it will petition another application at another signaling point, or possibly even within the same signaling point. Resources include things like recordings and databases.

When accessing a database to add information to a customer database or when retrieving information from a customer record, the identity of the called party will be used to identify the record. If the record does not exist, an error of missing customer record will be sent to the originating application. Customer records are used to determine the billing options for a call as well as the type of features a subscriber can use.

With certain CLASS features, the customer record will also provide specific instructions on how to handle specific services on a per-call basis.

Data unavailable could indicate that the database containing requested information is not available. This could be the case if there was a failure within the subsystem or if the application process failed and was unable to reach the database.

Task refused is returned when the application entity has been requested to perform some sort of task, but the entity chosen cannot perform the task. No reason is given, only the Reject.

With custom calling features and CLASS, certain features require a queue to temporarily store numbers. For example, automatic callback and automatic recall require the last dialed number or the last number that was called to be stored in a queue until the feature is invoked. The feature (application entity) then accesses this queue to complete the processing of the feature. There are three states associated with these queues: queue full, no queue, and not queued. Not queued is used when a number is to be removed from a called party queue.

When a parameter is sent with data that has already been received, the reject cause will be data that already exists. Only a parameter change operation can change data that has already been received. Several reject causes are related to *directory numbers* (DNs). These reject causes are sent for a variety of reasons, but are usually related to the lack of resources or are unauthorized to access services and/or databases being requested.

A VMSR system is now being offered in many areas. This enables voice-mail equipment to be installed at the central office, rather than the subscriber purchasing a voice-mail system. The subscriber must purchase the service, and the service must be an entry in the customer record before access is allowed. If access is attempted and the directory number is not a subscriber to the VMSR system, a reject cause of "VMSR system identification did not match user profile" is sent.

Parameters

Parameters are associated with individual components and are the last items in the component portion of the TCAP message. There are three elements in any parameter: the parameter identifier, length, and contents.

The parameter identifier is used to identify the individual parameters and consists of a one-octet field. All parameter identifiers are listed in the following table. The length field indicates the length of the contents field, which is a variable field. The contents can be one-octet specifiers or implementation-dependent codes.

The following parameters are found in the Bellcore publication, TR-NWT-000246, Issue 2, Revision 2, December 1992.

Parameter Name	**H**	**G**	**F**	**E**	**D**	**C**	**B**	**A**
Timestamp	0	0	0	1	0	1	1	1
ACG indicators	1	0	0	0	0	0	0	1
Standard announcement	1	0	0	0	0	0	1	0
Customized announcement	1	0	0	0	0	0	1	1
Digits	1	0	0	0	0	1	0	0
Standard user error code	1	0	0	0	0	1	0	1
Problem data	1	0	0	0	0	1	1	0
SCCP calling party address	1	0	0	0	0	1	1	1
Transaction ID	1	0	0	0	1	0	0	0
Package type	1	0	0	0	1	0	0	1
Service key	1	0	1	0	1	0	1	0
Busy/idle status	1	0	0	0	1	0	1	1
Call forwarding status	1	0	0	0	1	1	0	0
Originating restrictions	1	0	0	0	1	1	0	1
Terminating restrictions	1	0	0	0	1	1	1	0
DN to line service type mapping	1	0	0	0	1	1	1	1
Duration	1	0	0	1	0	0	0	0
Returned data	1	0	1	1	0	0	0	1
Bearer capability requested	1	0	0	1	0	0	1	0
Bearer capability supported	1	0	0	1	0	0	1	1
Reference ID	1	0	0	1	0	1	0	0
Business group	1	0	0	1	0	1	0	1
Signaling network identifier	1	0	1	1	0	1	1	0
Generic name	1	0	0	1	0	1	1	1
Message waiting indicator type	1	0	0	1	1	0	0	0
Look ahead for busy	1	0	0	1	1	0	0	1
Circuit identification code	1	0	0	1	1	0	1	0
Precedence identifier	1	0	0	1	1	0	1	1
Call reference identifier	1	0	0	1	1	1	0	0

Parameter Values

The following are all the parameters just listed and the values of their contents. Parameters complement the components already described and their operation codes. Operation codes may choose any number of the following parameters as a set. These are all national parameters as described and are defined in the Bellcore publication TR-NWT-000246, Issue 2, Revision 2, December 1992.

Timestamp	**H**	**G**	**F**	**E**	**D**	**C**	**B**	**A**
Octets 1–2	Year in binary (such as 93)							
Octets 3–4	Month in binary (such as 07)							
Octets 5–6	Day in binary (such as 28)							
Octets 7–8	Hour in binary (such as 17)							
Octets 9–10	Minutes in binary (such as 30)							
Octet 11	1 or 2 (ahead or behind GMT)							
Octets 12–13	Hours (see above description)							
Octets 14–15	Minutes (see above description)							

The timestamp provides the time and date when an event occurred. Both local time and the difference between local time and *Greenwich Mean Time* (GMT) are provided. The first hour and minutes are those of local time, and the second hour and minutes fields reflect the difference between local time and GMT. For example, if the local time is 1730, and the local time is in Atlanta (which is five hours behind Greenwich time), the second hour and minutes field would reflect a difference of 20500.

Automatic Code Gap (ACG)	**H**	**G**	**F**	**E**	**D**	**C**	**B**	**A**
Control Cause Indication								
Vacant code	0	0	0	0	0	0	0	1
Out-of-band	0	0	0	0	0	0	1	0
Database overload	0	0	0	0	0	0	1	1
Destination mass calling	0	0	0	0	0	1	0	0
Operation Support System (OSS) initiated	0	0	0	0	0	1	0	1

The *Automatic Code Gap* (ACG) is a network management function that enables the network to throttle traffic for a specified period of time. ACG can be initiated manually or automatically. The preceding codes indicate the cause for invoking ACG.

Vacant code indicates that calls are being received for an unassigned code. Out-of-band is related to calls for a band that a subscriber does not subscribe to. Database overload indicates a database that is overloaded, while destination mass calling indicates that an excessive number of calls are being received for a destination.

When the ACG is initiated manually, it is initiated by an *Operations Support System* (OSS). When this is the case, the cause code will indicate that the OSS initiated the ACG.

Duration (in seconds)	**H**	**G**	**F**	**E**	**D**	**C**	**B**	**A**
Not used	0	0	0	0	0	0	0	0
1	0	0	0	0	0	0	0	1
2	0	0	0	0	0	0	1	0
4	0	0	0	0	0	0	1	1
8	0	0	0	0	0	1	0	0
16	0	0	0	0	0	1	0	1
32	0	0	0	0	0	1	1	0
64	0	0	0	0	0	1	1	1
128	0	0	0	0	1	0	0	0
256	0	0	0	0	1	0	0	1

512	0	0	0	0	1	0	1	0
1024	0	0	0	0	1	0	1	1
2048	0	0	0	0	1	1	0	0

A one-octet field indicates the time duration in seconds in which an ACG should be applied:

Gap (in seconds)	**H**	**G**	**F**	**E**	**D**	**C**	**B**	**A**
Remove gap control	0	0	0	0	0	0	0	0
0.00	0	0	0	0	0	0	0	1
0.10	0	0	0	0	0	0	1	0
0.25	0	0	0	0	0	0	1	1
0.50	0	0	0	0	0	1	0	0
1.00	0	0	0	0	0	1	0	1
2.00	0	0	0	0	0	1	1	0
5.00	0	0	0	0	1	0	0	0
15.00	0	0	0	0	1	0	0	1
30.00	0	0	0	0	1	0	1	0
60.00	0	0	0	0	1	0	1	1
120.00	0	0	0	0	1	1	0	0
300.00	0	0	0	0	1	1	0	1
600.00	0	0	0	0	1	1	1	0
Stop all calls	0	0	0	0	1	1	1	1

A one-octet field indicates the interval between applications of the ACG control. Time is measured in seconds.

Standard Announcement	**H**	**G**	**F**	**E**	**D**	**C**	**B**	**A**
Not used	0	0	0	0	0	0	0	0
Out-of-band	0	0	0	0	0	0	0	1
Vacant code	0	0	0	0	0	0	1	0
Disconnected number	0	0	0	0	0	0	1	1
Reorder (120 pulses per minute) tone	0	0	0	0	0	1	0	0
Busy (60 pulses per minute)	0	0	0	0	0	1	0	1
No circuit available	0	0	0	0	0	1	1	0
Reorder recording	0	0	0	0	0	1	1	1
Audible ringing	0	0	0	0	1	0	0	0

When an announcement is to be applied to a particular call, either a standard announcement or a customized announcement may get requested. The standard announcements for a Bellcore network are depicted in the preceding table.

Customized Announcement. These are implementation dependent and enable networks to address their own network-specific announcements. The parameter consists of two elements: the announcement set and the individual announcement. Length is variable.

Customized announcements are unique within the network in which they reside. Independent companies and private networks may implement their own announcements for usage within their own networks. Both standard and customized announcements can be used within the same network. These two conventions enable either one to be requested for a specific call.

Digits	H	G	F	E	D	C	B	A
	Type of digits							
	Nature of number							
	Number plan and encoding							
	Number of digits							
	Digits							

Type of digits								
Not used	0	0	0	0	0	0	0	0
Called party number	0	0	0	0	0	0	0	1
Calling party number	0	0	0	0	0	0	1	0
Caller interaction	0	0	0	0	0	0	1	1
Routing number	0	0	0	0	0	1	0	0
Billing number	0	0	0	0	0	1	0	1
Destination number	0	0	0	0	0	1	1	0
Local Access and Transport Area (LATA)	0	0	0	0	0	1	1	1
Carrier	0	0	0	0	1	0	0	0
Last calling party	0	0	0	0	1	0	1	0
Calling directory number	0	0	0	0	1	0	1	1
VMSR identified	0	0	0	0	1	1	0	0
Original called number	0	0	0	0	1	1	0	1
Redirecting number	0	0	0	0	1	1	1	0
Connected number	0	0	0	0	1	1	1	1

When digits are being received in TCAP for invoking features, the type of digits (source) and the coding of the digits (BCD) must be specified so that the receiving entity knows how to decode the digits. The preceding types indicate the source type of the digits and enable the receiver to determine how to handle the received digits.

In some cases (CLASS features, that is), the subscriber may be requested to dial digits. When this is the case, the digits' type is caller interaction. An example of this would be when a caller inputs a calling card number when requested by an announcement.

Dialed numbers can also be used for network routing (routing number) or billing information (billing number). When information is being requested regarding a particular line, the destination number may be provided as a reference to the subscriber who owns that line.

Digits can also specify the *Local Access Transport Area* (LATA) a particular caller is calling from that will be used in routing and accessing line information. All carriers are numbered as well. When callers wish to dial their long-distance carrier's operator direct, they dial a 10xxx number, the last three digits being the carrier number. These digits can be carried through TCAP as well to access a billing record and record usage of a carrier or when a customer line record is accessed to specify the long-distance carrier for that subscriber.

Last calling party digits identify the last directory number to dial a certain number. This is used with CLASS features, such as automatic callback, to identify the directory number of the last party to call a number. Last party called identifies the last party that was called (directory number).

A *Voice Mail Send/Receive* (VMSR) is also identified by digits and is referred to in messages accessing and/or invoking a VMSR system for a call.

Redirecting number identifies the number of the party who last invoked a forwarded call to a specific number. For example, with follow-me forwarding, a number can be reforwarded from any phone whenever needed. The redirecting number indicates the last directory number to forward the specified number.

Nature of number	**H**	**G**	**F**	**E**	**D**	**C**	**B**	**A**
National								0
International								1
No presentation restriction							0	
Presentation restriction							1	

Encoding	**H**	**G**	**F**	**E**	**D**	**C**	**B**	**A**
Not used					0	0	0	0
Binary Coded Decimal (BCD)					0	0	0	1
IA5					0	0	1	0

Numbering plan	**H**	**G**	**F**	**E**	**D**	**C**	**B**	**A**
Unknown or not applicable	0	0	0	0				
ISDN numbering	0	0	0	1				
Telephony numbering	0	0	1	0				
Data numbering	0	0	1	1				
Telex numbering	0	1	0	0				
Maritime mobile numbering	0	1	0	1				
Land mobile numbering	0	1	1	0				
Private numbering plan	0	1	1	1				

Number of digits	**H**	**G**	**F**	**E**	**D**	**C**	**B**	**A**
For BCD encoding:	2nd digit *n*th digit				1st digit (*n* − 1)th digit			
Digits are coded as follows:								
Digit 0 or filler	0	0	0	0				
Digit 1	0	0	0	1				
Digit 2	0	0	1	0				
Digit 3	0	0	1	1				
Digit 4	0	1	0	0				
Digit 5	0	1	0	1				
Digit 6	0	1	1	0				
Digit 7	0	1	1	1				
Digit 8	1	0	0	0				
Digit 9	1	0	0	1				
Spare	1	0	1	0				
Code 11	1	0	1	1				
Code 12	1	1	0	0				
*	1	1	0	1				
#	1	1	1	0				
ST	1	1	1	1				

All of the preceding identify the type of digits as well as the digits themselves. These are used when presenting digits into the TCAP message so the receiver can understand the origin of the digits and know how they are to be decoded. Digits appear in a variety of TCAP transactions and are being used more and more as AIN features begin working their way into the network.

Standard User Error Code	**H**	**G**	**F**	**E**	**D**	**C**	**B**	**A**
Not used	0	0	0	0	0	0	0	0
Caller abandon	0	0	0	0	0	0	0	1
Improper caller response	0	0	0	0	0	0	1	0

User error codes define the reason for a user-induced failure. When a transaction is interrupted and aborted because of subscriber actions, these cause codes are used to describe the reason. Presently, only two causes are defined.

Caller abandon indicates that a subscriber (calling party) hung up before the transaction could be completed. The transaction could have been a calling card call to another number, which involves TCAP to access the calling card database for verification and billing. If any type of call forwarding or access to a voice-mail system (VMSR) is used, TCAP is required for completing the informational transactions.

Improper caller response indicates that the caller was queried to enter digits and did not enter the correct digits, or entered in the wrong number of digits. A good example of this would be entering in a calling card number when prompted. Another example is with VMSR systems where a caller may be prompted by an announcement to enter in a callback number, but enters in an incorrect number of digits.

Problem Data. The problem data cites the specific reason for the error. This field is implementation dependent and is coded as contextual and primitive. The format is the same as for other parameters: a one-octet parameter identifier, the length of the parameter contents, and the parameter contents (variable-length field). This is provided in the protocol as an optional source of information when a problem occurs. Systems can elect to send additional information and codes regarding a specific set of problems encountered.

SCCP Calling Party Address. This is the address field used by the receiver of a handover procedure to determine the calling party address. The address structure is the same as discussed in Chapter 7. The address can consist of global title digits, a point code, or a subsystem number. This information is part of the temporary handover parameter.

Transaction ID. This is the same format used in the transaction portion. The difference is that this is used during the temporary handover procedure by the receiver of a temporary handover message. This information is part of the temporary handover parameter.

Package Type. The package type is used here to notify the receiver of the temporary handover parameter what package type to respond with. The package type indicated here will be used to return some sort of response to the calling party address (if included in the transaction).

Service Key. The service key is used to specify which parameters should be used to access a record. This is used in database queries, which are used to access customer database records and service records. The service key is coded as contextual and constructor.

Busy/Idle Status	**H**	**G**	**F**	**E**	**D**	**C**	**B**	**A**
Not used	0	0	0	0	0	0	0	0
Busy	0	0	0	0	0	0	0	1
Idle	0	0	0	0	0	0	1	0

Busy and idle statuses are used to provide information regarding the status of a subscriber line when particular CLASS features and custom calling features are deployed.

Call Forwarding Status	**H**	**G**	**F**	**E**	**D**	**C**	**B**	**A**
Call Forwarding Variable								
Service not supported	0	0						
Active	0	1						
Not active	1	0						
Spare	1	1						

Call Forwarding on Busy	**H**	**G**	**F**	**E**	**D**	**C**	**B**	**A**
Service not supported			0	0				
Active			0	1				
Not active			1	0				
Spare			1	1				

Call Forwarding Don't Answer	**H**	**G**	**F**	**E**	**D**	**C**	**B**	**A**
Service not supported					0	0		
Active					0	1		
Not active					1	0		
Spare					1	1		

Selective Forwarding	**H**	**G**	**F**	**E**	**D**	**C**	**B**	**A**
Service not supported							0	0
Active							0	1
Not active							1	0
Spare							1	1

These call-forwarding parameters present the status of a line using call-forwarding features. These are used when custom calling or CLASS is being offered in a calling area. A call-forwarding variable is used to forward a call immediately before ringing the called party.

Call forwarding on busy will forward a call only if the called party is off-hook. *Call forwarding don't answer* forwards a call when the called party does not answer within the predetermined number of rings (defined by the subscriber). *Selective forwarding* enables only select calls to be forwarded, based upon criteria defined by the user (such as calling party number).

Originating Restrictions	**H**	**G**	**F**	**E**	**D**	**C**	**B**	**A**
Denied origination	0	0	0	0	0	0	0	0
Fully restricted origination	0	0	0	0	0	0	0	1
Semirestricted origination	0	0	0	0	0	0	1	0
Unrestricted origination	0	0	0	0	0	0	1	1

Restrictions are assigned to business groups (such as Centrex service) to define the type of outside calling a station is permitted to make. Denied origination indicates that calls are not allowed to be originated from the specified line. Business groups are used to define Centrex lines and other similar services.

Fully restricted origination enables a line to originate calls to lines within the business group, but not to the attendant (the local business group operator, usually at the reception desk of a business) and not to lines outside of the business group.

Semirestricted origination enables a line to call outside the business group, but not by direct dial. The line can be forwarded, conferenced, or transferred via an attendant, but cannot direct dial an outside line.

Unrestricted origination enables a line to call any number within the business group or outside the business group. This allows full, unrestricted access to any outside line.

Terminating Restrictions	**H**	**G**	**F**	**E**	**D**	**C**	**B**	**A**
Denied termination	0	0	0	0	0	0	0	0
Fully restricted termination	0	0	0	0	0	0	0	1
Semirestricted termination	0	0	0	0	0	0	1	0
Unrestricted termination	0	0	0	0	0	0	1	1
Call rejection applies	0	0	0	0	0	1	0	0

Terminating restrictions are like the originating restrictions, but apply only to incoming calls. Denied termination prevents any calls from being terminated to the line. The line may be allowed to dial within the group or may even be allowed to dial outside the group (depending on the originating restriction applied).

Fully restricted termination prevents all calls from outside the business group being terminated to this line. Calls cannot be transferred from any other line or the attendant or forwarded.

Semirestricted lines are allowed to receive calls from within the business group. Outside calls must be transferred or forwarded to the line. Lines outside the business group cannot direct dial this line.

Unrestricted termination enables full access to the line from within the business group and outside the business group. Call rejection enables a line to request a rejection of an incoming call. This is communicated using electronic Centrex phones with displays. The display shows the calling party number (ANI), allowing the called party to determine whether to accept or reject the call.

Directory Number to Line Service Type Mapping

Match Status	**H**	**G**	**F**	**E**	**D**	**C**	**B**	**A**
Spare	0	0						
No match	0	1						
Match	1	0						
Spare	1	1						

Line Service Type	**H**	**G**	**F**	**E**	**D**	**C**	**B**	**A**
Individual			0	0	0	0	0	0
Coin			0	0	0	0	0	1
Multiline hunt			0	0	0	0	1	0
PBX			0	0	0	0	1	1
Choke			0	0	0	1	0	0
Series completion			0	0	0	1	0	1
Unassigned DN			0	0	0	1	1	0
Multiparty			0	0	0	1	1	1
Nonspecific			0	0	1	0	0	0
Temporarily out of service			0	0	1	0	0	1

Line Service Type identifies the type of line service for a given subscriber line. This information must be retrieved from a database line record for a given subscriber line. Numbers that are not in service or have been disconnected are also indicated with these codes.

Duration	**H**	**G**	**F**	**E**	**D**	**C**	**B**	**A**
	Hours				Hours			
	Minutes				Minutes			
	Seconds				Seconds			

Duration is used for features such as automatic callback, where the called party must be monitored until it is free. The duration parameter enables a duration to be set for monitoring the called party. After the duration period, if the called party is still busy, the feature is restarted for another duration.

Bearer Capability Requested. This information is related to the type of bearer capability a subscriber is allowed. The information is stored in the line information database and retrieved via TCAP when queried by an end office.

Octet 1

Extension indicator	**H**	**G**	**F**	**E**	**D**	**C**	**B**	**A**
Octet extended to next octet	1							
Octet not extended to next octet	0							

Coding standard	**H**	**G**	**F**	**E**	**D**	**C**	**B**	**A**
ITU-TS standardized		0	0					
Reserved for other international standards		0	1					
National standard		1	0					
Reserved		1	1					

Information transfer capability	**H**	**G**	**F**	**E**	**D**	**C**	**B**	**A**
Speech				0	0	0	0	0
Unrestricted digital information				0	1	0	0	0
Restricted digital information				0	1	0	0	1
3.1 kHz audio				1	0	0	0	0
7 kHz audio				1	1	0	0	1
15 kHz audio				1	0	0	1	0
Video				1	1	0	0	0

Octet 2

Extension indicator	**H**	**G**	**F**	**E**	**D**	**C**	**B**	**A**
Octet extended to next octet	1							
Octet not extended to next octet	0							

Transfer mode	**H**	**G**	**F**	**E**	**D**	**C**	**B**	**A**
Circuit mode		0	0					
Packet mode		1	0					

Information transfer rate	**H**	**G**	**F**	**E**	**D**	**C**	**B**	**A**
Channel size				0	0	0	0	0
64 kbps				1	0	0	0	0
384 kbps				1	0	0	1	1
1536 kbps				1	0	1	0	1
1920 kbps				1	0	1	1	1

The information transfer rate can be used to indicate the transfer rate in both directions through the use of octet 2b. When octet 2b is not included, the bidirectional transfer rate is symmetrical with the rate indicated in octet 2. When octet 2b is included, then the transfer rate in the direction origination to destination is that of the rate indicated in octet 2b. This enables the setting of different transfer rates in each direction or the same transfer rate in both directions.

Octet 2a

Extension indicator	**H**	**G**	**F**	**E**	**D**	**C**	**B**	**A**
Octet extended to next octet	1							
Octet not extended to next octet	0							

Structure	**H**	**G**	**F**	**E**	**D**	**C**	**B**	**A**
Default (see note for default values)		0	0	0				
8 kHz integrity		0	0	1				
Service data unit integrity		1	0	0				
Unstructured		1	1	1				

Note: The default values assigned (if field = 000, or octet 2a is omitted) are as follows:

Transfer mode	*Transfer capability*	*Structure*
Circuit	Speech	8 kHz integrity
Circuit	Unrestricted digital	8 kHz integrity
Circuit	Restricted digital	8 kHz integrity
Circuit	Audio	8 kHz integrity
Circuit	Video	8 kHz integrity
Packet	Unrestricted digital	Service data unit integrity

Configuration	**H**	**G**	**F**	**E**	**D**	**C**	**B**	**A**
Point-to-point					0	0		
Multipoint					1	0		

Establishment	**H**	**G**	**F**	**E**	**D**	**C**	**B**	**A**
Demand							0	0

Octet 2b

This octet can be omitted, unless it is desirable to indicate a different transfer rate in one direction other than what is specified in octet 2. When this octet is used, the transfer rate indicated is applicable to the direction origination to destination.

Extension indicator	**H**	**G**	**F**	**E**	**D**	**C**	**B**	**A**
Octet extended to next octet	1							
Octet not extended to next octet	0							

Symmetry	**H**	**G**	**F**	**E**	**D**	**C**	**B**	**A**
Bidirectional symmetric		0	0					
Bidirectional asymmetric		0	1					
Unidirectional (origination to destination)		1	0					
Unidirectional (destination to origination)		1	1					

Information transfer rate	**H**	**G**	**F**	**E**	**D**	**C**	**B**	**A**
Channel size				0	0	0	0	0
64 kbps				1	0	0	0	0
384 kbps				1	0	0	1	1
1536 kbps				1	0	1	0	1
1920 kbps				1	0	1	1	1

Octet 3

This is an optional octet that can be omitted or repeated. When there is a need to identify more than one protocol at higher layers, this field can be repeated to reflect each of the other protocols.

Extension indicator	**H**	**G**	**F**	**E**	**D**	**C**	**B**	**A**
Octet extended to next octet	1							
Octet not extended to next octet	0							

Multiplier or layer identification	**H**	**G**	**F**	**E**	**D**	**C**	**B**	**A**
Bearer capability multiplier		0	0					
User information layer 1 protocol		0	1					
User information layer 2 protocol		1	0					
User information layer 3 protocol		1	1					

User information layer 1 protocol identification	**H**	**G**	**F**	**E**	**D**	**C**	**B**	**A**
ITU-TS (CCITT) Rec. I.412		0	0	0			0	0
Rate Adaptation (see note)				0	0	0	0	1
ITU-TS (CCITT) Rec. G.711 u-law		0	0	0			1	0
ITU-TS (CCITT) Rec. G.711 A-law		0	0	0			1	1
ITU-TS (CCITT) Rec. G.721 32 kbps ADPCM				0	0	1	0	0
ITU-TS (CCITT) Rec. G.722, G.725, 7 kHz audio		0	0	1			0	1

User information layer 2 protocol identification	**H**	**G**	**F**	**E**	**D**	**C**	**B**	**A**
Undefined				0	0	0	0	0
ITU-TS (CCITT) Rec. Q.921 (I.441)				0	0	0	1	0
ITU-TS (CCITT) Rec. Q.710				0	0	0	1	1
ITU-TS (CCITT) Rec. X.25 link level				0	0	1	1	0

User information layer 3 protocol identification	**H**	**G**	**F**	**E**	**D**	**C**	**B**	**A**
Undefined				0	0	0	0	0

ITU-TS (CCITT) Rec. Q.931 (I.451)	0	0	0	1	0
ITU-TS (CCITT) Rec. X.25 packet level	0	0	1	1	0

Note: When the multiplier or layer identification field indicates one of the user information layers (1 through 3), the user information protocol identification fields are indicated in the same octet. Each of these protocol identifiers is directly related to the layer specification in bits GF. For example, if bits GF indicate a layer-1 protocol, then the bits EDCBA equal that of the appropriate layer-1 protocol, as shown in the preceding table under User information layer 1 protocol identification.

Bearer Capability Supported	**H**	**G**	**F**	**E**	**D**	**C**	**B**	**A**
Not used	0	0	0	0	0	0	0	0
Bearer capability is supported	0	0	0	0	0	0	0	1
Bearer capability is not supported	0	0	0	0	0	0	1	0
Bearer capability not authorized	0	0	0	0	0	0	1	1
Bearer capability not presently available	0	0	0	0	0	1	0	0
Bearer capability not implemented	0	0	0	0	0	1	0	1

Reference ID. This field is used to identify the transaction between the database and the exchange during the assist service. The format consists of an identifier, a length indicator, and the variable contents field. The contents field carries the identification number (four octets) used by this service to correlate the transaction with the database access. The identification number is assigned by the database.

Business Group Identifier

Octet 2

Length of parameter

The length indicates the entire length of the parameter, not counting this field.

Octet 3

Attendant status	**H**	**G**	**F**	**E**	**D**	**C**	**B**	**A**
No indication		0						
Attendant line		1						

Business group identifier	**H**	**G**	**F**	**E**	**D**	**C**	**B**	**A**
Multilocation business group			0					
Interworking private number			1					

Line privileges information indicator	**H**	**G**	**F**	**E**	**D**	**C**	**B**	**A**
Fixed line privileges				0				
Customer-defined line privileges				1				

Party selection	**H**	**G**	**F**	**E**	**D**	**C**	**B**	**A**
No indication					0	0	0	0
Calling party number					0	0	0	1
Called party number					0	0	1	0
Connected party number					0	0	1	1
Redirecting number					0	1	0	0
Original called number					0	1	0	1

Octet 4 and 5

Subgroup ID	H	G	F	E		D	C	B	A
No indication	0	0	0	0		0	0	0	0
Customer-assigned subgroup codes	0	0	0	0		0	0	0	1
					to				
	1	1	1	1		1	1	1	1

Octet 6

This field is associated with the line privileges field in the preceding table. If the line privileges field indicates customer-defined line privileges, then this one-octet field is used to indicate those customer-defined line privilege codes. If the line privileges field indicates fixed line privileges, this field is divided into two subfields. The bits HGFE indicate the terminating restrictions, while the bits DCBA indicate the originating restrictions.

Line privileges	H/D	G/C	F/B	E/A
Unrestricted	0	0	0	0
Semirestricted	0	0	0	1
Fully restricted	0	0	1	0
Fully restricted intraswitch	0	0	1	1
Denied	0	1	0	0

Business groups are used to offer PBX-type services to business customers who do not wish to purchase a PBX. Probably the most common type of service offered under this category is Centrex. These parameters are used to identify what class of Centrex is being used with the specified line.

In addition to defining the business group, these parameters also enable the definition of the various members within a business group, and they enable lines to be placed into subgroups. Subgroups may be located within the same area or may be in another calling area. When this is the case, this information must be shared with the remote offices.

Signaling Networks Identifier. The signaling network identifier is used to indicate to the receiver which networks the sender anticipates going through to reach the destination. Each network ID requires two octets. The signaling networks identifier is a variable parameter.

Generic Name

Type of Name	H	G	F	E	D	C	B	A
Spare	0	0	0					
Calling name	0	0	1					
Original called name	0	1	0					
Redirecting name	0	1	1					
Connected name	1	0	0					
Spare	1	0	1					
		to						
	1	1	1					

Availability	**H**	**G**	**F**	**E**	**D**	**C**	**B**	**A**
Name available/unknown				0				
Name not available				1				

Presentation	**H**	**G**	**F**	**E**	**D**	**C**	**B**	**A**
Presentation allowed							0	0
Presentation restricted							0	1
Blocking toggle							1	0
No indication							1	1

Message Waiting Indicator Type	**H**	**G**	**F**	**E**	**D**	**C**	**B**	**A**
	2nd digit				1st digit			

The message waiting indicator is a two-octet parameter predefined by the customer. This parameter and its contents notify the customer about the type of message waiting in a telephone-company-provided voice-mail system. The message waiting type must be defined at service deployment.

Look Ahead for Busy Response

Acknowledgment type	**H**	**G**	**F**	**E**	**D**	**C**	**B**	**A**
Path reservation denied	0	0						
Negative acknowledgment	0	1						
Positive acknowledgment	1	0						
Spare	1	1						

Location	**H**	**G**	**F**	**E**	**D**	**C**	**B**	**A**
User					0	0	0	0
Private network serving the local user					0	0	0	1
Public network serving the local user					0	0	1	0
Transit network					0	0	1	1
Public network serving the remote user					0	1	0	0
Private network serving the remote user					0	1	0	1
Local interface controlled by this signaling link					0	1	1	0
International network					0	1	1	1
Network beyond interworking point					1	0	0	0

Acknowledgment type	**H**	**G**	**F**	**E**	**D**	**C**	**B**	**A**
Path reservation denied	0	0						
Negative acknowledgment	0	1						
Positive acknowledgment	1	0						
Spare	1	1						

Circuit Identification Code. The *circuit identification code* (CIC) identifies the trunk circuit used to send the voice or data to the called party. The CIC is also indicated in any ISUP messages, which are used to set up the connection between end offices. The CIC is a two-octet field.

Precedence. Used in military systems to determine the type of call precedence used. Call precedence enables calls in progress to be terminated and the trunk to be released so that those of a higher military rank may use the trunk for an outgoing call. This feature enables the military to maintain a low number of trunks at all military installations, while still allowing high-ranking officials access to lines when they need them. All lines within the military installation are coded, and all users are prompted to input an identification number, which identifies their rank and precedence level.

Octet 1

Precedence level	**H**	**G**	**F**	**E**	**D**	**C**	**B**	**A**
Flash override					0	0	0	0
Flash					0	0	0	1
Immediate					0	0	1	0
Priority					0	0	1	1
Routine					0	1	0	0
Service								

These fields are used to indicate the service code as assigned by the code administrator that applies to this call. The service codes are used to identify the type of service subscribed to in this particular network.

Call Reference. Call references are only used with military installation calls. These codes enable the tracking of calls independent of the circuit number. The call reference pertains to the identification input when callers dial their IDs and telephone numbers (called party). This enables the tracking of both the line number and the individual code used to place the call.

This is a six-octet field, divided into three-octet sections. The first three octets identify the identification number assigned to this call. This identification number is used to identify a particular call and is separate from the circuit identification code.

The next three octets are used to indicate the point code in which the identification code has been assigned. The identification code is only significant to the signaling point indicated in the last three octets.

Summary

As new services are defined, these parameters and operation codes will grow. The TCAP protocol itself is becoming an important aspect of the SS7 network and will become more important as the IN grows.

New switches become more and more sophisticated, and offer new and improved services and features. These features will require the support of TCAP for remote activation. There is no doubt that TCAP will be an important part of the communications network well into the next decade.

The current traffic mix in SS7 networks today tends to be primarily ISUP messages with some TCAP traffic. This is quickly changing as TCAP becomes the predominant traffic generator and as the SS7 network expands and becomes more sophisticated.

Local Number Portability (LNP) has increased the amount of TCAP traffic across the SS7 network in giant proportions. As subscribers begin switching to competitive local access providers, the amount of TCAP traffic will grow exponentially.

As wireless services and *Personal Communications Services* (PCS) become more and more popular, they will demand more and more of the network's resources. TCAP will prove to be the most valuable of all the SS7 protocols.

Chapter

9

Overview of ISUP

The *ISDN User Part* (ISUP) has been used in U.S. networks for many years now as an alternative to the European equivalent, the *Telephone User Part* (TUP). In early implementations of *Signaling System 7* (SS7), TUP was found to be far too limited for the scope of North American networks and was modified to align with the future services of *Integrated Services Digital Network* (ISDN) and many other network features still under development. Today, many of those features are under implementation, and the SS7 network is being utilized more and more. However, much of its potential is still untapped.

ISUP has been a good protocol for circuit-related messages, but is already under modification to support new broadband services soon to be offered by major telephone companies. The new broadband services being offered for tomorrow's networks will require a new version of ISUP called *broadband ISUP* (BISUP).

The ISUP is used to set up and tear down all circuits used for data or voice calls in the *Public Switched Telephone Network* (PSTN). In addition to its usage in the PSTN, ISUP can also be found in wireless networks for establishing trunk connections between switching centers.

ISUP is not widely used throughout the world; in fact, the United States was the first to adopt ISUP for usage in its networks. The *International Telecommunications Union* (ITU) is currently developing an international version of ISUP, which will be used in the international plane. Other countries use ISUP's predecessor, the TUP.

TUP does not offer the same services and capabilities as ISUP, which was designed with the ISDN in mind, and is fully compatible with the signaling in ISDN. It was for this reason that ISUP quickly replaced TUP in U.S. networks.

In fact, the TUP is considered almost obsolete for those wanting to offer more control over their circuits. TUP is good for physical circuit connections, but is not capable of handling virtual circuits, which are permanent in digital networks.

Another shortcoming of the TUP is its incapability to support *bearer* circuits. In a digital network, there are both physical and logical circuits that are dependent on the amount of data being sent by the user. This bearer traffic determines how many virtual circuits will be needed to accommodate the data. The ISUP provides the mechanisms for supporting bearer traffic, but does not fully support broadband signaling, which uses a different scheme altogether.

Additional work is currently underway to accommodate the new broadband services to be offered by the telephone companies. *Asynchronous transfer mode* (ATM) and *broadband ISDN* (BISDN) are making their way into the PSTN, replacing the existing DS3 and DS1 facilities that were used for so many years between exchanges. These new facilities will bring new configuration parameters and choices to be made by the protocol.

The support of BISDN (which will become the subscriber interface to the broadband network) through the SS7 network is accommodated by a new signaling protocol—BISUP. Many similarities exist between ISUP and BISUP; in fact, the same procedures and message types are used in both. The exception in BISUP lies in additional message types and changes in how circuits will be assigned to a call connection.

Another fundamental difference being introduced with broadband signaling is the advent of fully associated signaling rather than quasi-associated signaling. Fully associated signaling is accomplished by using the same path as the voice circuit, as would be the case when a channel from a DS3 circuit is used for signaling, and the other channels are used for voice and data. Once ATM has been deployed in the telephone networks, SS7 will be sent through the ATM network along with the voice and data.

This will work just fine and accomplishes the same task as quasi-associated signaling, which relies on *signal transfer points* (STPs) to relay the messages from the originating exchange to the destination exchange. ATM will not eliminate the need for SS7 networks, but it will change the protocols and add additional functions. *Signaling ATM adaptation layer* (SAAL) will eliminate *Message Transfer Part* (MTP) level 2, for example, on ATM links.

The STP will not disappear, but its role may change somewhat. The STP is still needed as a gateway into networks or even as a gateway into certain regions within a network. The STP continues to provide global title translation services as well as database access. Additional features and functions are likely placed on STPs to justify their existence.

There is no problem with sending all of the BISUP traffic through the ATM network and leaving the SS7 network for database access and other control functions. In fact, as the *Advanced Intelligent Network* (AIN) is deployed and implemented in existing networks, the traffic mix within the SS7 network will become predominantly based on *Transaction Capabilities Application Part* (TCAP) and *Signaling Connection Control Part* (SCCP) anyway.

BISUP has been included in this chapter because of its similarities with normal ISUP. The new protocol is explained in less detail than normal ISUP as the standards are still being defined.

ISUP Services

There are two types of ISUP services: basic and supplementary. Basic service provides the support for establishing connections for circuits within the network. These circuits can be audio circuits for voice transmission or data circuits for any digital information, voice or data. Supplementary services are all other circuit-related services, which typically encompass message transport after a call path is established.

In addition to the two types of services, ISUP uses two methods for end-to-end signaling. End-to-end signaling is the process of sending circuit-related information from one exchange to a distant exchange. These two exchanges may be adjacent to each other or across *Local Access Transport Areas* (LATAs).

The method currently used for passing signaling information to the distant exchange is called the *pass-along method*. With the pass-along method, the signaling information moves from one exchange to the next. All subsequent information related to the same circuit is then passed using the same path that was used to send the initial call setup information. This, of course, means that information must follow the same ISUP hops as the setup messages, which is not the most efficient method of routing.

The alternative method is called the SCCP method and uses the services of the SCCP protocol to route the message through the network. When using the SCCP protocol, the information does not have to follow the same path as the call setup information. In fact, it can follow any path, provided the final destination is the same.

The SCCP method uses true network routing and is probably more favorable for services that require information to be shared between exchanges when a call is in progress. However, this method is not used today.

The ISUP message provides important data regarding the service being requested of the remote exchange. These services are related to the circuit specified in the *initial address message* (IAM), which is the initial setup message used in this protocol. The receiver of an IAM then must determine if it has the resources necessary to provide the type of service being requested.

The IAM provides the distant exchange with the calling and called party numbers as well as information regarding the availability of SS7 signaling, whether or not the ISUP protocol is required end to end, and the type of network signaling available (if SS7 is not used throughout the network). The IAM also indicates whether or not further information will be available using subsequent messages.

The ISUP protocol uses the services of the MTP to send signaling messages from one signaling point to another. The *American National Standards Institute* (ANSI) standards and the ITU standards do allow for the use of SCCP services as well, although currently no applications in U.S. networks are available. The concept of using SCCP with ISUP is to allow end-to-end signaling without having to send messages to each intermediate exchange.

An example of how ISUP messages travel from one exchange to another is found in the diagram in Figure 9.1. This diagram shows that an ISUP setup message or IAM is used to connect both ends of the voice trunk between the originating exchange and the next exchange or tandem exchange. Once the connection is established, another

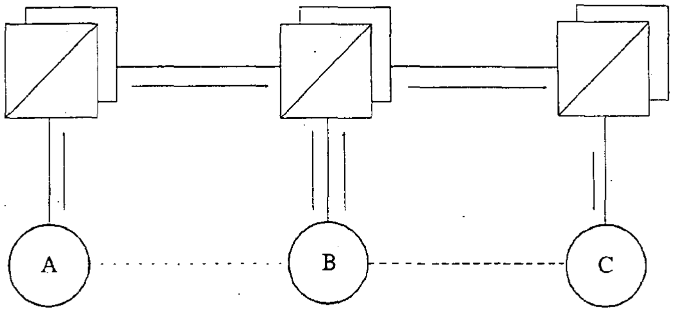


Figure 9.1 This figure depicts the path of ISUP setup messages for a typical telephone call. The first setup takes place from A to B. Exchange B must then initiate a circuit connection toward exchange C.

connection must be set up between exchange B and exchange C by sending another ISUP setup message (IAM) from exchange B to exchange C.

This continues through the network until there are voice circuits connected from end to end. End-to-end signaling using ISUP therefore requires many ISUP hops. If any message needs to be sent from one exchange to another exchange during the duration of the call, the same path is used.

The reason for this is due to the limited routing capabilities of the MTP. The MTP routing function only has knowledge of the adjacent signaling points and only provides node-to-node routing. The SCCP routing function knows the final destination of the message and is capable of providing routing based on the final destination, or *endpoint*, without having to know all the intermediate signaling points.

This would enable ISUP messages to travel through the network with a minimal number of hops and still be able to maintain association with a call in progress, without using the same path of the call setup messages. At this time, it is unknown if this feature will ever be implemented, although it is very likely that the *Intelligent Network* (IN) will find this a useful feature.

Another fundamental change in signaling with ISUP is the handling of the service tones used by the local exchange. Before ISUP and SS7, when a local call was placed to another exchange, the service tones (busy, ringback, and so on) were set by the distant exchange through the voice circuit to the calling party. With SS7, this is no longer necessary. In fact, the voice circuit does not need to be connected until the called party answers.

The service tones can be sent by the originating exchange. This is accomplished by the distant exchange sending ISUP messages indicating the status of the call (status information is implied within many ISUP messages, such as the *address complete message* [ACM]). For example, when the distant exchange receives an IAM, it will send an ACM in return. The ACM is used as an acknowledgment, and it also implies that ringing is being sent to the called party.

In most networks the service tones are sent by the destination exchange. The trunk circuits that have been reserved along the call path are not yet cut-through in both directions, but they are cut-through in one direction—from the destination exchange back to the originating exchange. This enables service tones to be sent in the backward direction to the originating exchange. If there is no answer or the call is disconnected for any other reason, the trunks can be quickly released in the one direction (see Figure 9.2).

If the calling party or the called party is using an ISDN interface, then the call setup and call status information can be much more complex than with service tones. In an ISDN call, the ISUP protocol is used to carry setup information as well as call status information through the PSTN and to the exchange. This is especially advantageous when the called or calling party is terminated at a *public branch exchange* (PBX).

When a PBX is used, the PBX can share status and setup information with the distant PBX. This is a feature never before possible with conventional signaling because conventional signaling was all analog and unable to support such a broad base of information.

Even information about the station users within a PBX, such as class of service information and dialing privileges, can be shared through the PSTN to the distant PBX through the use of the ISUP protocol. Large corporations with multiple PBXs can enjoy the benefits of a large network without leasing private lines between PBXs (see Figure 9.3).

The voice circuit in an ISUP call setup is not actually cut-through (connected) in both directions until the called party goes off-hook and answers the call. The voice circuit is reserved for the call and can even be tested before the setup begins. However, the connection is not made in both directions until the called party answers the call.

The most commonly asked question, then, is how does SS7 save time in call setup? The answer lies in the speed of the digital call setup and teardown versus analog signaling.

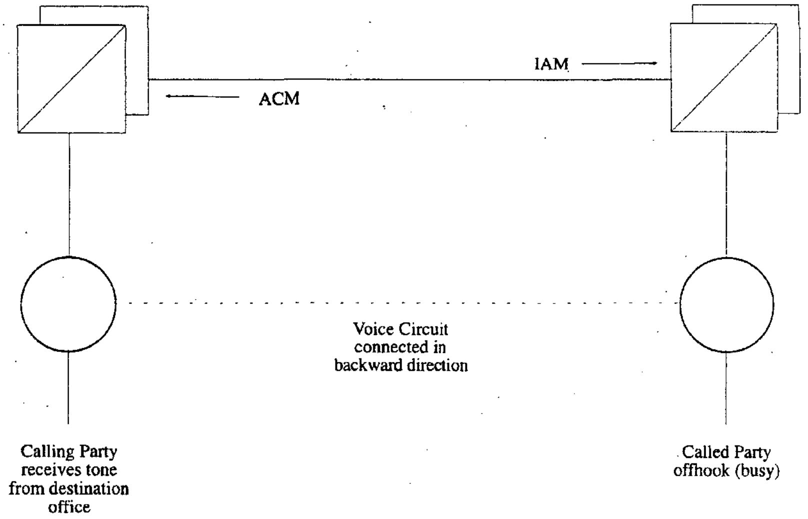

IAM
ACM
Voice Circuit connected in backward direction
Calling Party receives tone from destination office
Called Party offhook (busy)

Figure 9.2 In this figure, the voice circuit is reserved but not cut-through. Service tones are sent by the local office instead of the remote exchange.

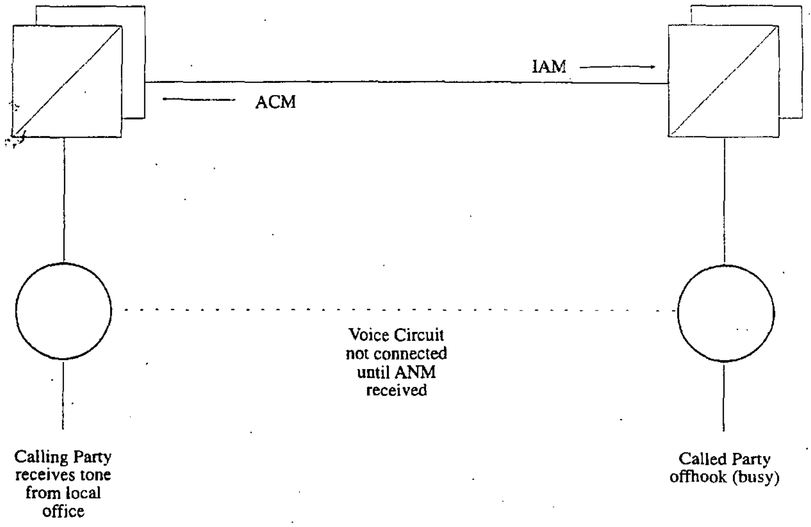

IAM
ACM
Voice Circuit not connected until ANM received
Calling Party receives tone from local office
Called Party offhook (busy)

Figure 9.3 The ANM message is used to trigger cut-through on the voice trunk when the called party is within the United States. If it is an international call, the trunk is cut-through upon receipt of the ACM message.

Digital messages travel at high speeds, which enable calls to be set up much quicker than analog signaling. If you don't believe this concept, pick up a phone and dial a number; then time how long it takes for the call to connect. Now compare the setup time with a call five years ago, before SS7 was deployed throughout the network. Many times, ringback is heard as soon as the final digit is dialed.

When a called party is not available because of a busy condition, the reserved voice circuit is released and is immediately available for another call. In the meantime, a digital message is sent back to the originating exchange notifying the originating exchange of the busy condition. The originating exchange then sends a busy tone to the calling party.

While the originating exchange is sending a busy tone to the calling party, the voice circuit has been released (between the exchanges) and only the circuit between the calling party and the local exchange is maintained.

There are many other advantages of using ISUP and TUP for call setup and teardown. If you understand the basic concepts of how a call is set up and then released, the advantages become much clearer. In the next section, we will talk about how a call is set up in greater detail, providing examples of different call setup situations.

Call Setup and Teardown

To understand how ISUP works and its advantages, we need to first understand the basics of how a call is set up and released in the SS7 network. The ISUP protocol is used to accomplish this. These examples will assume there are analog lines being used between both parties and the telephone company. We will later examine the procedures used with ISDN circuits.

When a caller lifts the receiver, the local exchange (exchange A) determines that the caller is off-hook by the presence of a current on the subscriber line interface (DC signaling). The local exchange acknowledges the presence of a loop current by sending a service tone (dial tone) to the calling party.

The calling party then dials digits, which signals to the local exchange the address (telephone number) of the distant called party. The local exchange must wait until all digits have been dialed and then examine the first three digits dialed to determine if the calling party dialed an area code or a prefix (determined by the North American Numbering Plan), as shown in Figure 9.4.

If the exchange determines that the number dialed was a long-distance number, the call is routed to a long-distance carrier through a *point-of-presence* (POP) in the LATA of the calling party. The prefix and the subscriber number (the last four digits of the telephone number) are then routed to the distant exchange.

The local exchange determines how it will connect this call based on information in its trunk routing tables. These routing tables identify which voice circuits to use to establish an end-to-end circuit with the least number of hops. When it determines which voice circuits to use, a call setup message is created and sent to the exchange that will provide the first voice connection (exchange B).

This exchange may not be the final destination for this call. In fact, it may be a tandem that is used as an intermediate switch to reach the final destination. Intermediate tandem switches are used to prevent all exchanges from needing to have voice circuits to all other exchanges within any given LATA.

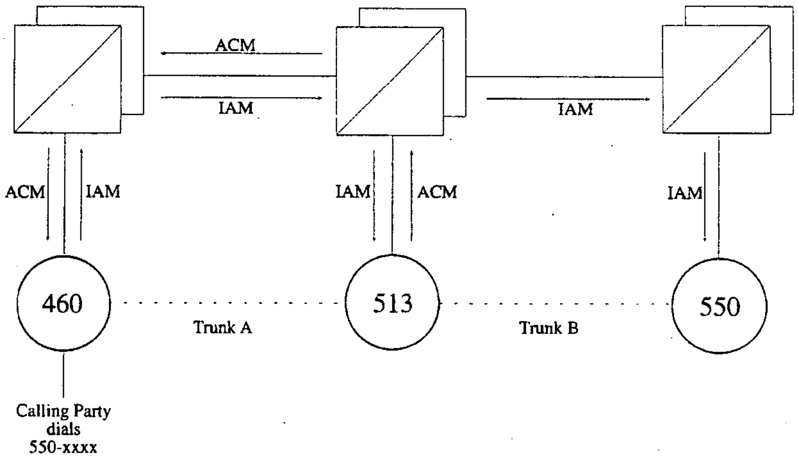


Figure 9.4 When the calling party finishes dialing digits, the local exchange determines which trunk needs to be reserved for the call. An IAM is then generated toward the first exchange. When the ACM has been received, the local exchange begins delivering the ringback tone, even before the called party's telephone begins ringing in some cases.

The call setup is sent using the ISUP protocol through the SS7 network. The STP serves as a network router for these messages by simply routing SS7 messages to their proper destination and does not play any significant role in setting up the voice circuits. In general, the STP has no real knowledge of the ISUP message; it only delivers it to the proper exchange.

The IAM is created by the local exchange (exchange A) and sent to the intermediate tandem exchange (exchange B). All the information necessary for the tandem exchange to establish a connection can be found in this IAM. The IAM does not contain information for the final destination as it is not establishing a connection from the originating exchange A to the destination exchange C.

The tandem exchange will acknowledge receipt of the IAM by sending an ACM to the originating exchange A when it has received an ACM from the destination exchange. This indicates that the circuit designated as reserved in the IAM has been reserved at exchange B, and a connection can be made when ready.

The tandem exchange can begin setting up the next circuit between itself and the destination exchange C. This is accomplished by generating another IAM, including the called and calling party address information provided by the originating exchange A and sending the IAM to the destination exchange C.

The IAM will also specify the signaling method to be used for this call. If the IAM specifies that the ISUP protocol is to be used end to end (required), then the call must be set up using the ISUP protocol. If the tandem exchange (B) cannot set up the call using ISUP (in the event the exchange does not support ISUP to the destination or there are no facilities available that use ISUP), the call is rejected and a message indicating the reason for rejection is sent back to the originator.

If the IAM indicates that ISUP is preferred, the receiving exchange will check for available resources to determine if the call can be set up using ISUP. If not, the call is still set up, but using a different method, such as *multifrequency* (MF) signaling or the TUP protocol.

The IAM can also indicate that ISUP is not required "all the way," in which case the call is set up using whatever method is available. These methods could be ISUP, TUP, or a non-SS7 signaling method such as MF (discussed in Chapter 1, "Signaling System #7").

An intermediate exchange (such as the tandem in our example) can change some of the information in the IAM. The first six digits of the called party number, information regarding the connection (the nature of connection), and the end-to-end method indicator can be modified. All other fields are passed transparently to the distant exchange.

As is sometimes the case, exchange C does not have to be the final destination. There can be more exchanges or tandem exchanges required when establishing this end-to-end voice circuit, but for simplicity we will only discuss three connection points.

Upon receipt of the IAM, the distant exchange must examine the message to determine if there will be any further information in subsequent messages. If not, the called party number is examined and the exchange determines if the called party is available or busy. If the called party is busy, a *release message* (REL) is sent to the originator and the circuit is immediately released for another call.

The distant exchange may find that the called party number is not included in the IAM. When this occurs, the distant exchange (exchange C) must request the called party number using the protocol services specified in the IAM. Two methods can be used: end to end (SCCP services) or link to link (pass-through using MTP). Currently, only the pass-through method is used in ANSI networks.

If the called party is not busy and the call can be accepted, an ACM is sent to exchange B. Exchange B then sends an ACM to the originating exchange. The distant exchange (exchange C) then signals the called party that there is a call by sending ringing to the called party on its subscriber line (DC signaling).

No message is returned until the called party answers the phone. When exchange C determines that the called party has lifted their receiver by detecting loop current (DC signaling) on the subscriber interface, an *answer message* (ANM) is generated and returned to the tandem exchange (exchange B). The same path used to send the IAM from exchange B to exchange C is used for the ANM. This means that the same links and the same STPs are used for all associated ISUP messages. The voice circuit between exchange C and exchange B is immediately cut-through when exchange B receives the ANM.

When the tandem exchange receives the ANM, it sends an ANM to the originating exchange A using the same path on which the IAM was received. The originating exchange can then begin cut-through on the voice circuit between itself (exchange A) and the tandem exchange (exchange C).

Once the voice circuit is connected, conversation can begin and no messages are necessary through the SS7 network until either party goes on-hook. There are some features associated with class and some other calling features that may require exchanges to share information with each other during the duration of the call. Currently, the communications is handled using the same path as the setup messages. However, the standards do allow the use of SCCP to carry such information from one end to the distant end without following the call setup path (see Figure 9.5).

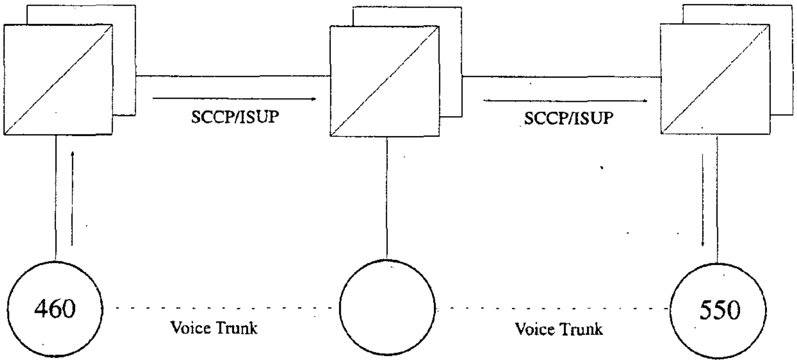


Figure 9.5 If SCCP services are used for ISUP, the intermediate exchange can be bypassed and ISUP messages can be sent directly to the remote exchange. This applies only to messages between two exchanges after a connection is established and enables the switches at both ends of a call to exchange status information.

The previous discussion applies to normal routing procedures. *Local Number Portability* (LNP) has changed the way calls are routed. The ISUP protocol has been modified to provide additional information needed to route calls to numbers that have been *ported*. Numbers are considered ported when the subscriber changes from one local service provider to another, keeping his or her telephone number. In the past, you had to give up your telephone number and obtain a new telephone number from the new service provider.

With LNP, calls are no longer routed based on the digits dialed. Each call made to an NPA-NXX that has a number ported in it requires a database transaction to obtain additional routing information. The database will identify whether or not the dialed number has been ported and, if so, where to route the call.

For routing purposes, each end office (and tandem) is assigned a 10-digit *location routing number* (LRN). The LRN is usually the same NPA-NXX currently assigned to the end-office switch, with all zeros for the last four digits. For example, the end office serving (919) 460-xxxx will be assigned the LRN of (919) 460-0000.

The called party number field in the ISUP IAM will now contain the LRN if the dialed number has been ported to another carrier's network. The dialed digits will be placed in the *generic access parameter* (GAP). Before a call can be routed, the end office must first access the LNP database within its region and obtain the LRN for the dialed number. For more information about LNP, see Chapter 10, "Local Number Portability."

If the digits dialed are for an international number, the call setup sequence is the same. The only difference is the usage of the international network to route the call. In the international plane, there is no knowledge of the dialed digits other than the country and city code. The STPs at the international plane route messages based on their international country and city code, which is used by the gateway STP to determine how to route the call within its own network (see Figure 9.6).

The addressing of SS7 entities is virtually the same format. The primary difference between national and international addressing is the point code structure of each SS7 signaling point. The ITU-TS standard calls for a 14-bit point code structure that is divided as 3-bit zone identification, 8-bit area/network identification, and 3-bit signaling point identification.

National point codes follow a simple 14-bit point code and are usually found to have no divisions. The exception to this rule is in the case of the ANSI point code structure, which uses a 24-bit point code, with a network identification, cluster identification, and member identification (refer to Chapter 6, "Message Transfer Part Level 3").

The call setup messages must be routed according to the dialed digits (or the LRN as in the case of LNP), which must then be translated at some point to a point code. The point code translation within the national plane is usually that of a gateway STP, which then translates the point code into an international point code (14 bit) and routes the message to the proper gateway STP according to the country code dialed.

When a gateway is used to route a call setup message into another network, the *exit message* (EXM) is used to indicate that a call connection has been completed in the other network. The EXM may include the outgoing trunk group number used to connect the voice circuit to the other exchange, although this information is optional.

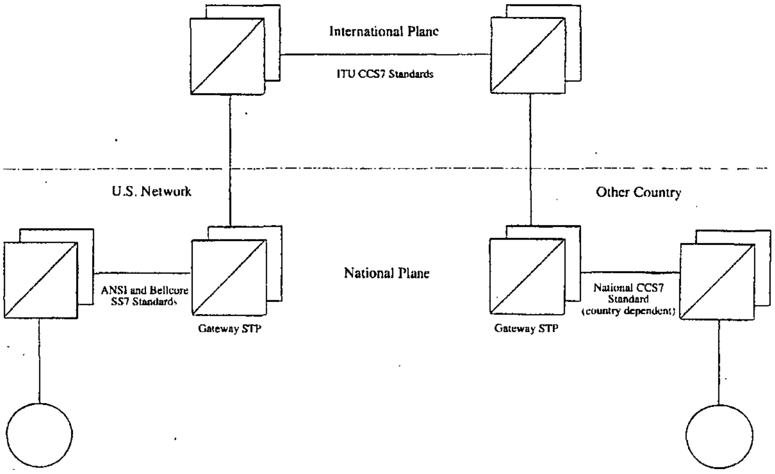


Figure 9.6 International calls rely on the international SS7 network. This network uses *International Telecommunications Union—Telecommunications Standardization Sector* (ITU-TS) SS7 standards, which are somewhat different from the national standards. The national standard used in the United States is ANSI and Bellcore standards used within the *Bell Operating Company* (BOC) networks.

The voice circuit in the case of an international connection can be connected as soon as the ACM is received. This depends on the network and may or may not be true. In theory, the trunk does not have to be cut-through until after the called party answers the call, but this does not imply the actual practice.

When taking a call down and releasing all circuits associated with that call, several steps must take place. When either caller goes on-hook, the subscriber line is released immediately (DC signaling). The exchange then sends a *suspend message* (SUS) to its tandem or whichever exchange it is directly adjacent to (exchange B in Figure 9.7).

When the adjacent exchange (exchange B) receives the SUS, it sends an SUS to its adjacent exchange. This continues through the network until the SUS has reached the originating exchange. As soon as the calling party returns to an on-hook condition, the REL is sent toward the distant exchange.

When an exchange receives an REL, it returns a *release complete message* (RLC) as an acknowledgment. The RLC indicates that the circuit has been returned to an idle condition. The tandem exchange in the diagram must then generate an REL to its adjacent exchange (exchange A in Figure 9.7) and follow the same procedure to release its circuit. Exchange A in this example would then generate an RLC. Upon receipt of the RLC, exchange B would then release its circuit.

While all this is going on, the circuit between exchange B and exchange C has been released and could be set up for another connection associated with a different call. With conventional analog signaling, this would not be possible; the circuit would remain seized until the distant exchange was ready to release. This often would result in "hung" circuits, especially when the calling party did not hang up his or her phone.

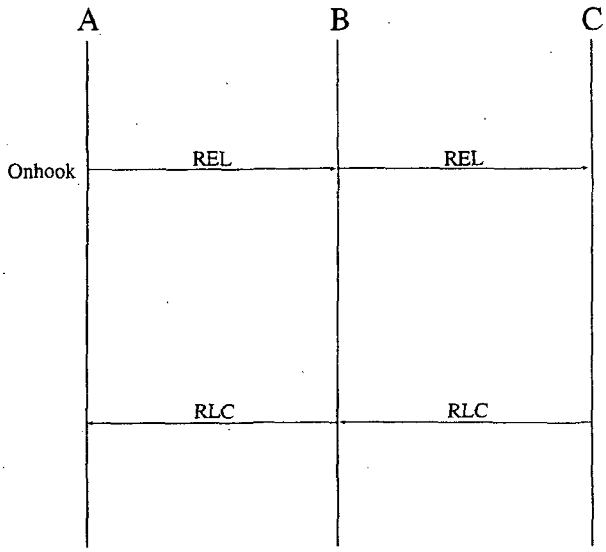


Figure 9.7 Normal call release procedure.

Call Setup and Teardown of ISDN Circuits

The ISUP protocol was intended for use with digital subscriber interfaces such as ISDN. The intent was to provide a protocol with more versatility, enabling the exchange of status information and other forms of circuit data to be exchanged from the local ISDN network to the distant ISDN network.

If distance and other factors prevent ISDN networks from being connected to one another, SS7 can be used to bridge the gap. The ISDN protocol is transparent to SS7 because there is direct mapping from the ISDN parameters to the ISUP protocol. This makes interworking very easy while still maintaining the security of the network.

The procedures for setting up and tearing down an ISDN connection are somewhat simpler than those for conventional signaling or analog subscriber circuits. The ISDN messages are compatible, but somewhat different than those used in ISUP. Figure 9.8 illustrates how ISDN and ISUP protocol messages map to one another, showing a typical call setup and teardown.

In the ISDN circuit, a SETUP message is sent from the originating (calling) party to the local exchange. The local exchange telephone switch is capable of interpreting ISDN messages at the line circuit level and converts this into an SS7 IAM. The IAM contains the same information as the SETUP message. In fact, there is direct mapping from the ISDN protocol to the SS7 protocol, making them compatible interworking protocols.

The IAM is then sent in the forward direction to the next exchange. As seen in the drawing, there may be several exchanges involved in order for the voice circuit to be connected end to end. The IAM must be sent between each exchange and acknowledged

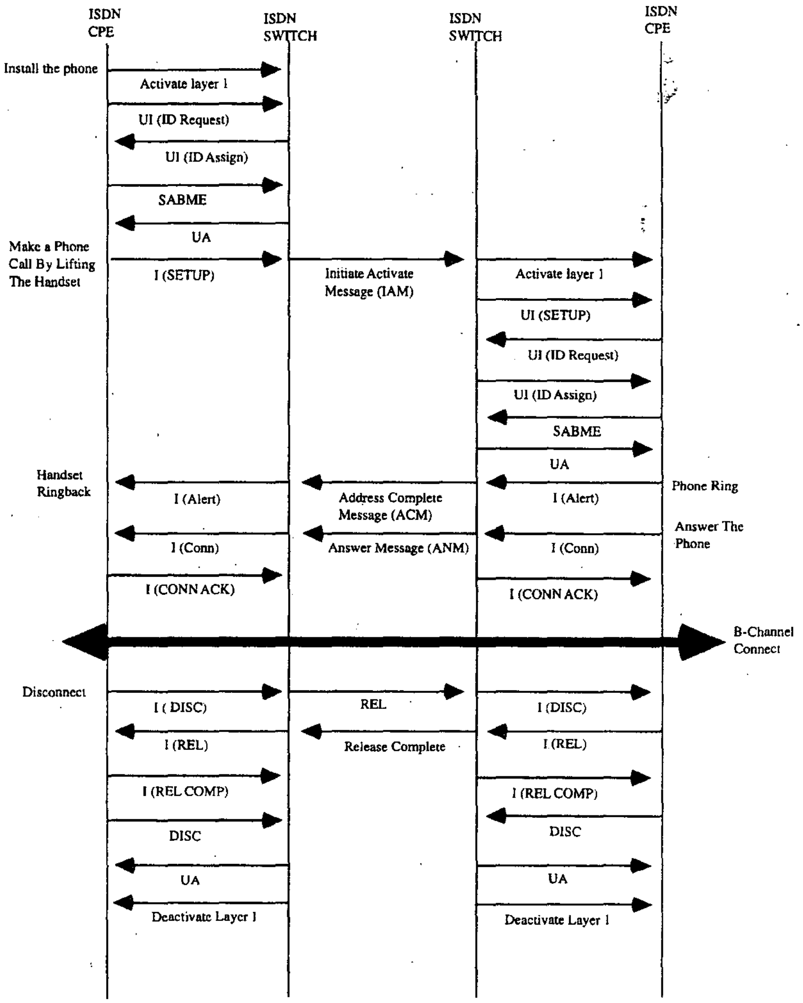


Figure 9.8 This figure illustrates how ISDN signaling maps to SS7 signaling. Backward and forward setup messages are used to exchange additional information regarding a call between two exchanges.

between each exchange. Additional information may be sent between exchanges before the circuit has been completed using end-to-end signaling.

When the distant exchange has been reached, the IAM is converted back into an ISDN setup message. The setup message is then sent to the called party device (either an ISDN network terminal or an ISDN PBX). When the called party has accepted the setup message, the called party returns an ISDN alerting message. This message indicates that all the addressing signals have been received and the ISDN terminal (telephone) is now ringing. No ringing generator is sent from the local exchange because

ISDN is digital. The ringing is generated at the telephone device itself based on the receipt of the setup message.

When the exchange receives the alerting message, it is converted into the SS7 ACM, which indicates that the SS7 exchange has received all of the addressing signals and the called party is being signaled. The ACM at the originating exchange is converted into an alerting message.

When an ISDN terminal receives an alerting message, it indicates that the called party is being signaled. No further action is necessary, although charging may begin after receipt of the ACM. This is optional and not widely acceptable in most cases.

When the called party answers, a *connect message* (CON) is sent in the backward direction. The SS7 equivalent to the CON is the ANM. The ANM is passed through each of the exchanges until the originating exchange is reached, where the ANM is converted back into a CON.

The CON indicates that the called party has answered, and it usually triggers cut-through on the voice circuit (this is network dependent). The ANM is usually used in the SS7 network to trigger billing.

When the call has been completed, either party may terminate the call by hanging up. When either party hangs up, the ISDN network creates a *disconnect message* (DISC). The local exchange receives the DISC and converts it to an REL. The REL is sent to the next exchange, which acknowledges the REL with an RLC. The RLC triggers the actual release of the voice circuit. Note that this continues on through the network between each of the exchanges until all of the circuits have been released. As soon as a circuit has been released, it is immediately available for another call. It is quite possible that the local loop can be released and already connected to another call before all the circuits from a previous call have been released end to end, although timers are usually used to prevent this from occurring and creating troubles.

Call Setup and Teardown of BISDN Circuits

BISDN brings on many challenges for the ISUP protocol. There are different addressing schemes used in BISDN that have support of virtual paths as well as virtual connections within a virtual path. The ISUP protocol does not support this addressing and has been modified as BISUP.

The standards for BISUP are still evolving, although the first Bellcore publications have been released and included in the GR-246-CORE. In this publication, several new sections have been added describing the services and procedures for BISUP.

The ANSI standards for BISDN signaling are the same as those formulated by the ITU-TS, which is good news for all of those in international networks. This means simpler provisioning and implementation. This section will define the message types and procedures used to set up and tear down broadband circuits.

Before a circuit may be assigned for a call, it must be determined which signaling point will be responsible for the assignment of bandwidth and virtual path/virtual channels for a given circuit. One of the inherent problems in broadband is the possibility for *glare*, or *dual seizure*. This occurs when two signaling points assign calls to the same virtual path/virtual connection combination. For this reason, end nodes will share the responsibility of assigning these characteristics 50-50.

All odd-numbered *virtual path connection identifiers* (VPCIs) are the responsibility of one signaling point, whereas even-numbered VPCIs are the responsibility of the other signaling point. This prevents the possibility of glare, or dual seizure. The exchange with the highest point code will be responsible for even-numbered VPCIs.

In the event that there are no available VPCIs controlled by an exchange, a setup may be issued for a VPCI that is not controlled by the exchange. In this event, the virtual path and virtual connection ID are not provided (these parameters are found in the connection element identifier parameter, which is normally included in the setup message).

If there is an incoming call on a circuit controlled by the receiving exchange, the connection element identifier is not included. It is a request for service, and the controlling exchange must assign the virtual path and virtual connection. If this information has not been provided, then it must be assumed that the originating exchange does not have any available virtual paths/virtual connections within its control and needs assignment from the other half.

When assigning a circuit, the exchange must determine what bandwidth is necessary for the call. This is determined by reading the parameters in the setup message. Once it is determined, the exchange assigns the appropriate virtual path connection within its control.

As previously mentioned, each exchange is responsible for the assignment of virtual path connections and bandwidth for one-half of the available virtual path connections. If there is not enough bandwidth available at an originating exchange, it sends a request to the remote exchange (for which it wants to establish a connection) without any virtual path connection information.

The receiving exchange then determines if there is ample bandwidth available for the connection and, if so, provides the virtual path connection information to the requester. If there is not enough bandwidth available, then the request is denied using the REL with the cause "cell rate not available."

Once an exchange has received all of the information necessary to establish a connection, the receiving exchange must determine if the call is to be routed to another exchange (intermediate exchange) or if the call is to be terminated within itself. If the call is to be routed through another exchange, then normal ISUP routing is duplicated.

Normal ISUP procedures call for the same setup procedures used to establish a connection to the first exchange, which must be repeated for all subsequent exchanges and be used to reach the final destination. This is also true in the case of BISDN.

Routing information may be stored within the exchange itself or in a central database accessible by all exchanges. The latter is becoming true in wireless networks, where the *home location register* (HLR) used to store location information as well as subscriber information is moving to a central location, which is accessible by all other wireless providers.

The TCAP is then used to access the centralized database for additional routing instructions before call setup procedures can begin. This allows for fewer resources within the exchange and optimizes the individual service providers' network.

Several parameters are used to determine the best route for a connection. The called party address is the most useful, but the broadband bearer capability and the ATM cell rate must also be considered. The ATM cell rate determines which interoffice facility will need to be used to get the connection through the PSTN.

So far, we have only discussed procedures for assigning virtual path connections between two exchanges. The message structures for these procedures are virtually the same as those for normal ISDN signaling. The exception is the addition of new parameters that specifically support BISDN and some new message types that augment existing message types.

When service is requested, the IAM is sent to the remote exchange. This IAM is almost identical to the one we talked about in the previous sections, but it has additional parameters supporting BISDN and ATM (see the section "ISUP Parameters").

Preceding the BISUP IAM parameter is the routing label, which may or may not specify the actual ATM circuit information. According to the rules discussed previously about controlling exchanges, the IAM may act as a request. The receiving exchange may have to provide the circuit identification.

If this is the case, then an IAM acknowledgment will be sent to the originating exchange providing the circuit identification. This is a new message type for ISUP and is only used in the BISUP protocol.

Once the circuit identification has been provided, an ACM can be returned by the originating exchange, which serves as an acknowledgment that the addressing information has been received.

Other than additional or different parameters in the various message types, the sequence of messages is almost identical to normal ISUP. The major changes have to do with the assignment of the actual circuit, which is split between the two exchanges. Also keep in mind that these are logical connections, not physical connections as in *Plain Old Telephone Service* (POTS). This complicates the procedures somewhat and expands the message sizes exponentially.

Message types and parameters for BISDN support are found at the end of this chapter and are labeled as such to differentiate them from normal ISUP messages and parameters. These are still evolving and what is published in this book represents the first version as published by ANSI and Bellcore.

Interworking with Non-SS7 Networks

Although most all of the PSTNs now use SS7 throughout the network, some segments are still using conventional signaling. Conventional signaling in most of these cases consists of MF signaling.

SS7 must be able to function and internetwork with networks using conventional signaling. One of the primary issues is the reservation of a voice circuit. Unlike conventional signaling, the voice circuit does not get connected until the distant party answers or at least until both exchanges have sent and received all the addressing information required to connect the call.

In MF signaling, the circuit is connected when the calling party completes dialing the digits and is used to signal the distant exchange. Therefore, some method must be established for enabling voice circuits to be reserved and tested in conjunction with the SS7 protocol.

To accomplish this, a circuit reservation procedure is used. Prior to sending an IAM, which is used to send addressing information necessary to establish the circuit connection, a *circuit reservation message* (CRM) is sent to the non-SS7 exchange. This is then converted to MF signaling by that exchange and MF signaling may be used from that point on. The exchange acknowledges receipt of the CRM by sending in the backward direction (back to the originating exchange) a *circuit reservation acknowledgment* (CRA) message.

The originating exchange can then specify a continuity test to be invoked by the exchange using MF signaling by sending a *continuity check request* (CCR) message in the forward direction. This message invokes a loopback test at the remote exchange on the voice circuit. The results of the continuity test are then returned to the originating exchange using the *continuity test* (COT) message.

After receipt of the COT message, if the continuity test proved successful, an IAM is sent by the originating exchange to begin the normal call setup procedure. The voice circuit is reserved even though MF signaling is used and is ready for cut-through end to end when the calling party answers or the address information has been successfully passed from the originating exchange to the destination exchange, depending upon network deployment.

Interworking with MF networks is no longer efficient with LNP becoming commonplace nationwide (and in the future worldwide). When the MF network is reached, any contents of the ISUP IAM are lost, with the exception of the dialed digits. The LRN found in the called party address is lost and the dialed digits found in the GAP parameter are sent via MF to the distant or adjacent network.

If the adjacent network is equipped with SS7, it must then perform a database query to regain the lost routing information. This, of course, is redundant and adds unnecessary delay to the call setup. It also places an additional burden on the network resources. This can be of major concern when you consider the number of database queries the adjacent SS7 network is responsible for making.

Consider this scenario. The adjacent SS7 network will be receiving scores of calls from the MF network. The SS7 network will have to perform database queries for each of those calls, even though other networks have already done this. The burden lies on the SS7 network to provide this service when the originating network should have been the responsible network.

Circuit Testing

The SS7 network provides several mechanisms for testing circuits and switches remotely. This testing is usually performed from the *Operations Support Systems* (OSSs) located regionally within the network. The tests can also be performed locally by testboard technicians or maintenance center technicians.

The OSS is an operations and maintenance center that allows complete network monitoring and testing. These are fairly new (within the last 10 years) and have been deployed within regional areas of the network.

Before SS7 and automation on the network, testing was conducted using testboard positions at every exchange. These test positions were capable of connecting to every circuit entering and leaving the central office and allowed technicians to test the continuity, capacitance, and other properties of the circuit.

Today, these testboard positions have disappeared, and all the testing has been moved to remote locations where many exchanges can be tested by one maintenance center. The SS7 network is used for passing those maintenance and test messages through the network to the remote switches.

The ISUP protocol also provides a means for testing circuits as well as translations in various nodes. A translation is a routing instruction, which translates dialed digits into a routable address, such as a signaling point code. This section describes two of those tests: the continuity test and the circuit validation test.

Continuity Testing

Because SS7 uses a separate facility than the voice circuit for sending information to another exchange, there is no knowledge of the operational status of the voice circuit. Most voice switching systems today provide some level of circuit testing and fault isolation, providing alarms when a circuit fails. This alarm information enables diagnostics software to "busy out" the trunk, preventing calls from being routed to the failed circuit.

In many networks, however, digital circuits are used to carry the voice and data. These digital facilities are usually DS1 or DS3 facilities, which require a series of multiplexers. For example, connecting to the voice switch is a DS0 circuit (64 kbps). This DS0 is sent through a multiplexer, which aggregates 24 DS0s and groups them into one DS1. The DS1 signal is then sent with other DS1s to another multiplexer, which aggregates 28 DS1s into one DS3. The DS3 is then used to reach another exchange where these signals must be demultiplexed back down to their original DS0s.

In order for these circuits to reach the proper switch, they are crossconnected through digital crossconnect systems, which enable incoming circuits to be electronically routed to the appropriate switch in the central exchange. This only adds to the problem of circuit testing and diagnostics because the alarm and circuit status information gets lost in the multiplexing and crossconnecting and never makes its way through to the rest of the network.

To counter this issue, SS7 provides the capability to test the voice circuit before connecting a call to it. This takes place even before the IAM is sent. The test is known as the continuity test and uses the COT. There are two instances when the continuity test procedure may be used: within the same network or when connecting to networks that use *Exchange Access Signaling* (EAS).

EAS uses a series of tones to indicate signaling. This method has been replaced in most instances in the United States by the SS7 standard of signaling, but it may still exist in some rural areas. Because EAS does not support the digital messages of SS7, a conversion must be made converting the SS7 message into the analog format of EAS signaling.

The exchange that will perform this conversion will use both EAS and DC signaling (a method called the *wink*, where the polarity of the trunk is temporarily reversed and then returned to its original polarity as an acknowledgment).

When it is determined that a continuity test is needed before a call can be set up and the call is to another exchange using EAS or some other method of signaling other than SS7, a reservation message is sent to the adjacent exchange (exchange B). The CRM enables the voice circuit to be reserved without actually connecting the circuit. A request for a continuity test is sent in the CRM. The acknowledgment expected is the CRA.

The CRM also contains information regarding the nature of the connection to be established. This information includes satellite requirements (if any) as well as information regarding the usage of echo cancelers and end-to-end ISUP. The nature of connection parameters is also used in the IAM, which follows the continuity test.

After the circuit has been reserved, the COT message is sent via SS7. The COT message indicates to the adjacent switch that the voice circuit should be tested for continuity using conventional testing methods (usually a loopback test). The results of the COT test (if successful) are carried in the IAM.

In the event that the circuit fails the continuity test, the circuit is released and a COT message with "test failed" is returned to the originating exchange (originating meaning the exchange that requested the continuity test to be performed). Another circuit is selected and the procedure begins again on the new circuit.

As shown in Figure 9.9, the COT is sent in the same direction as the IAM. When the voice circuit is tested, the requester of the message will determine whether or not the test was successful. This is done by sending a signal or current on the indicated circuit and, when the adjacent switch performs a loopback, by receiving the same signal back at the originating switch. If the signal is the same as what was sent, the continuity test is said to be successful.

The continuity test is not required on every call. Its usage is network dependent. Some networks will perform a continuity test on every circuit selected before every call, even when SS7 is used throughout the network. Other networks will perform a continuity test on any one circuit every 100 calls. The network operator must configure the voice switch to perform the continuity test at whatever intervals are used in their network. An STP has no knowledge of these tests as its function is to pass the information along to another *service switching point* (SSP). This is an SSP function only.

Circuit Validation Test

The circuit validation test is typically used when a new facility or translation is added to the network. The purpose of this test is to validate the translation data and ensure that circuits between two exchanges can be selected by the routing function properly.

The translation is tested locally first to validate the local routing entry. Once maintenance personnel has verified the circuit can be accessed locally by *Common Language Location Identifier* (CLLI) code, the technician generates a *circuit validation test message* (CVM) to verify whether the distant end can route to the new entry using the new translations.

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However, the connection must already be established and the information must be related to that call's circuit or services.

Any information about the subscriber or network features or anything that is not directly related to the circuit itself uses the TCAP. This protocol was established (as described in Chapter 8, "Overview of TCAP") for noncircuit-related messages.

The type of information provided by the ISUP protocol includes resource requirements for completion of the connection (resources such as ISUP all the way through the network and echo cancelers on the voice circuit). Bandwidth information and service information (call waiting and call forwarding, for example) are service related and require that information be sent between both end-to-end exchanges.

Intermediate exchanges do not need to see this information as it does not involve their interaction. The objective of the intermediate exchange is to provide the connection through their facilities to the next exchange until an end-to-end path is established for the bearer traffic.

ISUP Services

The ISUP protocol provides two methods for reaching the end destination. As mentioned in the previous discussion about ISUP functions and call setups, two types of services are provided by ISUP: basic and supplementary services.

In addition to these two types of services, there are two ways to reach the end destination. End-to-end signaling can use either the SCCP method or the pass-along method. Even though these have been defined already, we will repeat them here as a reference.

Basic Service

Basic service is defined as the setup and teardown of circuits in the telephone network. These circuits are used for voice, data, and video transmission. Currently, most networks use some form of digital transmission for all transmission, regardless of the source.

This digital transmission is now being further enhanced by the addition of fiber optics into the network. Basic services will still be used for setting up and tearing down these connections as well.

As broadband technologies such as ATM and BISDN are deployed, basic services will be used to control these circuits. The protocol does not care what is being transmitted, although it does carry some indication of the source and the type of transmission being carried.

Supplementary Service

Supplementary service is defined as all of the other services needed to support these circuits. Other services may include sending caller information from one endpoint to another while a call is in progress. This information may be feature or caller related.

The IN relies on supplementary service to send information about established calls. This is different than the usage of TCAP because TCAP is not circuit related.

End-to-End Signaling

End-to-end signaling is defined as signaling information that must be sent from the originating exchange to the final destination exchange. This information may be part of basic services or supplementary services.

End-to-end signaling will almost always involve intermediate signaling points even though they are not concerned with the call itself. There are two ways to reach the final destination exchange for end-to-end signaling. These are explained in the following text.

SCCP Method

SCCP can be used to provide network (layer 3) routing for messages, but it is not used in today's U.S. networks. The use of SCCP for end-to-end signaling would be an enhancement to the current method, however. The SCCP protocol enables ISUP messages to be routed to the distant exchange using any route, but only for messages related to a circuit already connected end to end.

The method currently used requires the ISUP message to use intermediate switches, using the same path as the messages used to set up the circuit associated with the information being sent.

Pass-Along Method

The pass-along method is widely used in today's network. This method uses the same path as the setup messages. This method works, but requires messages to make unnecessary stops at intermediate switches.

This is not necessary because the information does not concern the intermediate switch. It would be much more efficient if the message could be sent directly to the exchange through STPs by using any available route.

Message Format

The ISUP protocol uses message types to indicate the type of message being carried as well as the format of the message (see Figure 9.10). Each message type has a distinct format that has mandatory and optional parameters. The parameters depend on the message type.

As in the TCAP protocol, the ISUP uses mandatory fixed parts and mandatory variables. These are parameters that must always exist, depending on the type of ISUP message. Again, the parameters used depend on the message type.

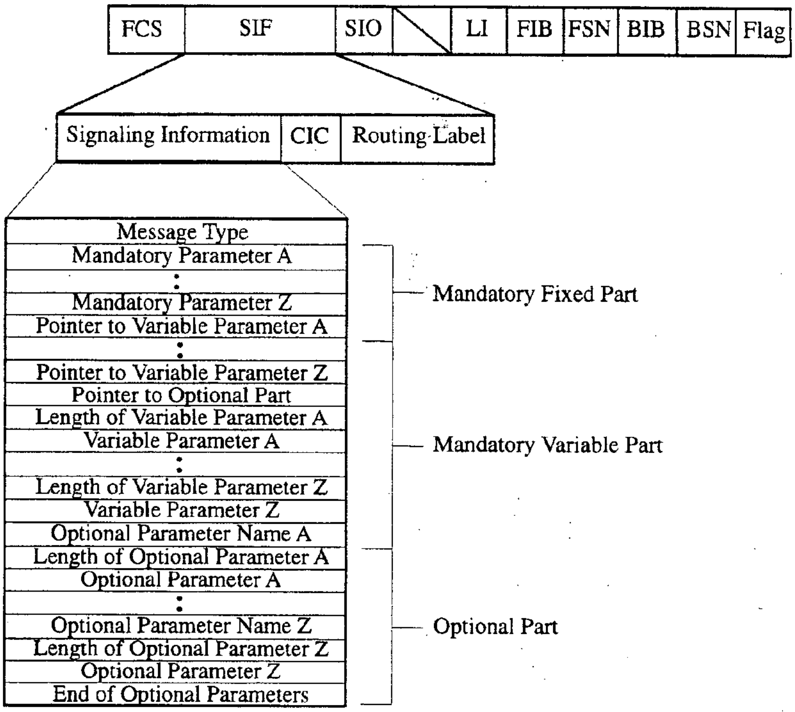


Figure 9.10 This figure shows the components used in an ISUP message.

Circuit Identification Code (CIC)

The *circuit identification code* (CIC) identifies the circuit that is being set up or released (see Figure 9.11). The CIC may be a voice trunk or any other transmission medium in the PSTN.

Currently, there are no defined standards for allocating circuit identifiers. These are determined by an agreement between the telephone companies. The CIC is provided to the originator of the ISUP message (SSP) by the end switch.

The end switch may be incorporated into the SSP as many of these systems are fully integrated. This means that a voice subsystem provides the switching functionality for the voice circuits, while the SS7 subsystem provides all the circuit control.

Message Type Codes for Normal ISUP

This is a one-octet field that is used to define the action that will be taken by the exchange. In addition, the message type also implicitly defines the message structure. The parameters used will depend on the message type (see Figure 9.12).

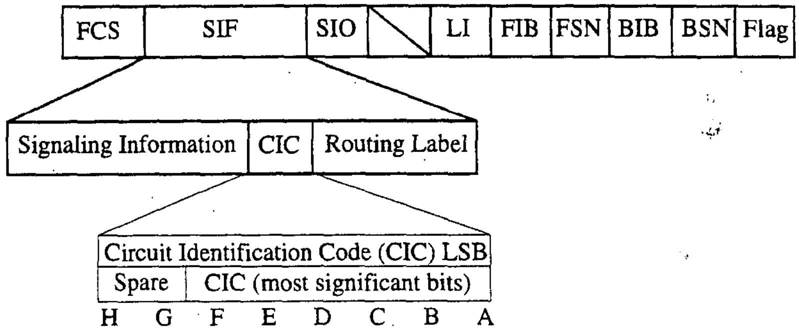


Figure 9.11 The CIC is used to identify the trunk circuit that will be connected and associated with this message. The CIC is not used in broadband services. Instead, virtual paths and virtual connections are identified in the BISUP message content.

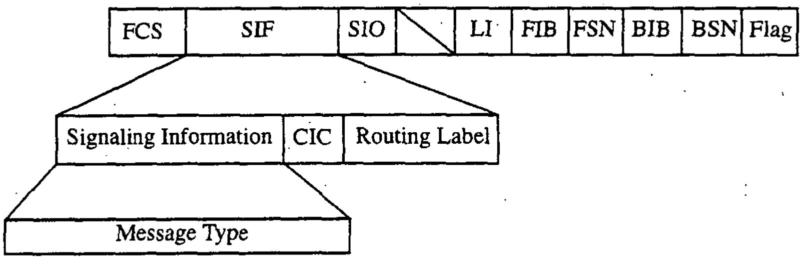


Figure 9.12 As in TCAP and SCCP, the message type field identifies the nature of the message. Each message type consists of specific parameters (both mandatory and optional).

The message type is found in the mandatory fixed part of the message. The coding of the message type follows the same general rule used in other SS7 protocols. Any message type intended for national usage (internetworking) is to be coded using the upper range of the code. The upper range begins at 1111 1111 and works backward.

The following are the message types currently used in U.S. networks and defined in ANSI T1.113-1992 as well as Bellcore GR-246-CORE, T1.113.3.

Message Types	**H**	**G**	**F**	**E**	**D**	**C**	**B**	**A**
Initial address (IAM)	0	0	0	0	0	0	0	1
*Subsequent address (SAM)	0	0	0	0	0	0	1	0
Information request (INR)	0	0	0	0	0	0	1	1
Information (INF)	0	0	0	0	0	1	0	0
Continuity (COT)	0	0	0	0	0	1	0	1
Address complete (ACM)	0	0	0	0	0	1	1	0
*Connect (CON)	0	0	0	0	0	1	1	1
Forward transfer (FOT)	0	0	0	0	1	0	0	0
Answer (ANM)	0	0	0	0	1	0	0	1

Release (REL)	0	0	0	0	1	1	0	0
Suspend (SUS)	0	0	0	0	1	1	0	1
Resume (RES)	0	0	0	0	1	1	1	0
Release complete (RLC)	0	0	0	1	0	0	0	0
Continuity check request (CCR)	0	0	0	1	0	0	0	1
Reset circuit (RSC)	0	0	0	1	0	0	1	0
Blocking (BLO)	0	0	0	1	0	0	1	1
Unblocking (UBL)	0	0	0	1	0	1	0	0
Blocking acknowledgment (BLA)	0	0	0	1	0	1	0	1
Unblocking acknowledgment (UBA)	0	0	0	1	0	1	1	0
Circuit group reset (GRS)	0	0	0	1	0	1	1	1
Circuit group blocking (CGB)	0	0	0	1	1	0	0	0
Circuit group unblocking (CGU)	0	0	0	1	1	0	0	1
Circuit group blocking acknowledgment (CGBA)	0	0	0	1	1	0	1	0
Circuit group unblocking acknowledgment (CGUA)	0	0	0	1	1	0	1	1
*Call modification request (CMR)	0	0	0	1	1	1	0	0
*Call modification completed (CMC)	0	0	0	1	1	1	0	1
*Call modification reject (CMRJ)	0	0	0	1	1	1	1	0
*Facility request (FAR)	0	0	0	1	1	1	1	1
*Facility accepted (FAA)	0	0	1	0	0	0	0	0
*Facility reject (FRJ)	0	0	1	0	0	0	0	1
Facility deactivated (FAD)	0	0	1	0	0	0	1	0
Facility information (FAI)	0	0	1	0	0	0	1	1
Loopback acknowledgment (LPA)	0	0	1	0	0	1	0	0
CUG selection and validation request (CSVR)	0	0	1	0	0	1	0	1
CUG selection and validation response (CSVS)	0	0	1	0	0	1	1	0
*Delayed release (DRS)	0	0	1	0	0	1	1	1
Pass-along (PAM)	0	0	1	0	1	0	0	0
Circuit group reset acknowledgment (GRA)	0	0	1	0	1	0	0	1
Circuit query (CQM)	0	0	1	0	1	0	1	0
Circuit query response (CQR)	0	0	1	0	1	0	1	1
Call progress (CPG)	0	0	1	0	1	1	0	0
*User-to-user information (USR)	0	0	1	0	1	1	0	1
Unequipped circuit identification code (UCIC)	0	0	1	0	1	1	1	0
Confusion (CFN)	0	0	1	0	1	1	1	1
*Overload (OLM)	0	0	1	1	0	0	0	0
*Charge information (CRG)	0	0	1	1	0	0	0	1
Facility (FAC)	0	0	1	1	0	0	1	1
Circuit reservation acknowledgment (CRA)	1	1	1	0	1	0	0	1
**Circuit reservation (CRM)	1	1	1	0	1	0	1	0
**Circuit validation response (CVR)	1	1	1	0	1	0	1	1
**Circuit validation test (CVT)	1	1	1	0	1	1	0	0
**Exit (EXM)	1	1	1	0	1	1	0	1

The previous message types have been defined in the ANSI and Bellcore standards as well as ITU-TS. Each of these message types has a distinct structure that has parameters. The following are descriptions of each of the message types and the message structure that accompanies each one.

The next section provides the one-octet value, which indicates the message type (in bold), followed by the parameter names and the one-octet values for the parameter names. Each parameter may have several additional bits defining the actual parameter. This section only provides the values for the parameter names.

The parameters shown as optical are supported within each message type, but their usage is dependent on the network and may or may not be used. Many of these parameters may be used in multiple message types, which is why they are discussed in the last section of this chapter.

The section "Message Type Structure" illustrates the message format and provides the basic structure for each message type. The section "ISUP Parameters" will provide the detailed structure for each parameter type along with a description of the parameter. Refer to the section "Broadband Parameters" for detailed information on parameters. All parameters with an asterisk are used in ITU ISUP only.

Message Type Structure

Initial Address Message (IAM)	**0**	**0**	**0**	**0**	**0**	**0**	**0**	**1**
Fixed Mandatory Parameters								
Nature of connection indicators	0	0	0	0	0	1	1	0
Forward call indicators	0	0	0	0	0	1	1	1
Calling party's category	0	0	0	0	1	0	0	1
Variable Mandatory Parameters								
User service information	0	0	0	1	1	1	0	1
Called party number	0	0	0	0	0	1	0	0
Optional Parameters								
Access transport	0	0	0	0	0	0	1	1
Business group	1	1	0	0	0	1	1	0
Call reference	0	0	0	0	0	0	0	1
Calling party number	0	0	0	0	1	0	1	0
Carrier identification	1	1	0	0	0	1	0	1
Carrier selection information	1	1	1	0	1	1	1	0
Charge number	1	1	1	0	1	0	1	1
Circuit assignment map	0	0	1	0	0	1	0	1
Connection request	0	0	0	0	1	1	0	1
*CUG interlock code	0	0	0	1	1	0	1	0
Egress service	1	1	0	0	0	0	1	1
Generic address	1	1	0	0	0	0	0	0
Generic digits	1	1	0	0	0	0	0	1
Generic name	1	1	0	0	0	1	1	1
*Generic notification indicator	0	0	1	0	1	1	0	0
*Generic number	1	1	0	0	0	0	0	0
*Generic reference	0	1	0	0	0	0	1	0
Hop counter	0	0	1	1	1	1	0	1
Information request indicators	0	0	0	0	1	1	1	0
Jurisdiction information	1	1	0	0	0	1	0	0
*Location number	0	0	1	1	1	1	1	1
*MLPP precedence	0	0	1	1	1	0	1	0

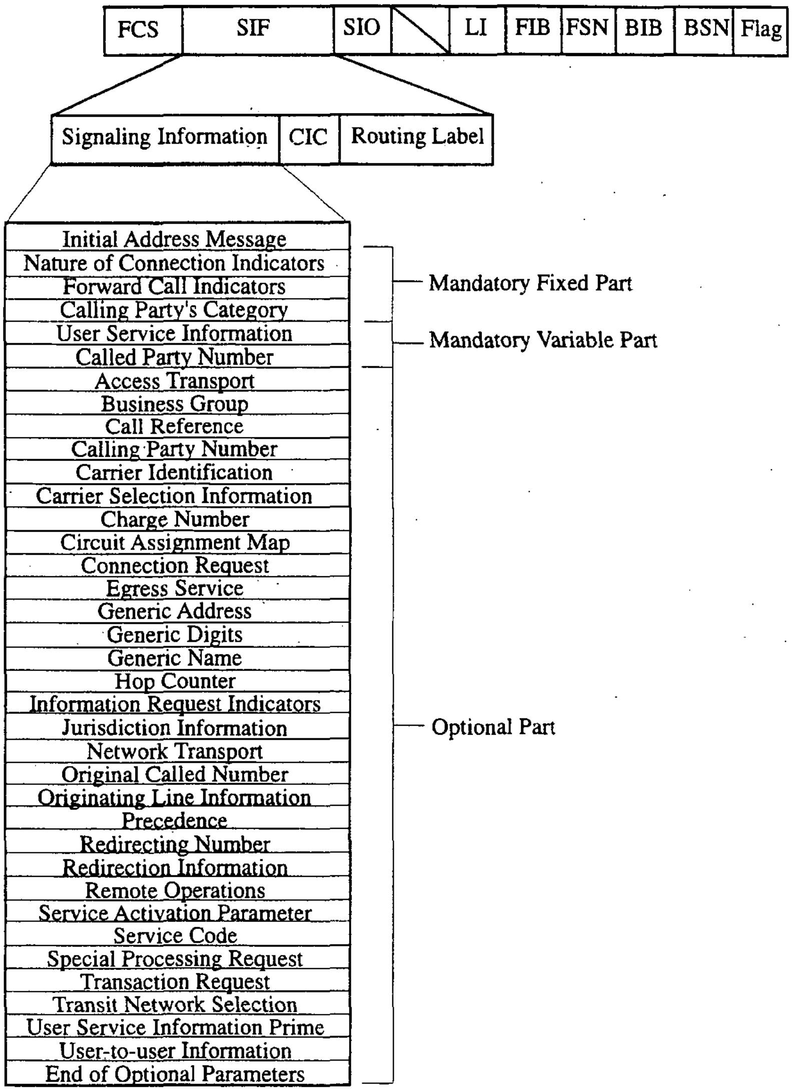


Figure 9.13

*Network-specific facility	0	0	1	0	1	1	1	1
Network transport	1	1	1	0	1	1	1	1
Operator services information	1	1	0	0	0	0	1	0
*Optional forward call indicators	0	0	0	0	1	0	0	0
Original called number	0	0	1	0	1	0	0	0
*Originating ISC point code	0	0	1	0	1	0	1	1
Originating line information	1	1	1	0	1	0	1	0

*Parameter compatibility information	0	0	1	1	1	0	0	1
Precedence	0	0	1	1	1	0	1	0
*Propagation delay counter	0	0	1	1	0	0	0	1
Redirecting number	0	0	0	0	1	0	1	1
Redirection information	0	0	0	1	0	0	1	1
Remote operations	0	0	1	1	0	0	1	0
Service activation parameter	1	1	1	0	0	0	1	0
Service code indicator	1	1	1	0	1	1	0	0
Special processing request	1	1	1	0	1	1	0	1
Transaction request	1	1	1	0	0	0	1	1
Transit network selection	0	0	1	0	0	0	1	1
*Transmission medium requirement	0	0	0	0	0	0	1	0
*Transmission medium requirement prime	0	0	1	1	1	1	1	0
User service information	0	0	0	1	1	1	0	1
User service information prime	0	0	1	1	0	0	0	0
User-to-user indicators	0	0	1	0	1	0	1	0
User-to-user information	0	0	1	0	0	0	0	0

This is the message used to establish a connection on a specified circuit. The IAM provides the circuit information that includes the carrier identification (the long-distance carrier that will be used for this call) and any special requirements to consider in the handling of this call. The IAM is by far the most comprehensive of the ISUP messages and has many parameters. Refer to the section "Broadband Parameters" at end of this chapter for the parameter values and definitions.

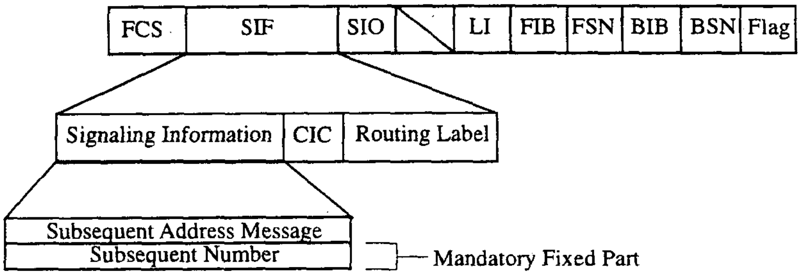


Figure 9.14

Subsequent Address (SAM)	**0**	**0**	**0**	**0**	**0**	**0**	**1**	**0**

No procedure is currently defined for ANSI networks. This message is used in ITU-TS networks only. The structure of this message has been omitted from this section as the scope of this book is ANSI and Bellcore networks. Refer to the ITU specifications for information regarding this and other messages.

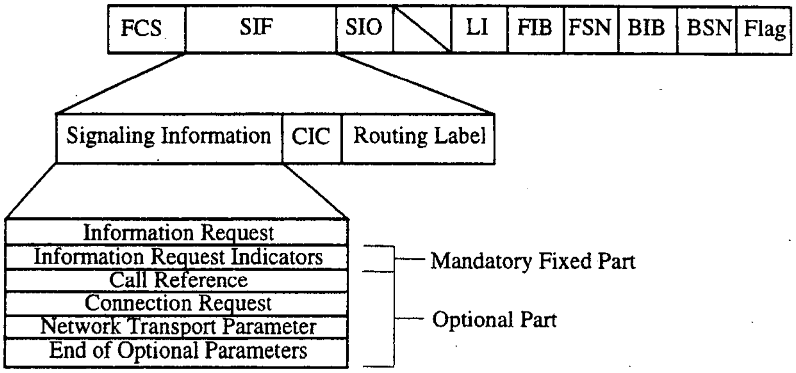


Figure 9.15

Information Request (INR)	**0**	**0**	**0**	**0**	**0**	**0**	**1**	**1**
Fixed Mandatory Parameters								
Information request indicators	0	0	0	0	1	1	1	0
Optional Parameters								
Call reference	0	0	0	0	0	0	0	1
Connection request	0	0	0	0	1	1	0	1
*Network-specific facility	0	0	1	0	1	1	1	1
Network transport parameter	1	1	1	0	1	1	1	1
*Parameter compatibility information	0	0	1	1	1	0	0	1

The *information request* (INR) can be sent by an exchange while a call is in progress to request additional information from another exchange. The additional information is carried in an *information message* (INF) and may provide redirection instructions (forwarding) or other call-handling information.

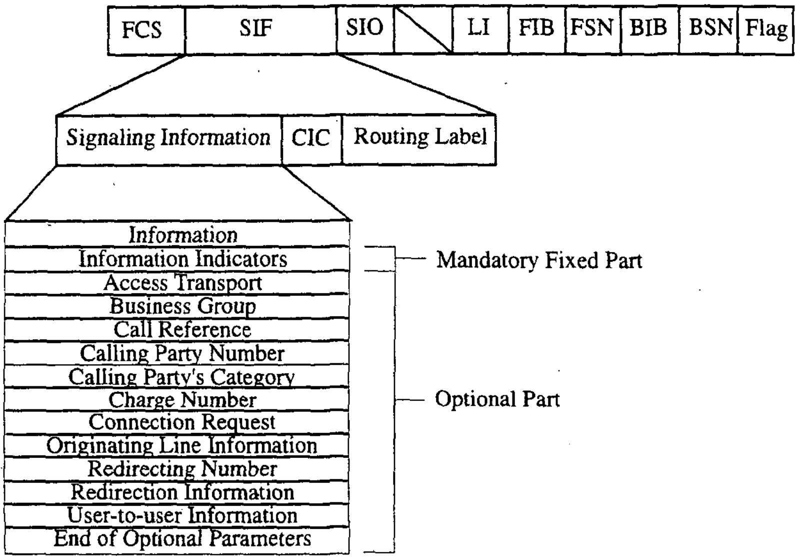


Figure 9.16

Information (INF)	**0**	**0**	**0**	**0**	**0**	**1**	**0**	**0**
Fixed Mandatory Parameters								
Information indicators	0	0	0	0	1	1	1	1
Optional Parameters								
Access transport	0	0	0	0	0	0	1	1
Business group	1	1	0	0	0	1	1	0
Call reference	0	0	0	0	0	0	0	1
Calling party number	0	0	0	0	1	0	1	0
Calling party's category	0	0	0	0	1	0	0	1
Charge number	1	1	1	0	1	0	1	1
Connection request	0	0	0	0	1	1	0	1
Originating line information	1	1	1	0	1	0	1	0
Redirection information	0	0	0	1	0	0	1	1
User-to-user information	0	0	1	0	0	0	0	0

The INF is used to pass additional information about a call upon request from the distant exchange. The information is requested from an exchange using the INR, and the reply is carried in the INF message. The type of information is usually call-handling information, such as the number to which to forward a call or a billing number.

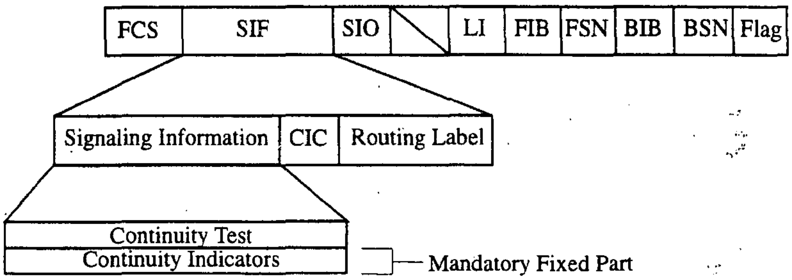


Figure 9.17

Continuity (COT)	**0**	**0**	**0**	**0**	**0**	**1**	**0**	**1**
Fixed Mandatory Parameters								
Continuity indicators	0	0	0	1	0	0	0	0

The COT message is used to indicate the success of a continuity test (or failure). The continuity test is performed on the voice circuit depending on criteria set by the network operator at the time of deployment. The COT is used to indicate the status of the preceding circuit and the circuit selected in the forward direction to the next exchange.

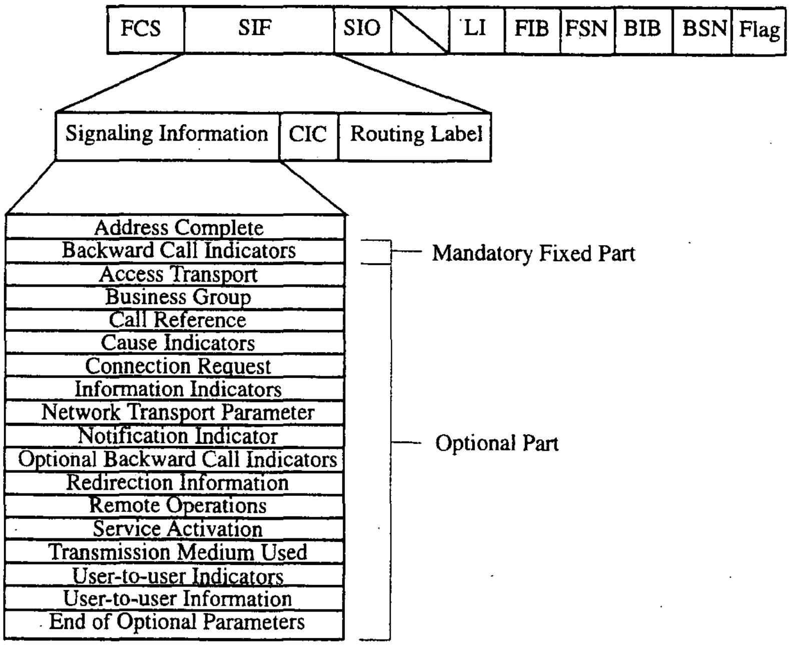


Figure 9.18

Address Complete (ACM)	**0**	**0**	**0**	**0**	**0**	**1**	**1**	**0**
Mandatory Fixed Parameters								
Backward call indicators	0	0	0	1	0	0	0	1
Optional Parameters								
*Access delivery information	0	0	1	0	1	1	1	0
Access transport	0	0	0	0	0	0	1	1
Business group	1	1	0	0	0	1	1	0
*Call diversion information	0	0	1	1	0	1	1	0
Call reference	0	0	0	0	0	0	0	1
Cause indicators	0	0	0	1	0	0	1	0
Connection request	0	0	0	0	1	1	0	1
*Echo control information	0	0	1	1	0	1	1	1
*Generic notification indicators	0	0	1	0	1	1	0	0
Information indicators	0	0	0	0	1	1	1	1
*Network-specific facility	0	0	1	0	1	1	1	1
Network transport parameter	1	1	1	0	1	1	1	1
Notification indicator	1	1	1	0	0	0	0	1
Optional backward call indicators	0	0	1	0	1	0	0	1
*Parameter compatibility information	0	0	1	1	1	0	0	1
Redirection information	0	0	0	1	0	0	1	1
*Redirection number	0	0	0	0	1	1	0	0
*Redirection number restriction	0	1	0	0	0	0	0	0
Remote operations	0	0	1	1	0	0	1	0
Service activation	0	0	1	1	0	0	1	1
Transmission medium used	0	0	1	1	0	1	0	1
User-to-user indicators	0	0	1	0	1	0	1	0
User-to-user information	0	0	1	0	0	0	0	0

The ACM is sent by a distant exchange upon receipt of all address signals (IAM and any subsequent information sent) needed to establish a connection on a circuit between the two exchanges. The ACM indicates that the call is being processed, and the distant exchange checks the availability of the called party. This could mean that the called party's telephone is being signaled (ringing occurs if analog or an alerting message is returned if ISDN). In some networks, cut-through on the voice circuit can take place after receipt of the ACM.

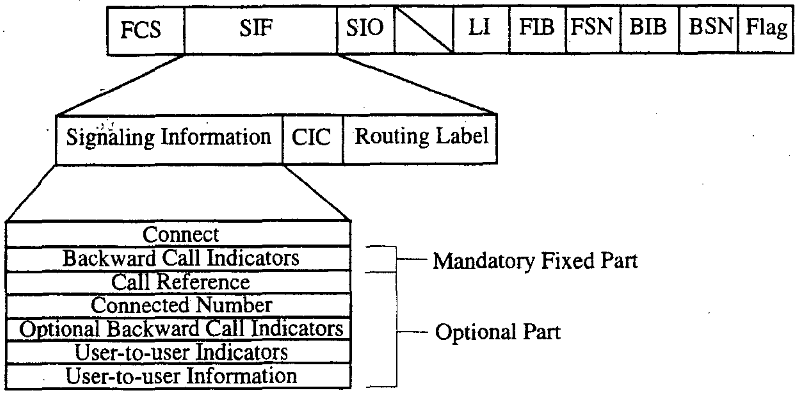


Figure 9.19

Connect (CON)	**0**	**0**	**0**	**0**	**0**	**1**	**1**	**1**

The CON is defined for use in international networks but not ANSI networks.

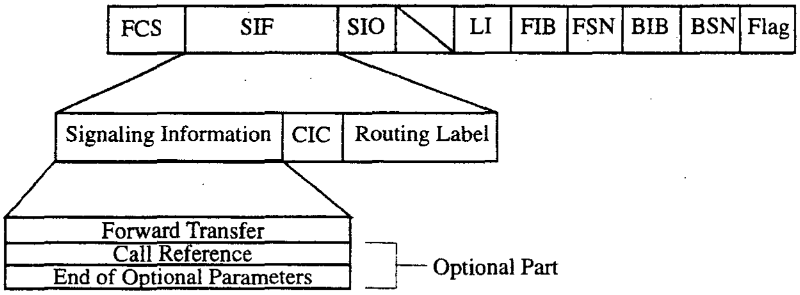


Figure 9.20

Forward Transfer (FOT)	**0**	**0**	**0**	**0**	**1**	**0**	**0**	**0**
Optional Parameters								
Call reference	0	0	0	0	0	0	0	1

The *forward transfer* (FOT) is used in conjunction with operator services. In exchanges where telephone calls are set up automatically (which is the case in all of North America today), an operator is only needed in certain circumstances. This message is sent in the forward direction to bring an operator into the circuit when operator assistance is required to complete the call. When the call has been completed, the operator can be recalled to terminate the call or initiate another call for the same calling party.

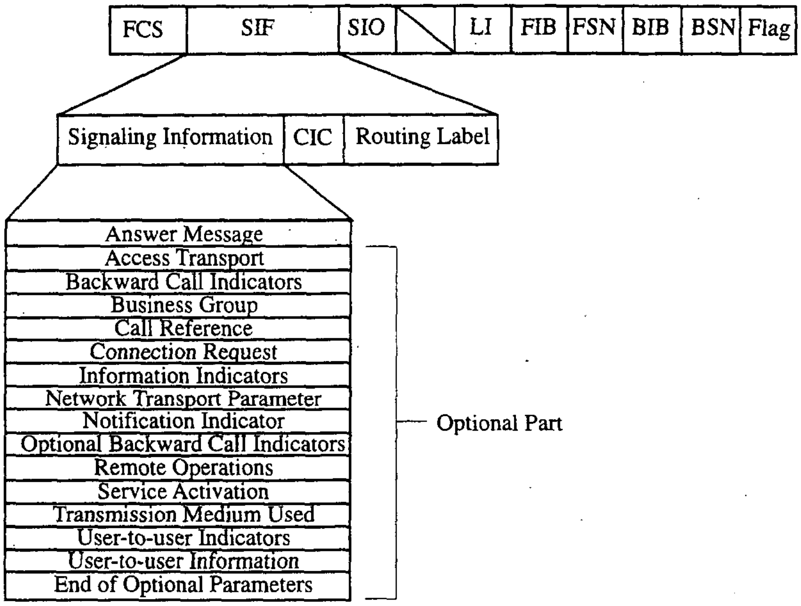


Figure 9.21

Answer (ANM)	**0**	**0**	**0**	**0**	**1**	**0**	**0**	**1**
Optional Parameters								
Access transport	0	0	0	0	0	0	1	1
Backward call indicators	0	0	0	1	0	0	0	1
Business group	1	1	0	0	0	1	1	0
Call reference	0	0	0	0	0	0	0	1
Connection request	0	0	0	0	1	1	0	1
Information indicators	0	0	0	0	1	1	1	1
Network transport parameter	1	1	1	0	1	1	1	1
Optional backward call indicators	0	0	1	0	1	0	0	1
Remote operations	0	0	1	1	0	0	1	0
Service activation	0	0	1	1	0	0	1	1
Transmission medium used	0	0	1	1	0	1	0	1
User-to-user indicators	0	0	1	0	1	0	1	0
User-to-user information	0	0	1	0	0	0	0	0

The ANM is sent in the backward direction to indicate that the called party has answered the call. The use of this parameter is really twofold. In semiautomatic networks, this parameter is used for call supervision. In automatic networks, the ANM is used to begin metering the call for billing purposes. Metering domestic and international calls can be activated using this parameter.

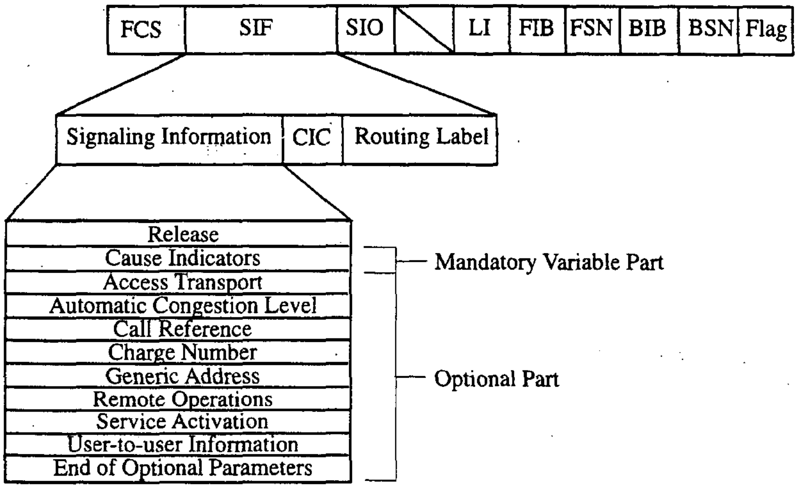


Figure 9.22

Release (REL)	**0**	**0**	**0**	**0**	**1**	**1**	**0**	**0**
Mandatory Variable Parameters								
Cause indicators	0	0	0	1	0	0	1	0
Optional Parameters								
Access transport	0	0	0	0	0	0	1	1
Automatic congestion level	0	0	1	0	0	1	1	1
Call reference	0	0	0	0	0	0	0	1
Charge number	1	1	1	0	1	0	1	1
Generic address	1	1	0	0	0	0	0	0
Service activation	0	0	1	1	0	0	1	1
User-to-user information	0	0	1	0	0	0	0	0

The REL is sent in either direction indicating that one of the parties (called or calling) has gone on-hook and the call is being terminated. The REL does not return the circuit back to its idle state, however. An RLC must be received before the circuit is returned to idle.

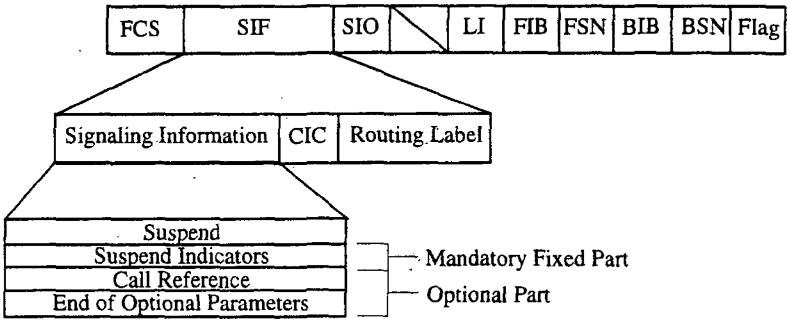


Figure 9.23

Suspend (SUS)	**0**	**0**	**0**	**0**	**1**	**1**	**0**	**1**
Mandatory Fixed Parameters								
Suspend/resume indicators	0	0	1	0	0	0	0	1
Optional Parameters								
Call reference	0	0	0	0	0	0	0	1

This message is used when a non-ISDN party is returned to an on-hook state. When an ISDN party is returned on-hook, only the REL is used, but with non-ISDN, the SUS is sent first and is followed by the REL and RLC. For complete call setup and teardown information, review the previous sections in this chapter on call setup and teardown.

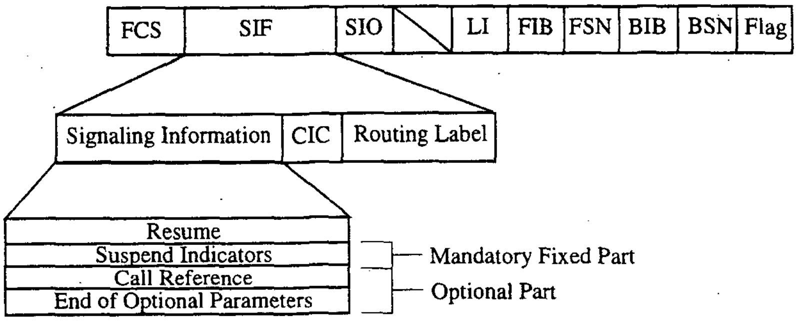


Figure 9.24

Resume (RES)	**0**	**0**	**0**	**0**	**1**	**1**	**1**	**0**
Mandatory Fixed Parameters								
Suspend/resume indicators	0	0	1	0	0	0	1	0
Optional Parameters								
Call reference	0	0	0	0	0	0	0	1

The RES is used in two circumstances. In a network where interworking is used, RES indicates that the interworking node has reanswered. In a network with non-ISDN circuits, the RES indicates that a non-ISDN called party went on-hook, but then went back off-hook again within a certain time (quickly) and the call connection should remain established. Had the called party stayed on-hook past the specified time (network dependent), the SUS message would have been sent in the backward direction to begin releasing the circuit.

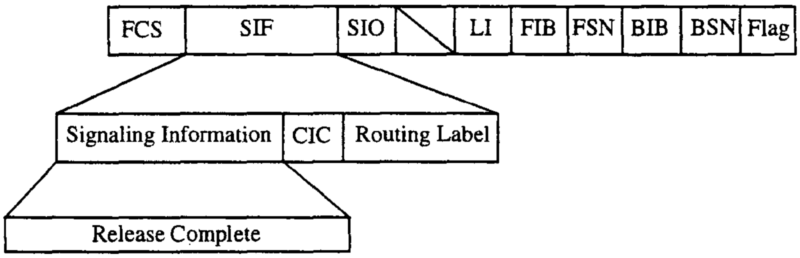


Figure 9.25

Release Complete (RLC) 0 0 0 1 0 0 0 0

No parameters are given in the RLC, only the message type field. The RLC is used to indicate receipt of an REL message and serves as an acknowledgment of the release. Once the RLC has been received, the indicated circuit can be released and returned to its idle state. The CIC is sent with this message, but is not an integral part of the message itself. The CIC is presented just after the routing label.

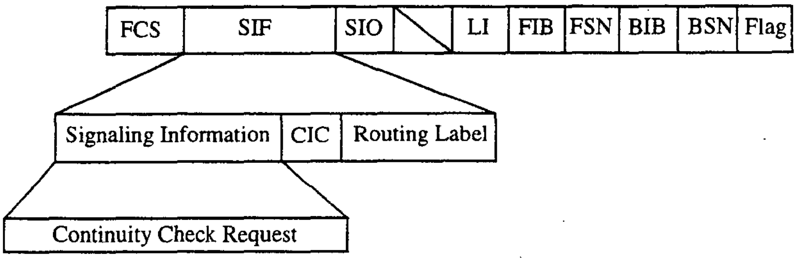


Figure 9.26

Continuity Check Request (CCR) 0 0 0 1 0 0 0 1

No parameters are given in the CCR. The CCR is used to request the continuity check equipment that will be attached to the circuit indicated in the CIC field of the message. The equipment is attached to the voice circuit for loopback testing. Once loopback has been detected, the status of "successful" is sent through the SS7 network using the IAM or the COT, depending on the network and the circumstances.

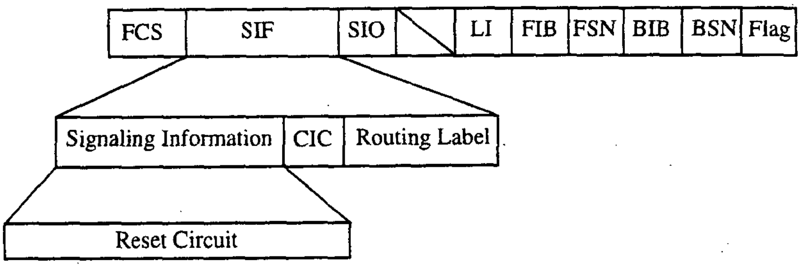


Figure 9.27

Reset Circuit (RSC) **0 0 0 1 0 0 1 0**

No parameters are given in the *reset circuit message* (RSC). The purpose of this message is to enable an exchange to reset a circuit to the state that exchange thinks the circuit should be in. This occurs when a memory error occurs at an exchange, and it no longer knows the state of the circuit in question. To restart from scratch, the RSC is sent. Any calls in progress or blocked conditions are released and the circuit is returned to an idle state after an alignment procedure (not to be confused with the alignment procedure used on SS7 links).

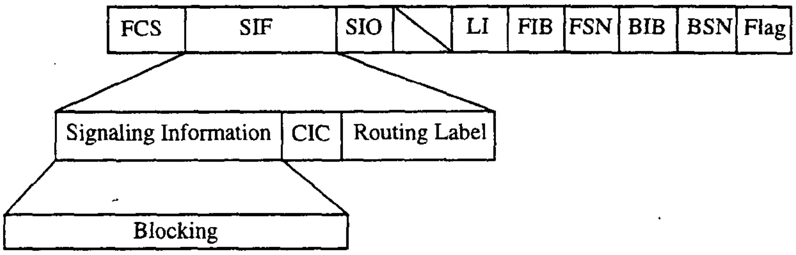


Figure 9.28

Blocking (BLO) **0 0 0 1 0 0 1 1**

No parameters are given in the *blocking message* (BLO). This message enables one exchange to block a voice circuit at a remote exchange, preventing voice calls from being reserved on the voice circuit from the remote end. The circuit is identified in the CIC field.

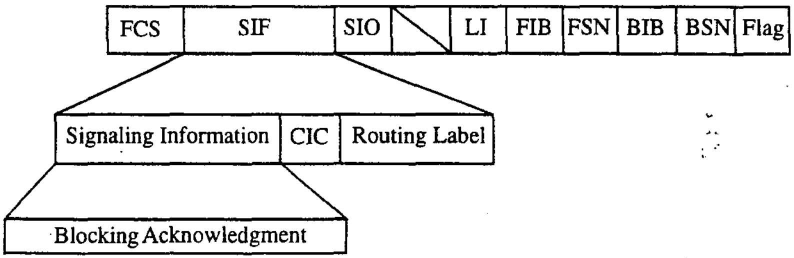


Figure 9.29

Blocking Acknowledgment (BLA) **0 0 0 1 0 1 0 1**

No parameters are given in the *blocking acknowledgment message* (BLA). This message acknowledges receipt of the BLO and indicates that the circuit has been blocked. The voice circuit is identified in the CIC field.

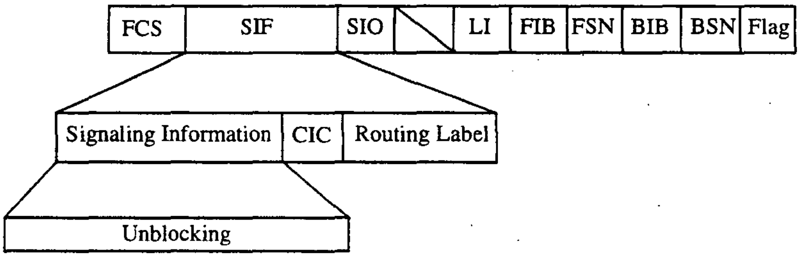


Figure 9.30

Unblocking (UBL) **0 0 0 1 0 1 0 0**

No parameters are given in the *unblocking message* (UBL). This message is sent by an exchange to remove a blocking condition at a remote exchange. The circuit being unblocked is indicated in the CIC field.

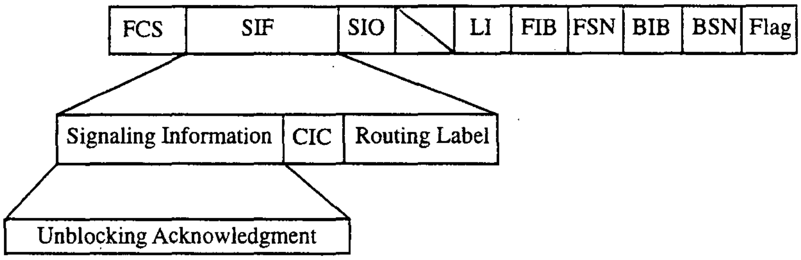


Figure 9.31

Unblocking Acknowledgment (UBA) 0 0 0 1 0 1 1 0

No parameters are given in the *unblocking acknowledgment message* (UBA). This message is sent to acknowledge receipt of the UBL. The acknowledgment also indicates that the circuit has been unblocked. The circuit is identified in the CIC field.

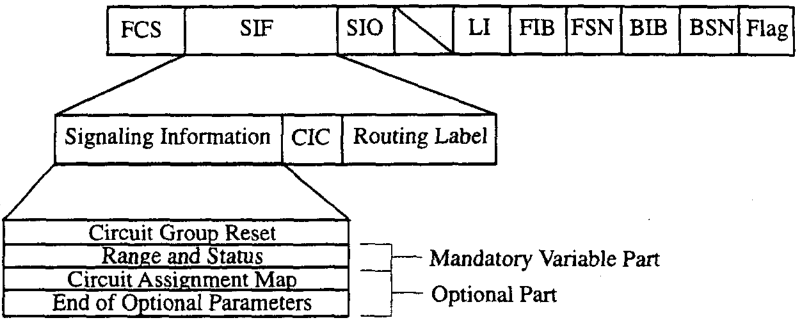


Figure 9.32

Circuit Group Reset (GRS)	0	0	0	1	0	1	1	1
Mandatory Variable Parameters								
Range and status	0	0	0	1	0	1	1	0
Optional Parameters								
Circuit assignment map	0	0	1	0	0	1	0	1

This message is used to reset a group of voice circuits when the exchange no longer knows the status of the voice circuits. This could be the result of memory malfunction or some other error that caused it to lose track of the circuits' status. The range parameter is used to identify the range of voice circuits to be reset. Any calls in progress or blocked conditions will be canceled and the voice circuits that were indicated are released. However, they must go through diagnostics and alignment procedures (voice alignment, not SS7 alignment) before becoming available for calls again.

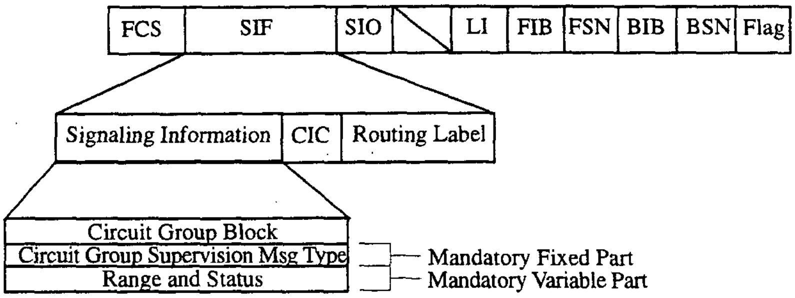


Figure 9.33

Circuit Group Blocking (CGB)	**0**	**0**	**0**	**1**	**1**	**0**	**0**	**0**
Fixed Mandatory Parameters								
Circuit group supervision message type indicator	0	0	0	1	0	1	0	1
Mandatory Variable Parameters								
Range and status	0	0	0	1	0	1	1	0

This message is sent by maintenance personnel from an operations terminal to block voice circuits from being used for voice calls during maintenance routines. The voice circuits are manually busied until maintenance procedures are completed, at which point they must be unblocked manually. The circuit group supervision message type indicator parameter indicates what type of blocking to invoke, and the range and status parameter indicates what range of circuits will block the status (blocked or unblocked) of the specified circuits.

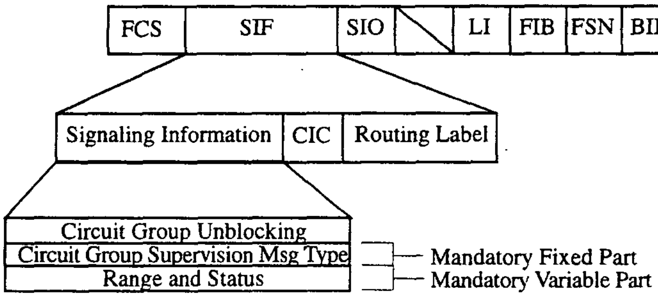


Figure 9.34

Circuit Group Unblocking (CGU)	**0**	**0**	**0**	**1**	**1**	**0**	**0**	**1**
Fixed Mandatory Parameters								
Circuit group supervision message type indicator	0	0	0	1	0	1	0	1
Mandatory Variable Parameters								
Range and status	0	0	0	1	0	1	1	0

This message is sent by maintenance personnel from an operations terminal to unblock voice circuits that were previously blocked for maintenance purposes. The circuit group supervision message type indicator parameter indicates what type of unblocking to invoke, and the range and status parameter shows the range of circuits that were blocked and their present status (blocked).

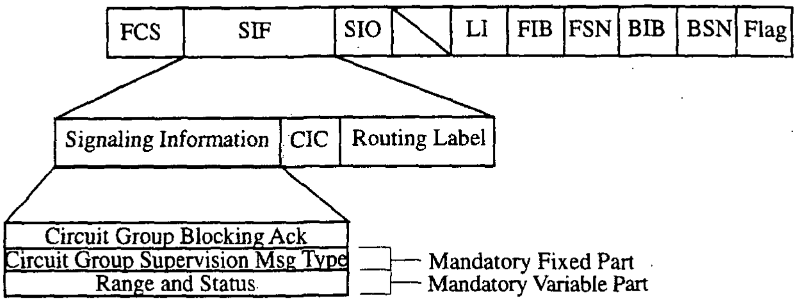


Figure 9.35

Circuit Group Unblocking Acknowledgement (CGBA)	**0**	**0**	**0**	**1**	**1**	**0**	**1**	**0**
Fixed Mandatory Parameters								
Circuit group supervision message type indicator	0	0	0	1	0	1	0	1
Mandatory Variable Parameters								
Range and status	0	0	0	1	0	1	1	0

This message is used to acknowledge receipt of a CGB message and indicates that the circuits have been blocked. The supervision message type indicator shows the type of blocking invoked and the range and status parameter shows the range of circuits that were blocked and their present status (blocked).

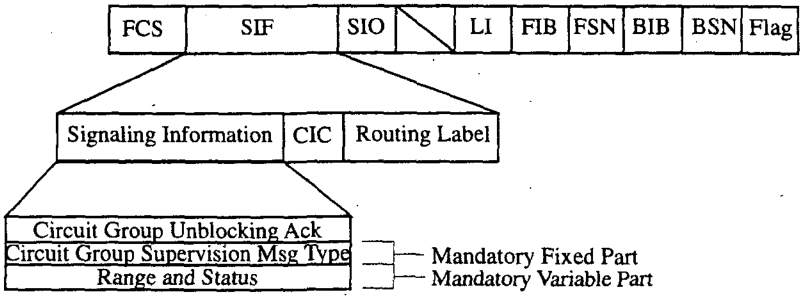


Figure 9.36

Circuit Group Unblocking Acknowledgement (CGUA)	**0**	**0**	**0**	**1**	**1**	**0**	**1**	**1**
Fixed Mandatory Parameters								
Circuit group supervision message type indicator	0	0	0	1	0	1	0	1
Mandatory Variable Parameters								
Range and status	0	0	0	1	0	1	1	0

This message is used to acknowledge receipt of a CGU message and indicates that the circuits have been unblocked. The supervision message type parameter indicates the type of unblocking used, and the range and status parameter indicates the range of circuits that have been unblocked and the status of those circuits (unblocked).

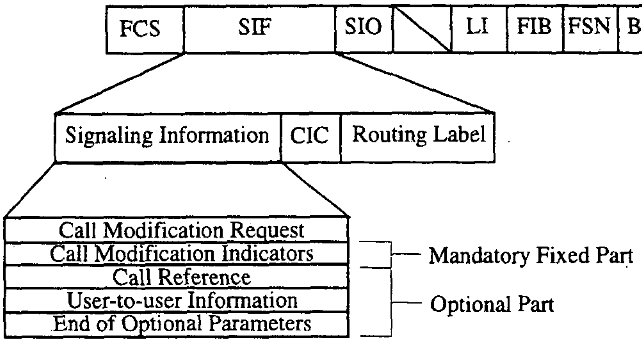


Figure 9.37

Call Modification Request (CMR) 0 0 0 1 1 1 0 0

There are currently no procedures written for this in ANSI networks. This message is only found in ITU-TS networks.

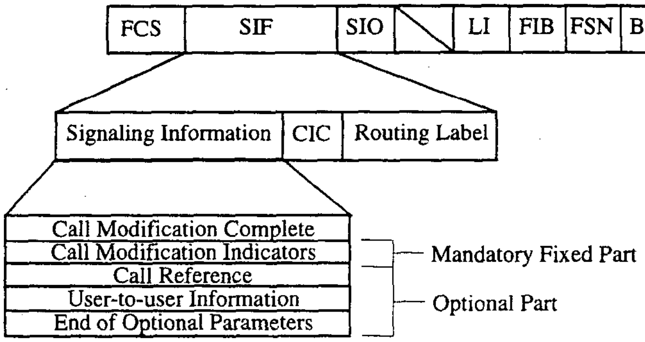


Figure 9.38

Call Modification Completed (CMC) 0 0 0 1 1 1 0 1

There are currently no procedures written for this in ANSI networks. This message is only found in ITU-TS networks.

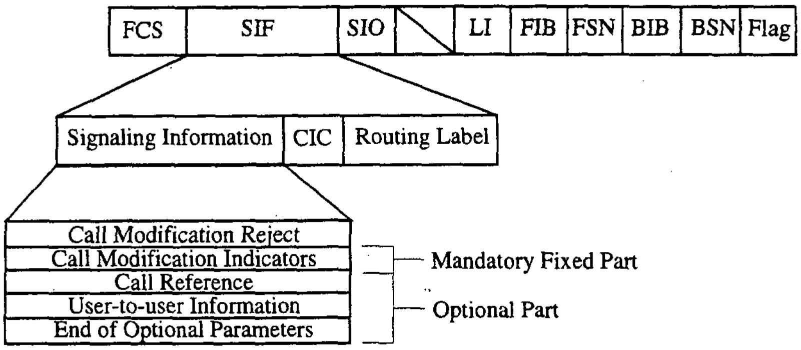


Figure 9.39

Call Modification Reject (CMRJ) 0 0 0 1 1 1 1 0

There are currently no procedures written for this in ANSI networks. This message is only found in ITU-TS networks.

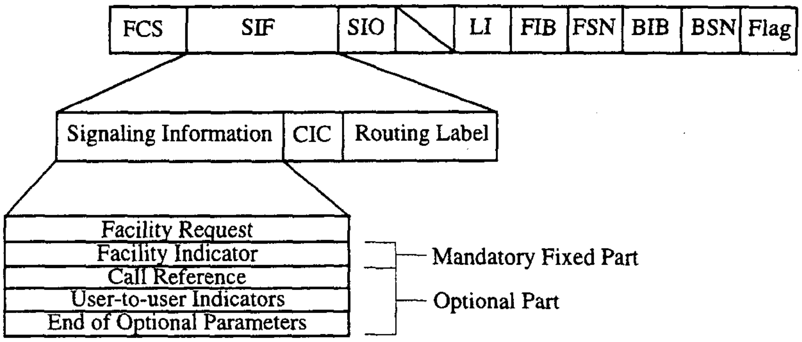


Figure 9.40

Facility Request (FAR) 0 0 0 1 1 1 1 1

There are currently no procedures written for this in ANSI networks. This message is only found in ITU-TS networks.

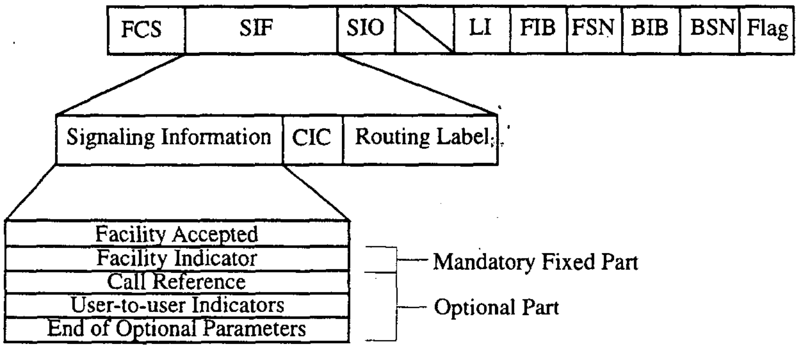


Figure 9.41

Facility Accepted (FAA) 0 0 1 0 0 0 0 0

There are currently no procedures written for this in ANSI networks. This message is only found in ITU-TS networks.

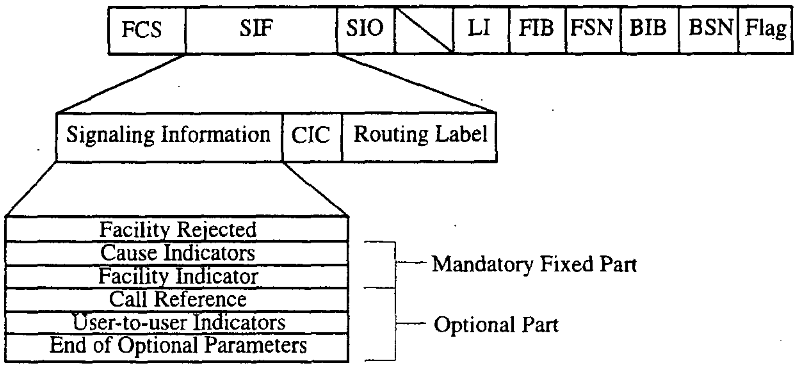


Figure 9.42

Facility Reject (FRJ) 0 0 1 0 0 0 0 1

There are currently no procedures written for this in ANSI networks. This message is only found in ITU-TS networks.

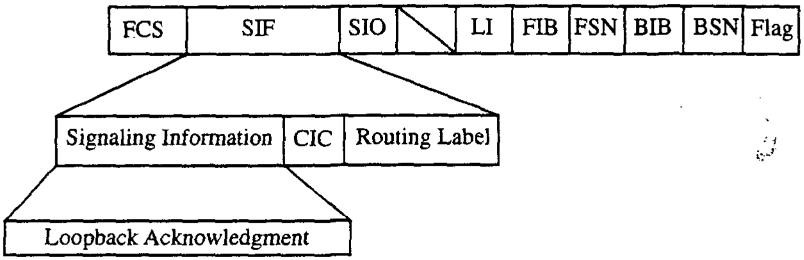


Figure 9.43

Loopback Acknowledgment (LPA) 0 0 1 0 0 1 0 0

No parameters are given in this message. This message is used to indicate that loopback equipment has been connected in response to a CCR and that loopback testing is being performed. The voice circuit identification is provided in the CIC field.

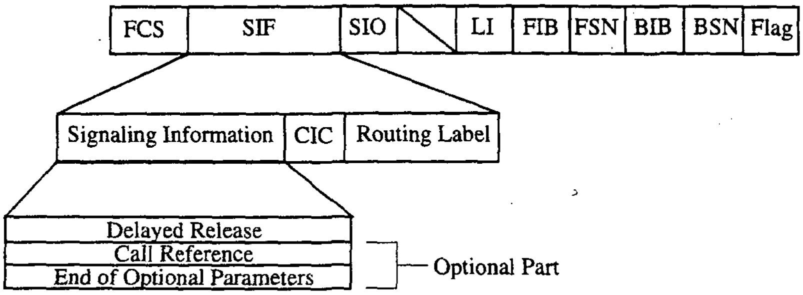


Figure 9.44

Delayed Release (DRS) 0 0 1 0 0 1 1 1

There are currently no procedures written for this in ANSI networks. This message is only found in ITU-TS networks.

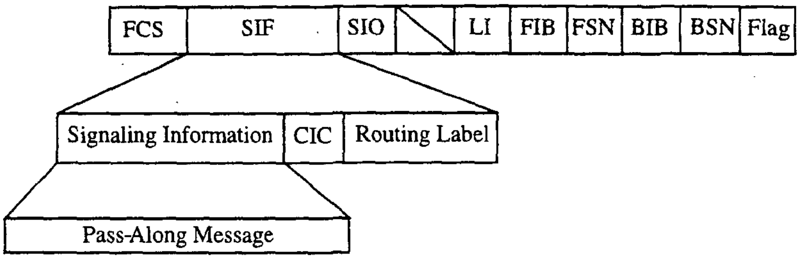


Figure 9.45

Pass-Along (PAM)	**0**	**0**	**1**	**0**	**1**	**0**	**0**	**0**

There are no specific parameters associated with this command; however, when the *pass-along message* (PAM) type is given, another message type is normally contained within (as if parameters). This enables a message to be routed to the exchange associated with the specified voice circuit connection so that information may be passed along using the same path as that used for the call setup messages.

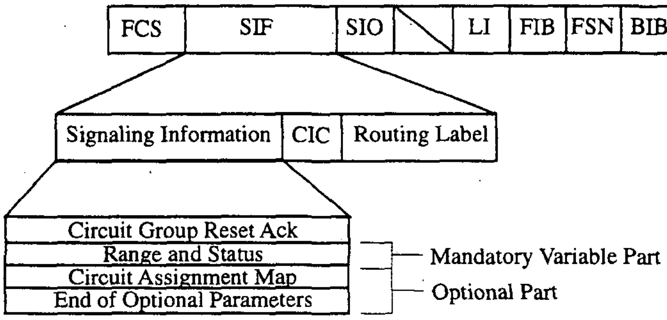


Figure 9.46

Circuit Group Reset Acknowledgment (GRA)	**0**	**0**	**1**	**0**	**1**	**0**	**0**	**1**
Mandatory Variable Parameters								
Range and status	0	0	0	1	0	1	1	0
Optional Parameters								
Circuit assignment map	0	0	1	0	0	1	0	1

This message is used to indicate receipt of a GRS message. This message also indicates that the reset has been performed on the circuits identified in the range parameter. The status parameter indicates the current status of those circuits.

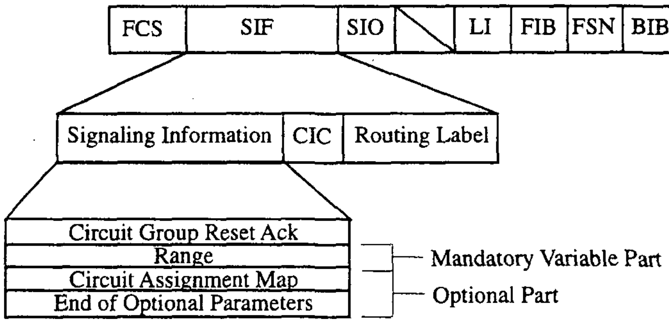


Figure 9.47

Circuit Query (CQM)	**0**	**0**	**1**	**0**	**1**	**0**	**1**	**0**
Mandatory Variable Parameters								
Range and status	0	0	0	1	0	1	1	0
Optional Parameters								
Circuit assignment map	0	0	1	0	0	1	0	1

This message is sent to a distant exchange to learn the status of a range of voice circuits (blocked or unblocked). The range of voice circuits is specified in the range parameter, which normally also has a status subfield. However, the status information is not returned with this message; therefore, the status field is not used (set to zeros). A *circuit query response message* (CQR) is used to inform the querying exchange of the status information.

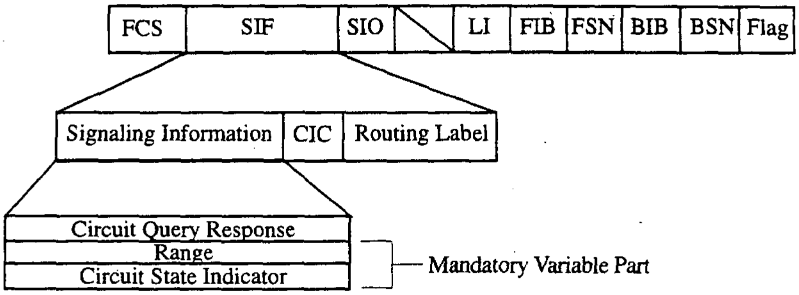


Figure 9.48

Circuit Query Response (CQR)	**0**	**0**	**1**	**0**	**1**	**0**	**1**	**1**
Mandatory Variable Parameters								
Range and status	0	0	0	1	0	1	1	0
Circuit state indicator	0	0	1	0	0	1	1	0

The CQR is sent in response to a *circuit query message* (CQM) and provides the status of the specified voice circuits. The range of voice circuits is specified in the range parameter, and the status of those circuits is provided in the circuit state indicator. The status subfield of the range parameter is not used in this message (set to zeros).

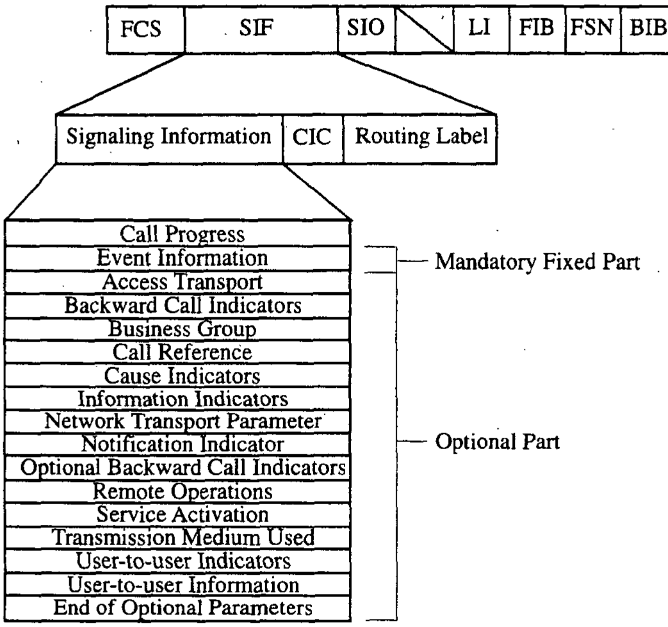


Figure 9.49

Call Progress (CPG)	**0**	**0**	**1**	**0**	**1**	**1**	**0**	**0**
Mandatory Fixed Parameters								
Event information	0	0	1	0	0	1	0	0
Optional Parameters								
Access transport	0	0	0	0	0	0	1	1
Backward call indicators	0	0	0	1	0	0	0	1
Business group	1	1	0	0	0	1	1	0
Call reference	0	0	0	0	0	0	0	1
Cause indicators	0	0	0	1	0	0	1	0
Information indicators	0	0	0	0	1	1	1	1
Network transport parameter	1	1	1	0	1	1	1	1
Notification indicator	1	1	1	0	0	0	0	1
Optional backward call indicators	0	0	1	0	1	0	0	1
Redirecting number	0	0	0	0	1	0	1	1
Remote operations	0	0	1	1	0	0	1	0
Service activation	0	0	1	1	0	0	1	1
Transmission medium used	0	0	1	1	0	1	0	1
User-to-user indicators	0	0	1	0	1	0	1	0
User-to-user information	0	0	1	0	0	0	0	0

The call progress message is used to notify a distant exchange that some event has occurred during the progress of a call. The event is not a catastrophic or error-related event, but a call-related event. The event information parameter indicates what type of event occurred (an alerting message was received, the call was forwarded because of a busy signal, and so on), whereas the optional parameters provide additional support information required depending on the event.

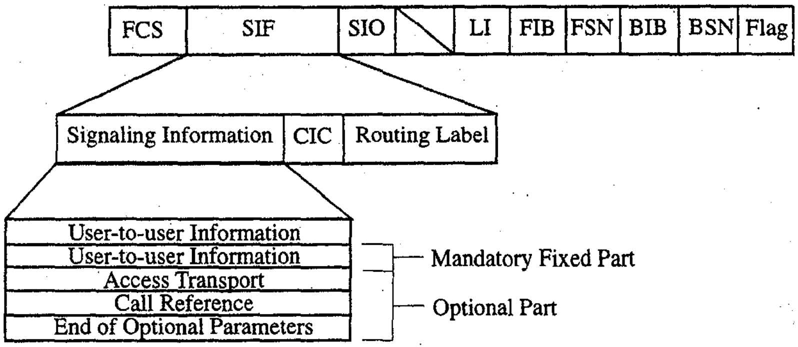


Figure 9.50

User-to-User Information (USR) **0 0 1 0 1 1 0 1**

This message does not have any definition in ANSI networks and is used only in international ITU-TS networks.

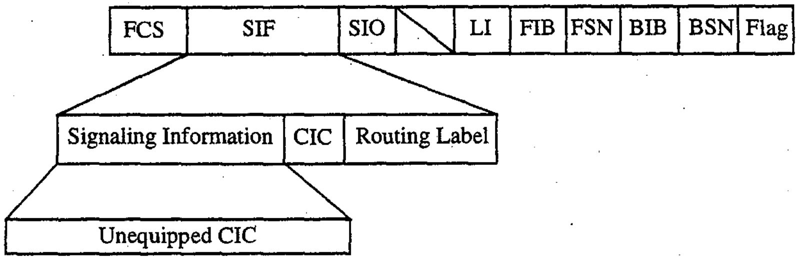


Figure 9.51

Unequipped Circuit Identification Code (UCIC) **0 0 1 0 1 1 1 0**

There are no parameters in this message. This message is used to notify a distant exchange that is the originator of an ISUP IAM that the CIC it has requested to be connected is not equipped. Upon receipt of this message, the exchange that originated the IAM must select a different CIC while marking the first CIC as unavailable.

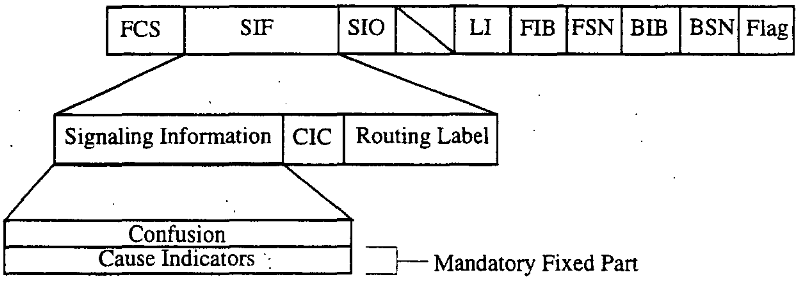


Figure 9.52

Confusion (CFN)	**0**	**0**	**1**	**0**	**1**	**1**	**1**	**1**
Mandatory Variable Parameters								
Cause indicators	0	0	0	1	0	0	1	0

The *confusion message* (CFN) indicates that the exchange has received a message it does not recognize, and it does not know how to handle the message. The CFN is sent to the originator of the ISUP message. This only applies to ISUP messages and does not apply to TCAP, SCCP, or any other protocol message other than ISUP. The cause indicators parameter indicates where the CFN originated as well as why the message is being sent.

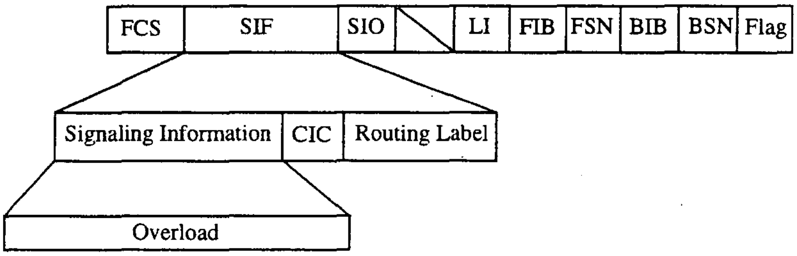


Figure 9.53

Overload (OLM)	**0**	**0**	**1**	**1**	**0**	**0**	**0**	**0**

This message is not supported in ANSI networks and is used only in international ITU-TS networks.

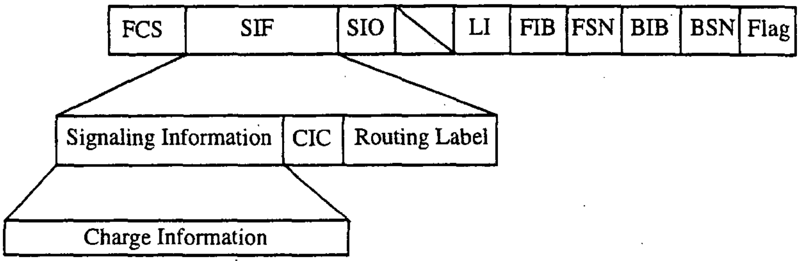


Figure 9.54

Charge Information (CRG)	**0**	**0**	**1**	**1**	**0**	**0**	**0**	**1**

This message is not supported in ANSI networks and is used only in international ITU-TS networks.

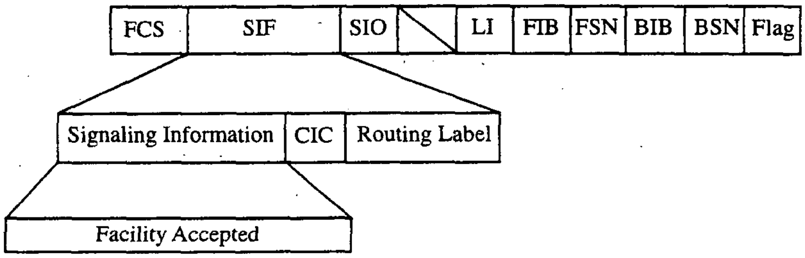


Figure 9.55

Facility (FAC)	**0**	**0**	**1**	**1**	**0**	**0**	**1**	**1**
Optional Parameters								
Remote operations	0	0	1	1	0	0	1	0
Service activation	0	0	1	1	0	0	0	0

This message may be sent by either the local or distant exchange to request an action at that exchange. The same message may also be used as an acknowledgment that the action was performed successfully. The service activation parameter indicates the type of service that is being requested (or has been invoked in the case of an acknowledgment). Call waiting is defined in the Bellcore standard, but all other codes are considered as network specific and are undefined in the standards.

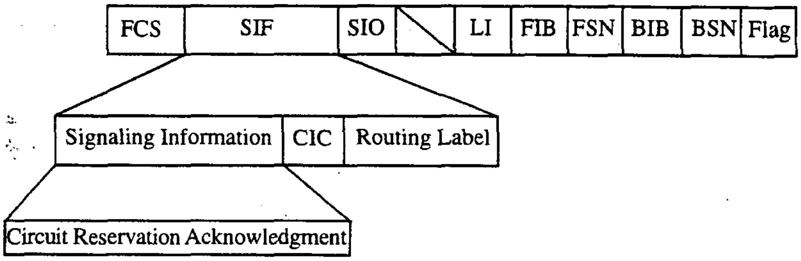


Figure 9.56

Circuit Reservation Acknowledgment (CRA)	**1**	**1**	**1**	**0**	**1**	**0**	**0**	**1**

There are no parameters in this message. This is sent to an exchange after receipt of a CRM as an acknowledgment that the circuit has been reserved for a call. This message only applies when the circuit reservation procedure is incorporated.

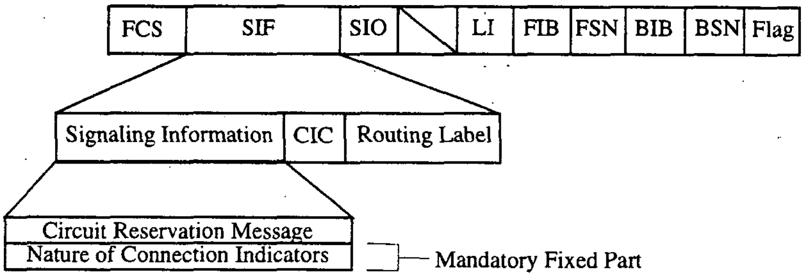


Figure 9.57

Circuit Reservation (CRM)	**1**	**1**	**1**	**0**	**1**	**0**	**1**	**0**
Mandatory Fixed Parameters								
Nature of connection indicators	0	0	0	0	0	1	1	0

This is used only when interworking with a non-SS7 network, such as an analog network using MF signaling. This is typically the case in rural areas here in the United States, although rural areas are quickly being converted to SS7 networks. Where SS7 is not available, the exchanges rely on conventional signaling methods (such as MF), which require special handling by the SS7 network. CRM enables the voice circuit to be reserved for a call. See the section "Interworking with Non-SS7 Networks" earlier in this chapter to understand how this works. This procedure is only used in ANSI networks.

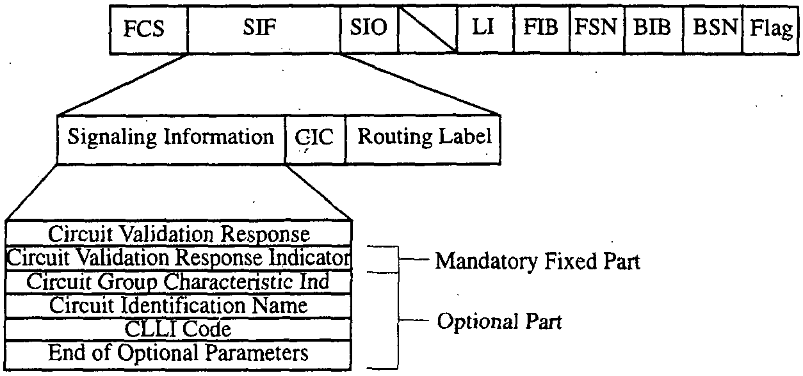

FCS
SIF
SIO
LI
FIB
FSN
BIB
BSN
Flag
Signaling Information
CIC
Routing Label
Circuit Validation Response
Circuit Validation Response Indicator
Circuit Group Characteristic Ind
Circuit Identification Name
CLLI Code
End of Optional Parameters
Mandatory Fixed Part
Optional Part

Figure 9.58

Circuit Validation Response (CVR)	**1**	**1**	**1**	**0**	**1**	**0**	**1**	**1**
Mandatory Fixed Parameters								
Circuit validation response indicator	1	1	1	0	0	1	1	0
Circuit group characteristic indicators	1	1	1	0	0	1	0	1
Optional Parameters								
Circuit identification name (sending end)	1	1	1	0	1	0	0	0
Common Language Location Identification (CLLI) Code (sending end)	1	1	1	0	1	0	0	1

The CVR message is sent in response to a CVM. The CVR message provides the results of the circuit validation test. The CVM provides a way for maintenance personnel to check the translations at far-end exchanges and verify that when a new translation is entered a CIC can be obtained by the exchange when establishing a new call and that the physical port associated with that CIC can be seized. The response message indicates whether this test passed or failed. This procedure is only used in ANSI networks.

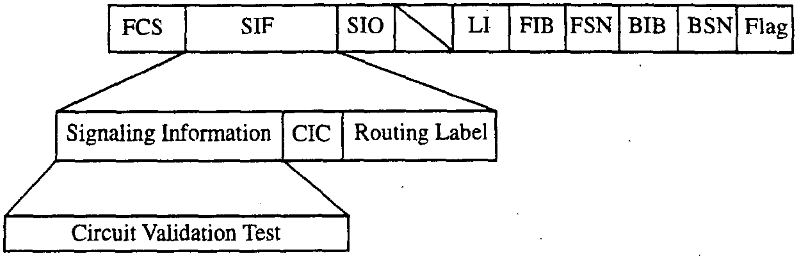


Figure 9.59

Circuit Validation Test (CVT)	1	1	1	0	1	1	0	0

There are no parameters for this message. This message is used to initiate a translations test at a distant exchange. The CVM provides the results of the circuit validation test. The primary purpose of this test is to verify that new translations were entered properly and that a physical port can be connected between two exchanges.

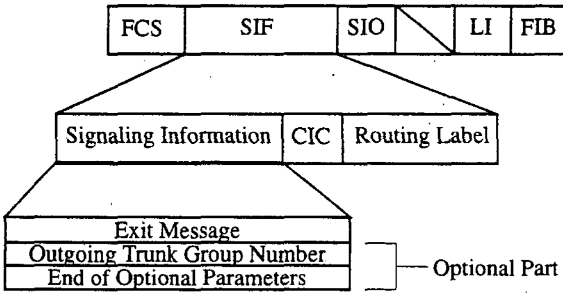


Figure 9.60

Exit (EXM)	1	1	1	0	1	1	0	1
Optional Parameters								
Outgoing trunk group number	1	1	1	0	0	1	1	1

The EXM is only used when interworking with another network. When an IAM is sent to a gateway to establish a connection in another network, the EXM is sent in the backward direction to indicate that the IAM has passed the gateway and is being forwarded to the other network.

ISUP Parameters

The previous section defined all the message types and their structures. Each of the parameters possible with any one message type was outlined and the parameter name value was provided. However, parameters contain additional information besides the parameter name.

Because parameters can be used in multiple message types, it is easier to list them all here in alphabetical order. In this section, all parameters for both ANSI and ITU-TS are listed, along with illustrations showing the message structure of each parameter. To find this information in any of the standards publications you would have to refer to several sections because of the way the information is segregated. All of the information regarding parameters has been grouped here in one section for easy reference.

Access Transport

The access transport parameter provides the parameters per the ANSI T1.607 standard. The method of transport parameter is found in the IAM as well as the ACM.

Automatic Congestion Level	**H**	**G**	**F**	**E**		**D**	**C**	**B**	**A**
Spare	0	0	0	0		0	0	0	0
Automatic congestion level 1	0	0	0	0		0	0	0	1
Automatic congestion level 2	0	0	0	0		0	0	1	0
Automatic congestion level 3	0	0	0	0		0	0	1	1
Spare	0	0	0	0		0	1	0	0
					to				
	1	1	1	1		1	1	1	1

This is an optional parameter that may be sent in an REL to indicate the level of congestion at the originating exchange. The meanings of the levels indicated are implementation dependent. The action taken by the receiving exchange is also implementation dependent. This parameter simply provides an alerting mechanism to show that some level of congestion exists in another exchange.

Backward Call Indicators	**H**	**G**	**F**	**E**	**D**	**C**	**B**	**A**
Octet 1								
Charge indicators								
No indication							0	0
No charge							0	1
*Charge							1	0
Spare							1	1
Called party's status indicator								
No indication					0	0		
Subscriber free					0	1		
*Connect when free					1	0		
Excessive delay					1	1		
Called party's category indicator								
No indication			0	0				
Ordinary subscriber			0	1				
*Pay phone			1	0				
Spare			1	1				
End-to-end method indicator								
No end-to-end method available	0	0						
Pass-along method available	0	1						
SCCP method available	1	0						
Pass-along and SCCP methods available	1	1						
Octet 2	**H**	**G**	**F**	**E**	**D**	**C**	**B**	**A**
Interworking indicator								
No interworking encountered								0
Interworking encountered								1
IAM segmentation indicator								
No indication							0	
Additional info has been received and added to IAM							1	

	H	G	F	E	D	C	B	A
ISUP indicator								
ISUP not used all the way (end to end)						0		
ISUP used all the way (end to end)						1		
Holding indicator								
Holding not required					0			
*Holding required					1			
ISDN access indicator								
Terminating access non-ISDN				0				
Terminating access ISDN				1				
Echo control device indicator								
Incoming half-echo control device not included			0					
Incoming half-echo control device included			1					
SCCP method indicator								
No indication	0	0						
*Connectionless method available	0	1						
*Connection-oriented method available	1	0						
*Connectionless and connection-oriented available	1	1						

The backward call indicators parameter is sent in the backward direction (back to the originating exchange) to provide information regarding charging, the status of the called party, and various other forms of information that may be needed to complete the processing of a call.

The charge indicator indicates whether or not the call is a chargeable call. If the call is chargeable, then the number to which the call is to be charged is also provided. The status parameter indicates whether or not the called party is available. If not, then no indication is given in the charge field and the calling party is returned a busy tone.

The category indicator indicates the category (pay phone, for example) of the called party. This information is used during call processing and may imply a special handling of the call (for example, a pay phone may require special handling for billing and operator assistance).

The end-to-end indicator is used by the protocol to designate the method of ISUP signaling available for this call. The protocol can then use this information to make a decision as to which method will be used. In ANSI networks, the pass-along method is most widely used. The SCCP method is not supported in ANSI networks.

Interworking encountered means that another signaling system other than SS7 will be encountered for this call. For example, an exchange may still be using MF signaling and is needed to complete the path for this call. In this case, the SS7 interworking procedures discussed early in this chapter will have to be used.

The ISUP indicator will tell whether or not ISUP is used through all segments of the network involved with this call. If this field shows that ISUP is used all the way, there will be no interworking encountered, and vice versa.

This parameter also indicates whether or not ISDN is the interface to the subscriber. If ISDN is the subscriber interface, then special handling is required at the interface to signal the subscriber being called. There is direct mapping for SS7 signaling to ISDN because the ISDN protocol was developed as an extension of SS7.

Echo cancelers are indicated in the echo control device indicator. This information may be used by the originating switch when encoding voice signals to prevent original signals from being mistaken for noise and may trigger special digitization of the voice.

The last field is the SCCP method indicator, which indicates the type of SCCP method available for this call. Because SCCP services are not supported with ISUP in ANSI networks, this should only be found in private and international networks.

Business Group

Octet 1	**H**	**G**	**F**	**E**	**D**	**C**	**B**	**A**
Party Selector								
No indication					0	0	0	0
Calling party number					0	0	0	1
Called party number					0	0	1	0
Connected party number					0	0	1	1
Redirecting number					0	1	0	0
Original called number					0	1	0	1
Spare					0	1	1	0
							to	
					1	1	1	1
Line privilege information indicator								
Fixed line privileges				0				
Customer-defined line privileges				1				
Business group identifier type								
Multilocation business group identifier			0					
Interworking with private networks identifier			1					
Attendant status								
No indication		0						
Attendant line		1						
Spare	0							

Octets 2, 3, and 4	**H**	**G**	**F**	**E**	**D**	**C**	**B**	**A**
Business group identifier								
No indication	0	0	0	0	0	0	0	0
Public network	0	0	0	0	0	0	0	1
Network dependent	0	0	0	0	0	0	1	0
					to			
	1	1	1	1	1	1	1	1

Binary representation of the actual number assigned may be three octets long.

Octets 5 and 6	**H**	**G**	**F**	**E**	**D**	**C**	**B**	**A**
Subgroup identifier								
No subgroups	0	0	0	0	0	0	0	0

All other values represent the number assigned to any subgroups and may be two octets long.

Octet 7	**H**	**G**	**F**	**E**		**D**	**C**	**B**	**A**
If line privileges information indicator = 0									
Terminating line privileges									
Unrestricted						0	0	0	0
Semirestricted						0	0	0	1
Fully restricted						0	0	1	0
Fully restricted, intraswitch						0	0	1	1
Denied						0	1	0	0
Spare						0	1	0	1
							to		
						1	1	1	1
Originating restrictions									
Unrestricted	0	0	0	0					
Semirestricted	0	0	0	1					
Fully restricted	0	0	1	0					
Fully restricted, intraswitch	0	0	1	1					
Denied	0	1	0	0					
Spare	0	1	0	1					
		to							
	1	1	1	1					
If line privileges information indicator = 1									
Customer-defined line privilege codes	0	0	0	0		0	0	0	0
					to				
	1	1	1	1		1	1	1	1

The business group parameter identifies the properties of a group of subscriber lines that belong to a common subscriber (such as Centrex services). The business group is assigned specific features and restrictions, which must be identified through the protocol during call setup and information sharing between exchanges.

The party indicator identifies the type of number, the called or calling number, the original number called (in the case of forwarded numbers), and connected numbers. The fixed line privileges indicate the type of restrictions to be applied, those already defined (fixed) by the protocol, or those defined by the customer.

The business group identifier indicates whether the business group is one of multiple locations or if the group requires interworking with a private network. The attendant status is also indicated when the business group attendant places calls through the network.

The business group identifier and subgroup identifiers are the actual numbers assigned to these groups by the network provider. The service provider allocates the business group numbers.

Call Reference

The call reference is a number assigned to a call for the tracking of messages and for use as a reference for additional information exchanged between two offices during the duration of the call. The first octet is the call identity number, which uniquely identifies each call. The number is of local significance only. The second octet bears the point code that assigned the call identity number. The point code indicated would be the only point code for which the identity number is of any significance.

Called Party Number

Octet 1	H	G	F	E		D	C	B	A
Nature of address indicator									
Spare		0	0	0		0	0	0	0
Subscriber number		0	0	0		0	0	0	1
Spare, reserved for national use		0	0	0		0	0	1	0
National significant number		0	0	0		0	0	1	1
International number		0	0	0		0	1	0	0
Spare		0	0	0		0	1	0	1
					to				
		1	1	1		0	0	0	0
Subscriber number, operator requested		1	1	1		0	0	0	1
National number, operator requested		1	1	1		0	0	1	0
International number, operator requested		1	1	1		0	0	1	1
No number present, operator requested		1	1	1		0	1	0	0
No number present, cut-through call to carrier		1	1	1		0	1	0	1
950 + call from local exchange carrier public station, hotel/motel, or nonexchange access end office		1	1	1		0	1	1	0
Test line code		1	1	1		0	1	1	1
Reserved for network-specific use		1	1	1		1	0	0	0
					to				
		1	1	1		1	1	1	0
Spare		1	1	1		1	1	1	1
Odd/even bits									
Even number of address signals	0								
Odd number of address signals	1								

Octet 2	H	G	F	E	D	C	B	A
Reserved					0	0	0	0
Numbering plan								
Unknown numbering plan		0	0	0				
ISDN numbering plan (Rec. E.164, E.163)		0	0	1				
Spare		0	1	0				
Reserved ITU-TS data numbering plan		0	1	1				
Reserved ITU-TS telex numbering plan		1	0	0				
Private numbering plan		1	0	1				

Spare		1	1	0				
Spare		1	1	1				
Spare	0							
Octet 3-n	**H**	**G**	**F**	**E**	**D**	**C**	**B**	**A**
Address signal—first address								
Digit 0					0	0	0	0
Digit 1					0	0	0	1
Digit 2					0	0	1	0
Digit 3					0	0	1	1
Digit 4					0	1	0	0
Digit 5					0	1	0	1
Digit 6					0	1	1	0
Digit 7					0	1	1	1
Digit 8					1	0	0	0
Digit 9					1	0	0	1
Spare					1	0	1	0
Code 11					1	0	1	1
Code 12					1	1	0	0
Spare					1	1	0	1
Spare					1	1	1	0
End of pulse signal					1	1	1	1
Address signal—second address								
Digit 0	0	0	0	0				
Digit 1	0	0	0	1				
Digit 2	0	0	1	0				
Digit 3	0	0	1	1				
Digit 4	0	1	0	0				
Digit 5	0	1	0	1				
Digit 6	0	1	1	0				
Digit 7	0	1	1	1				
Digit 8	1	0	0	0				
Digit 9	1	0	0	1				
Spare	1	0	1	0				
Code 11	1	0	1	1				
Code 12	1	1	0	0				
Spare	1	1	0	1				
Spare	1	1	1	0				
End of pulse signal	1	1	1	1				

The called party address can be as large as needed to accommodate the dialed digits. The previous list only defines the first three octets, although the actual parameter will be larger. Each of the dialed digits uses 4 bits. The purpose of this parameter is to provide the distant exchange of the number of the called party (dialed digits) or the LRN associated with the called party. This parameter can be used in a number of message types and is usually used when establishing a connection from one exchange to another.

If the dialed digits are to a ported number, the nature of address indicator will indicate a ported number. If this parameter contains the LRN to be used for routing, the dialed digits are placed in the GAP parameter.

Calling Party Number

Octet 1	H	G	F	E		D	C	B	A
Nature of address indicator									
Spare		0	0	0		0	0	0	0
Unique subscriber number		0	0	0		0	0	0	1
Spare, reserved for national use		0	0	0		0	0	1	0
Unique national significant number		0	0	0		0	0	1	1
Unique international number		0	0	0		0	1	0	0
Spare		0	0	0		0	1	0	1
					to				
		1	1	1		0	0	0	0
Nonunique subscriber number		1	1	1		0	0	0	1
Spare, reserved for national use		1	1	1		0	0	1	0
Nonunique national number		1	1	1		0	0	1	1
Nonunique international number		1	1	1		0	1	0	0
Spare		1	1	1		0	1	0	1
Spare		1	1	1		0	1	1	0
Test line code		1	1	1		0	1	1	1
Reserved for network-specific use		1	1	1		1	0	0	0
					to				
		1	1	1		1	1	1	0
Spare		1	1	1		1	1	1	1
Odd/even indicator									
Even number of address signals	0								
Odd number of address signals	1								

Octet 2	H	G	F	E	D	C	B	A
Screening								
Reserved (for user provided, not screened)							0	1
User provided, screening passed							1	0
Network provided							1	1
Address presentation								
Presentation allowed					0	0		
Presentation restricted					0	1		
Spare					1	0		
Spare					1	1		
Numbering plan								
Unknown numbering plan		0	0	0				
ISDN numbering plan (Rec. E.164, E.163)		0	0	1				
Spare		0	1	0				
Reserved ITU-TS data numbering plan		0	1	1				
Reserved ITU-TS telex numbering plan		1	0	0				
Private numbering plan		1	0	1				

Spare		1	1	0				
Spare		1	1	1				
Spare	0							
Octet 3-n	**H**	**G**	**F**	**E**	**D**	**C**	**B**	**A**
Address signal—first address								
Digit 0					0	0	0	0
Digit 1					0	0	0	1
Digit 2					0	0	1	0
Digit 3					0	0	1	1
Digit 4					0	1	0	0
Digit 5					0	1	0	1
Digit 6					0	1	1	0
Digit 7					0	1	1	1
Digit 8					1	0	0	0
Digit 9					1	0	0	1
Spare					1	0	1	0
Code 11					1	0	1	1
Code 12					1	1	0	0
Spare					1	1	0	1
Spare					1	1	1	0
End of pulse signal					1	1	1	1
Address signal—second address								
Digit 0	0	0	0	0				
Digit 1	0	0	0	1				
Digit 2	0	0	1	0				
Digit 3	0	0	1	1				
Digit 4	0	1	0	0				
Digit 5	0	1	0	1				
Digit 6	0	1	1	0				
Digit 7	0	1	1	1				
Digit 8	1	0	0	0				
Digit 9	1	0	0	1				
Spare	1	0	1	0				
Code 11	1	0	1	1				
Code 12	1	1	0	0				
Spare	1	1	0	1				
Spare	1	1	1	0				
End of pulse signal	1	1	1	1				

The calling party's address is much the same as the called party's address with the exception of the second octet. Some differences also exist in the nature of address indicator.

The screening indicator is used to indicate who provided the dialed digits in the address. The digits could have been provided by the subscriber (originating party) or by the network (global title translation). The purpose of this parameter is to provide a means for the network to determine the origin of the digits and whether or not this number is viewed by the network as the true called party number or as an alias. In the case of ported numbers, the parameter will not contain the LRN of the calling party. The LRN of the calling party is found in the *jurisdiction information parameter* (JIP).

The address presentation parameter serves the same purpose as in the calling party address. The default for this parameter is to be "presentation restricted." The presentation of the dialed digits enables called parties to identify who is calling them (caller party ID). When presentation is restricted, the number can still be presented to another network, but it is restricted from being presented to an end user.

Calling Party's Category	**H**	**G**	**F**	**E**	**D**	**C**	**B**	**A**
Calling party's category unknown (default)	0	0	0	0	0	0	0	0
*French language operator	0	0	0	0	0	0	0	1
*English language operator	0	0	0	0	0	0	1	0
*German language operator	0	0	0	0	0	0	1	1
*Russian language operator	0	0	0	0	0	1	0	0
*Spanish language operator	0	0	0	0	0	1	0	1
Reserved for network-dependent language selection	0	0	0	0	0	1	1	0
	0	0	0	0	0	1	1	1
	0	0	0	0	1	0	0	0
National networks—operator service	0	0	0	0	1	0	0	1
Ordinary calling subscriber	0	0	0	0	1	0	1	0
*Calling subscriber with priority	0	0	0	0	1	0	1	1
*Data call (voiceband data)	0	0	0	0	1	1	0	0
Test call	0	0	0	0	1	1	0	1
Spare	0	0	0	0	1	1	1	0
*Pay phone	0	0	0	0	1	1	1	1
Spare (ITU-TS)	0	0	0	1	0	0	0	0
to								
	1	1	0	1	1	1	1	1
Emergency service call in progress	1	1	1	0	0	0	0	0
High-priority call indication	1	1	1	0	0	0	0	1
National security and emergency preparedness call	1	1	1	0	0	0	1	0
Spare (ANSI)	1	1	1	0	0	0	1	1
to								
	1	1	1	0	1	1	1	1
Network-specific use	1	1	1	1	0	0	0	0
to								
	1	1	1	1	1	1	1	0
Reserved for expansion	1	1	1	1	1	1	1	1

The calling party's category indicates the type of subscriber originating the call. In the case of a special-language operator, the originator of the call (the operator services) will require special handling. The same is true of the pay phone, which may require special operator assistance.

The test call indicator in this parameter is used for the remote testing of translations. A technician may initiate a test call from a remote maintenance center terminal to test and verify newly added translations or routing information. The IAM would then include the calling party's category parameter with the test call value.

Carrier Identification

Octet 1	H	G	F	E	D	C	B	A
Network identification plan								
Unknown					0	0	0	0
Three-digit carrier identification code					0	0	0	1
Four-digit carrier identification code					0	0	1	0
Spare					0	0	1	1
						to		
					1	1	1	1
Type of network identification		0	0	0				
Spare		0	0	0				
Spare		0	0	1				
National network identification		0	1	0				
Spare		0	1	1				
			to					
		1	1	1				
Spare	0							

Octet 2	H	G	F	E	D	C	B	A
Digit one								
Digit 0					0	0	0	0
Digit 1					0	0	0	1
Digit 2					0	0	1	0
Digit 3					0	0	1	1
Digit 4					0	1	0	0
Digit 5					0	1	0	1
Digit 6					0	1	1	0
Digit 7					0	1	1	1
Digit 8					1	0	0	0
Digit 9					1	0	0	1
Spare					1	0	1	0
Code 11					1	0	1	1
Code 12					1	1	0	0
Spare					1	1	0	1
Spare					1	1	1	0
End of pulse signal					1	1	1	1
Digit two								
Digit 0	0	0	0	0				
Digit 1	0	0	0	1				
Digit 2	0	0	1	0				
Digit 3	0	0	1	1				
Digit 4	0	1	0	0				
Digit 5	0	1	0	1				
Digit 6	0	1	1	0				
Digit 7	0	1	1	1				
Digit 8	1	0	0	0				
Digit 9	1	0	0	1				
Spare	1	0	1	0				

Code 11	1	0	1	1				
Code 12	1	1	0	0				
Spare	1	1	0	1				
Spare	1	1	1	0				
End of pulse signal	1	1	1	1				
Octet 3	**H**	**G**	**F**	**E**	**D**	**C**	**B**	**A**
Digit three								
Digit 0					0	0	0	0
Digit 1					0	0	0	1
Digit 2					0	0	1	0
Digit 3					0	0	1	1
Digit 4					0	1	0	0
Digit 5					0	1	0	1
Digit 6					0	1	1	0
Digit 7					0	1	1	1
Digit 8					1	0	0	0
Digit 9					1	0	0	1
Spare					1	0	1	0
Code 11					1	0	1	1
Code 12					1	1	0	0
Spare					1	1	0	1
Spare					1	1	1	0
End of pulse signal					1	1	1	1
Digit four (if four-digit code is used)								
Digit 0	0	0	0	0				
Digit 1	0	0	0	1				
Digit 2	0	0	1	0				
Digit 3	0	0	1	1				
Digit 4	0	1	0	0				
Digit 5	0	1	0	1				
Digit 6	0	1	1	0				
Digit 7	0	1	1	1				
Digit 8	1	0	0	0				
Digit 9	1	0	0	1				
Spare	1	0	1	0				
Code 11	1	0	1	1				
Code 12	1	1	0	0				
Spare	1	1	0	1				
Spare	1	1	1	0				
End of pulse signal	1	1	1	1				
Carrier Selection Information	**H**	**G**	**F**	**E**	**D**	**C**	**B**	**A**
Carrier selection information								
No indication (default)	0	0	0	0	0	0	0	0
Subscriber's designated (preselected) carrier	0	0	0	0	0	0	0	1
Subscriber's designated carrier as input by caller	0	0	0	0	0	0	1	0
Subscriber's designated carrier (undetermined)	0	0	0	0	0	0	1	1

Carrier designated by caller at time of call	0	0	0	0	0	1	0	0
Spare	0	0	0	0	0	1	0	1
				to				
	1	1	1	1	1	1	1	0
Reserved	1	1	1	1	1	1	1	1

The carrier identification parameter is used to identify the carrier selected by the caller. The selection can be accomplished in one of several ways. All subscribers have preselected carriers listed in the *Line Information Database* (LIDB) as a part of their customer record. This is the carrier that will be used for all long-distance calls, except when another carrier is selected manually.

Any carrier can be selected by dialing the carrier access code. The carrier access code is the 10xxx number assigned to all long-distance carriers. This number enables a caller to use any long-distance provider for that call. After the call is finished, the next call defaults back to the preselected carrier.

A subscriber may also be calling from a pay phone or some other phone where a long-distance carrier has not been preselected, but the caller has entered in a 10xxx code for carrier selection. This is a rarity in U.S. networks because equal access has required that all subscribers have a designated carrier preselected for every line.

Cause Indicators

Octet 1	**H**	**G**	**F**	**E**	**D**	**C**	**B**	**A**
Location								
User					0	0	0	0
Local private network					0	0	0	1
Local public network					0	0	1	0
Transit network					0	0	1	1
Remote local network					0	1	0	0
Remote private network					0	1	0	1
Local interface controlled by this signaling link					0	1	1	0
International network					0	1	1	1
Network beyond interworking point					1	0	1	0
Spare				0				
Coding standard								
ITU-TS standard (default)		0	0					
Reserved for other international standards		0	1					
ANSI standard		1	0					
Reserved		1	1					
Extension bit								
Parameter continues to next octet	0							
Last octet	1							

Octet 2

The cause value provides the reason for the message failure. These cause codes are grouped as ITU-TS and ANSI codes. All cause codes are divided into two parts; bits ABCD represent the cause and bits EFG represent the class.

ITU-TS Cause Codes (Coding Standard 0 0)

Class 0 0 0 and 0 0 1, Normal Event	H	G	F	E	D	C	B	A
Unallocated (unassigned) number		0	0	0	0	0	0	1
No route to specified transit network		0	0	0	0	0	1	0
No route to destination		0	0	0	0	0	1	1
Send special information tone		0	0	0	0	1	0	0
*Misdialed trunk prefix		0	0	0	0	1	0	1
Preemption		0	0	0	1	0	0	0
Preemption—circuit reserved for reuse		0	0	0	1	0	0	1
Normal clearing		0	0	1	0	0	0	0
User busy		0	0	1	0	0	0	1
No user responding		0	0	1	0	0	1	0
No answer from user (user alerted)		0	0	1	0	0	1	1
Subscriber absent		0	0	1	0	1	0	0
Call rejected		0	0	1	0	1	0	1
Number changed		0	0	1	0	1	1	0
Redirect to new destination		0	0	1	0	1	1	1
Destination out of order		0	0	1	1	0	1	1
Address incomplete		0	0	1	1	1	0	0
Facility rejected		0	0	1	1	1	0	1
Normal—unspecified (default)		0	0	1	1	1	1	1

Class 0 1 0, Resource Unavailable	H	G	F	E	D	C	B	A
No circuit/channel available		0	1	0	0	0	1	0
Network out of order		0	1	0	0	1	1	0
Temporary failure		0	1	0	1	0	0	1
Switching equipment congestion		0	1	0	1	0	1	0
Access information discarded		0	1	0	1	0	1	1
Requested circuit/channel not available		0	1	0	1	1	0	0
Precedence call blocked		0	1	0	1	1	1	0
Resource unavailable—unspecified (default)		0	1	0	1	1	1	1

Class 0 1 1, Service or Option Not Available	H	G	F	E	D	C	B	A
Requested facility not subscribed		0	1	1	0	0	1	0
*Outgoing calls barred within closed user group		0	1	1	0	1	0	1
*Incoming calls barred within closed user group		0	1	1	0	1	1	1
Bearer capability not authorized		0	1	1	1	0	0	1
Bearer capability not currently available		0	1	1	1	0	1	0
Inconsistency in designated outgoing access and subscriber class		0	1	1	1	1	1	0
Service option not available—unspecified (default)		0	1	1	1	1	1	1

Class 1 0 0, Service or Option Not Implemented	H	G	F	E	D	C	B	A
Bearer capability not implemented		1	0	0	0	0	0	1
Requested facility not implemented		1	0	0	0	1	0	1
Only restricted digital info bearer capability available		1	0	0	0	1	1	0
Service not implemented—unspecified (default)		1	0	0	1	1	1	1

Class 1 0 1, Invalid Message	H	G	F	E	D	C	B	A
*User not member of closed user group		1	0	1	0	1	1	1
Incompatible destination		1	0	1	1	0	0	0
*Nonexistent closed user group		1	0	1	1	0	1	0
Invalid transit network selection		1	0	1	1	0	1	1
Invalid message, unspecified (default)		1	0	1	1	1	1	1

Class 1 1 0, Protocol Error (Unknown Message)	H	G	F	E	D	C	B	A
Message type nonexistent or not implemented		1	1	0	0	0	0	1
Information element parameter nonexistent/ not implemented		1	1	0	0	0	1	1
Recovery on timer expiration		1	1	0	0	1	1	0
Parameter nonexistent/not implemented—passed on		1	1	0	0	1	1	1
Message with unrecognized parameter discarded		1	1	0	1	1	1	0
Protocol error, unspecified (default)		1	1	0	1	1	1	1

Class 1 1 1, Interworking Class	H	G	F	E	D	C	B	A
Interworking, unspecified (default)		1	1	1	1	1	1	1

ANSI Cause Codes (Coding Standard 1 0)

Class 0 0 0 and 0 0 1, Normal Event	H	G	F	E	D	C	B	A
Unallocated destination number		0	0	1	0	1	1	1
Unknown business group		0	0	1	1	0	0	0
Exchange routing error		0	0	1	1	0	0	1
Misrouted call to a ported number		0	0	1	1	0	1	0
Number portability Query on Release (QoR) number not found		0	0	1	1	0	1	1
Class 0 1 0, Resource Unavailable								
Preemption		0	1	0	1	1	0	1
Precedence call blocked		0	1	0	1	1	1	0

Class 0 1 1, Service or Option Not Available	H	G	F	E	D	C	B	A
Call type incompatibility with service request		0	1	1	0	0	1	1
Call blocked due to group restrictions		0	1	1	0	1	1	0
Extension bit	0							

Octet 3	H	G	F	E	D	C	B	A
Diagnostics (if applicable)	0	0	0	0	0	0	0	0

The diagnostics field is dependent on the cause value. Not all cause codes will require a diagnostics field afterward. The diagnostic field uses the same format as the specified parameters (that is, the called party number). The following lists the cause codes that generate a diagnostics field and the parameter structure used for the diagnostic value.

Cause Code	Diagnostic	Structure
0 0 1 0 1 1 0	Called party number (new)	See called party parameter
0 1 0 0 1 1 0	Transit network identity	Transit network selection
0 1 0 1 0 1 0	Transit network identity	Transit network selection
0 1 1 1 0 0 1	Attribute identity	See below

0	1	1	1	0	1	0	Attribute identity	See below
1	0	0	0	0	0	1	Attribute identity	See below

Attribute Identity	H	G	F	E	D	C	B	A
Information transfer capability		0	1	1	0	0	0	1
Information transfer mode		0	1	1	0	0	1	0
Information transfer rate		0	1	1	0	0	1	1
Structure		0	1	1	0	1	0	0
Configuration		0	1	1	0	1	0	1
Establishment		0	1	1	0	1	1	0
Symmetry		0	1	1	0	1	1	1
Information transfer rate (destination to origination)		0	1	1	1	0	0	0
Layer identification and corresponding user info		0	1	1	1	0	0	1

This rather lengthy parameter is full of variables and is dependent on the cause as to the full contents of the parameter. The parameter can be found in RELs, ACMs, or CONs. The purpose is to identify the cause of the failure, disconnect, or message rejection. Appendix C provides full explanations for all cause codes and diagnostics.

Charge Number

Octet 1	H	G	F	E		D	C	B	A
Nature of address indicator									
Spare		0	0	0		0	0	0	0
Automatic number identification (ANI) of the calling party; subscriber number		0	0	0		0	0	0	1
ANI not available or not provided		0	0	0		0	0	1	0
ANI of the calling party; national number		0	0	0		0	0	1	1
Spare		0	0	0		0	1	0	0
ANI of the called party; subscriber number		0	0	0		0	1	0	1
ANI of the called party; no number present		0	0	0		0	1	1	0
ANI of the called party; national number		0	0	0		0	1	1	1
Spare		0	0	0		1	0	0	0
					to				
		1	1	1		0	1	1	1
Reserved for network-specific use		1	1	1		1	0	0	0
					to				
		1	1	1		1	1	1	0
Spare		1	1	1		1	1	1	1
Odd/even bit									
Even number of address signals	0								
Odd number of address signals	1								

Octet 2	H	G	F	E	D	C	B	A
Reserved					0	0	0	0
Numbering plan								
Unknown		0	0	0				
ISDN numbering plan (Rec E.164, E.163)		0	0	1				

Spare		0	1	0				
Reserved (ITU data numbering plan)		0	1	1				
Reserved (ITU telex numbering plan)		1	0	0				
Private numbering plan		1	0	1				
Spare		1	1	0				
Spare		1	1	1				
Spare	0							
Octet 3	**H**	**G**	**F**	**E**	**D**	**C**	**B**	**A**
First address signal								
Digit 0					0	0	0	0
Digit 1					0	0	0	1
Digit 2					0	0	1	0
Digit 3					0	0	1	1
Digit 4					0	1	0	0
Digit 5					0	1	0	1
Digit 6					0	1	1	0
Digit 7					0	1	1	1
Digit 8					1	0	0	0
Digit 9					1	0	0	1
Spare					1	0	1	0
Code 11					1	0	1	1
Code 12					1	1	0	0
Second address signal								
Digit 0	0	0	0	0				
Digit 1	0	0	0	1				
Digit 2	0	0	1	0				
Digit 3	0	0	1	1				
Digit 4	0	1	0	0				
Digit 5	0	1	0	1				
Digit 6	0	1	1	0				
Digit 7	0	1	1	1				
Digit 8	1	0	0	0				
Digit 9	1	0	0	1				
Spare	1	0	1	0				
Code 11	1	0	1	1				
Code 12	1	1	0	0				
Spare	1	1	0	1				
Spare	1	1	1	0				
End of pulse signal	1	1	1	1				

Circuit Assignment Map

Octet 1	**H**	**G**	**F**	**E**	**D**	**C**	**B**	**A**
Map type								
Spare			0	0	0	0	0	0
DS1 map format			0	0	0	0	0	1
Spare			0	0	0	0	1	0
				to				
			1	1	1	1	1	1
Spare	0	0						

Octets 2–4	H	G	F	E	D	C	B	A
Map	x	x	x	x	x	x	x	x
0 = 64 kbps circuit is not used								
1 = 64 kbps circuit is used								

The map portion of this parameter provides a 1-bit representation for each circuit. If the value of the circuit bit is 1, then the 64 kbps circuit is used. If the value is 0, it is not used. Up to 24 circuits may be represented in the map fields. All 24 circuits are represented, but set to 0 if the circuits are not used. This means that this parameter is always a fixed length and not variable.

This parameter is only used when DS0 circuits that are not contiguous are used. Noncontiguous means that a full DS1 is not being utilized for these circuits. There may be a partial DS1 or the various DS0s may be split between multiple DS1s. Either way, the map indicates which circuits are or are not used within a DS1.

Circuit Group Supervision Msg Type Indicator

	H	G	F	E	D	C	B	A
Circuit group blocking type indicator								
Block without release							0	0
Block with immediate release							0	1
Reserved for national use							1	0
Spare							1	1
Spare	0	0	0	0	0	0		

This parameter provides instructions to another exchange on the method of circuit blocking to be implemented on the designated circuit. Blocking enables craft personnel to perform tests on a circuit without the change of the circuit being seized for another call.

There are two methods of blocking: blocking with a release (which disconnects any call in progress) and blocking without release (which will not send an REL through the network).

Circuit Identification Name

Octet 1	H	G	F	E	D	C	B	A
Trunk number (first digit)		x	x	x	x	x	x	x
Spare	0							

Octets 2–4	H	G	F	E	D	C	B	A
Trunk number (digits two through four)		x	x	x	x	x	x	x
Spare	0							

Octet 5	H	G	F	E	D	C	B	A
CLLI code—office A		x	x	x	x	x	x	x
Spare	0							

Octets 6–26	H	G	F	E	D	C	B	A
CLLI code—office Z		x	x	x	x	x	x	x
Spare	0							

The circuit identification name parameter is used to identify the CLLI of a specific trunk to a distant exchange. This parameter is coded using IA5 characters, with each character using one octet.

Office A is designated as follows:

- If the trunk is a one-way trunk group, the office that originates the calls for this trunk is office A.
- If the trunk is a two-way trunk, the office with the lower alphanumeric CLLI code is office A.

The same rules apply to subgroups that can be one-way or two-way trunks. This parameter enables two exchanges to exchange information regarding their identity for use in routing tables.

Circuit State Indicator	**H**	**G**	**F**	**E**		**D**	**C**	**B**	**A**
Transient	0	0	0	0		0	0	0	0
Spare	0	0	0	0		0	0	0	1
Spare	0	0	0	0		0	0	1	0
Unequipped	0	0	0	0		0	0	1	1
Incoming circuit busy, active	0	0	0	0		0	1	0	0
Incoming circuit busy, locally blocked	0	0	0	0		0	1	0	1
Incoming circuit busy, remotely blocked	0	0	0	0		0	1	1	0
Incoming circuit busy, locally and remotely blocked	0	0	0	0		0	1	1	1
Outgoing circuit busy, active	0	0	0	0		1	0	0	0
Outgoing circuit busy, locally blocked	0	0	0	0		1	0	0	1
Outgoing circuit busy, remotely blocked	0	0	0	0		1	0	1	0
Outgoing circuit busy, locally and remotely blocked	0	0	0	0		1	0	1	1
Idle	0	0	0	0		1	1	0	0
Idle, locally blocked	0	0	0	0		1	1	0	1
Idle, remotely blocked	0	0	0	0		1	1	1	0
Idle, locally and remotely blocked	0	0	0	0		1	1	1	1
Spare	0	0	0	1		0	0	0	0
					to				
	1	1	1	1		1	1	1	1

This parameter enables exchanges to send status information regarding specific trunk circuits, providing the status of the circuit in the distant exchange's perspective. The CIC is carried in the field after the routing label.

This parameter may be from one octet up to *n* octets in length. There may be situations when a two-way trunk will have more than one status indicator. For example, a two-way trunk can be incoming circuit busy, active and also be outgoing circuit busy, locally blocked.

Circuit Validation Response Indicator	**H**	**G**	**F**	**E**		**D**	**C**	**B**	**A**
Successful	0	0	0	0		0	0	0	0
Failure (default)	0	0	0	0		0	0	0	1
Spare	0	0	0	0		0	0	1	0
					to				
	1	1	1	1		1	1	1	1

The circuit validation response indicator provides the results of a circuit validation in response to a distant exchange request.

Common Language Location Identifier (CLLI)

Octet 1	**H**	**G**	**F**	**E**	**D**	**C**	**B**	**A**
Town (first character)		0	0	0	0	0	0	0
Spare	0							

Octets 2–4	**H**	**G**	**F**	**E**	**D**	**C**	**B**	**A**
Town (second through fourth characters)		x	x	x	x	x	x	x
Spare	0							

Octet 5	**H**	**G**	**F**	**E**	**D**	**C**	**B**	**A**
State (first character)		x	x	x	x	x	x	x
Spare	0							

Octet 6	**H**	**G**	**F**	**E**	**D**	**C**	**B**	**A**
State (second character)		x	x	x	x	x	x	x
Spare	0							

Octet 7	**H**	**G**	**F**	**E**	**D**	**C**	**B**	**A**
Building (first character)		x	x	x	x	x	x	x
Spare	0							

Octet 8	**H**	**G**	**F**	**E**	**D**	**C**	**B**	**A**
Building (second character)		x	x	x	x	x	x	x
Spare	0							

Octet 9	**H**	**G**	**F**	**E**	**D**	**C**	**B**	**A**
Building subdivision (first character)		x	x	x	x	x	x	x
Spare	0							

Octet 10	**H**	**G**	**F**	**E**	**D**	**C**	**B**	**A**
Building subdivision (second character)		x	x	x	x	x	x	x
Spare	0							

Octet 11	**H**	**G**	**F**	**E**	**D**	**C**	**B**	**A**
Building subdivision (third character)		x	x	x	x	x	x	x
Spare	0							

The CLLI parameter is used during circuit validation to identify an exchange. All signaling points on the ANSI SS7 network must have a CLLI code. This provides the means for identifying by location where a particular signaling point is located. A typical CLLI may look like RLGHNCXA03W. All characters are IA5 characters.

Connection Request

The connection request parameter is sent in the forward direction for the SCCP function. This enables the ISUP protocol to establish an end-to-end connection on which the SCCP may send TCAP or ISUP messages (if ISUP is using the service of SCCP).

The *local reference number* (LRN) is the number assigned for the specific call and is used as a reference in the originating exchange. The LRN enables the exchange to monitor all messages and associate them to their proper calls.

The protocol class field identifies the protocol class to be used on this end-to-end connection. The protocol class is directly related to the SCCP protocol (refer to Chapter 7, "General Description of SCCP Functions," for more details on protocol class). The protocol class specifies whether or not the services on this connection will be connection oriented or connectionless.

The credit field is used for changing the window size of the exchange during the connection. This is only valid if class 3 or 4 is specified (connection-oriented services).

Continuity Indicators	**H**	**G**	**F**	**E**	**D**	**C**	**B**	**A**
Continuity indicator								
Continuity check failed								0
Continuity check successful								1
Spare	0	0	0	0	0	0	0	

The continuity indicator parameter indicates whether or not a continuity test was successful. The continuity check may be requested by an originating exchange according to predetermined criteria. The criteria for conducting a continuity check are found in the continuity check requirements indicator field of the circuit group characteristic indicator parameter.

Egress Service

This parameter is used to send network-specific information regarding a terminating exchange such as the interexchange carrier, the type of terminating access service, and the point of interconnection. This information is sent in the forward direction by the first incoming exchange to the terminating exchange.

End of Optional Parameters Fields Indicator	**H**	**G**	**F**	**E**	**D**	**C**	**B**	**A**
End of optional parameters	0	0	0	0	0	0	0	0

The end of optional parameters parameter is the last octet in a message containing any optional parameters.

Event Information	**H**	**G**	**F**	**E**	**D**	**C**	**B**	**A**
Event indicator								
Spare		0	0	0	0	0	0	0
Altering		0	0	0	0	0	0	1
Progress		0	0	0	0	0	1	0
In-band info or appropriate pattern now available		0	0	0	0	0	1	1
*Call forwarded on busy		0	0	0	0	1	0	0
*Call forwarded on no reply		0	0	0	0	1	0	1
*Call forwarded unconditional		0	0	0	0	1	1	0
Call deflected		0	0	0	0	1	1	1
Notification for supplementary service		0	0	0	1	0	0	0

Spare		0	0	0		1	0	0	1
					to				
		1	1	0		1	1	1	0
Service information included		1	1	0		1	1	1	1
Spare		1	1	1		0	0	0	0
					to				
		1	1	1		1	1	1	0
Reserved		1	1	1		1	1	1	1

	H	G	F	E	D	C	B	A
Event presentation restricted indicator (restrict)								
No indication	0							
Presentation restricted	1							

Forward Call Indicators

Octet 1	H	G	F	E	D	C	B	A
Incoming international call indicator								
Not an incoming international call								0
Incoming international call								1

	H	G	F	E	D	C	B	A
End-to-end method indicator								
No end-to-end method available						0	0	
Pass-along method available						0	1	
SCCP method available						1	0	
Pass-along and SCCP methods available						1	1	

	H	G	F	E	D	C	B	A
Interworking indicator								
No interworking encountered (SS7 all the way)					0			
Interworking encountered					1			

	H	G	F	E	D	C	B	A
IAM segmentation indicator								
No indication				0				
Additional info being sent by unsolicited info msg				1				

	H	G	F	E	D	C	B	A
ISUP indicator								
ISUP not used all the way			0					
ISUP used all the way			1					

	H	G	F	E	D	C	B	A
ISUP preference indicator								
ISUP preferred all the way (default)	0	0						
ISUP not required all the way;	0	1						
ISUP required all the way			1	0				
Spare	1	1						

Octet 2	H	G	F	E	D	C	B	A
ISDN access indicator								
Originating access non-ISDN								0
Originating access ISDN								1

	H	G	F	E	D	C	B	A
SCCP method indicator								
No indication						0	0	
*Connectionless method available						0	1	
*Connection-oriented method available						1	0	
*Connectionless and connection-oriented available						1	1	
Spare					0			

	H	G	F	E
Ported number translation indicator				
Number not translated				0
Number translated				1
No QoR routing attempt in progress			0	
QoR routing attempt in progress			1	
Reserved for national use	0	0		

The forward call indicators are sent with an IAM to alert the distant exchange of the services required for the call. The international call indicator identifies international calls that have entered through a gateway STP. Without this indicator, it would be difficult for the distant exchange to know if the call was international (mapping of the dialed digits to a conversion table would be necessary).

In addition to identifying international calls, the type of end-to-end signaling method available is also indicated through the end-to-end method indicator. This field identifies the method of signaling available for use; the pass-along method, SCCP method, or both are available. In U.S. networks, only the pass-along method is currently supported.

The interworking indicator identifies any networks encountered along the way that are not SS7 networks. The location of this network is not provided (because that is of no importance). The distant exchange only needs to be aware of its existence.

The IAM segmentation indicator shows when an IAM has been divided into separate messages because of length or any other reason. IAM information can be sent in an additional signal unit after the initial IAM.

The ISUP indicators are used to indicate whether ISUP is used end to end, whether it is required end to end, and whether the subscriber interface at the originating exchange is ISDN.

An SCCP indicator is also provided for those networks using the services of SCCP for supporting the ISUP. The SCCP method of end-to-end signaling is not supported in U.S. networks.

The ported number translation indicator is used with the LNP application. It is used to indicate when a specific number has been looked up in the LNP database and to prevent unnecessary queries.

Generic Address Parameter (GAP)

Octet 1	H	G	F	E		D	C	B	A
Type of address	x	x	x	x		x	x	x	x
Dialed number	0	0	0	0		0	0	0	0
Destination number	0	0	0	0		0	0	0	1
Supplemental user provided calling address—failed network screening	0	0	0	0		0	0	1	0
Supplemental user provided calling address—not screened	0	0	0	0		0	0	1	1
Completion number	0	0	0	0		0	1	0	0
ITU spare	0	0	0	0		0	1	0	1
					to				
	0	1	1	1		1	1	1	1
Network-specific use	1	0	0	0		0	0	0	0
					to				
	1	0	1	1		1	1	1	1
Ported number	1	1	0	0		0	0	0	0
ANSI spare	1	1	0	0		0	0	0	1
					to				
	1	1	1	1		0	1	1	1
Transfer number 6	1	1	1	1		1	0	0	0
Transfer number 5	1	1	1	1		1	0	0	1
Transfer number 4	1	1	1	1		1	0	1	0
Transfer number 3	1	1	1	1		1	0	1	1
Transfer number 2	1	1	1	1		1	1	0	0
Transfer number 1	1	1	1	1		1	1	0	1
Callers Emergency Service Identification (CESID)	1	1	1	1		1	1	1	0
Reserved for expansion	1	1	1	1		1	1	1	1

Octet 2

(For type dialed digits and destination number type of address)

	H	G	F	E		D	C	B	A
Nature of address indicator									
Spare		0	0	0		0	0	0	0
Subscriber number		0	0	0		0	0	0	1
Spare reserved, for national use		0	0	0		0	0	1	0
National (significant number)		0	0	0		0	0	1	1
International number		0	0	0		0	1	0	0
Spare		0	0	0		0	1	0	1
Abbreviated number		0	0	0		0	1	1	0
Spare		0	0	0		0	1	1	1
					to				
		1	1	1		1	1	1	1

(For type supplemental user provided calling address)

	H	G	F	E		D	C	B	A
Nature of address indicator									
Spare		0	0	0		0	0	0	0
Unique subscriber number		0	0	0		0	0	0	1
Spare, reserved for national use		0	0	0		0	0	1	0
Unique national significant number		0	0	0		0	0	1	1
Unique international number		0	0	0		0	1	0	0
Spare		0	0	0		0	1	0	1
					to				
		1	1	1		0	0	0	0
Nonunique subscriber number		1	1	1		0	0	0	1
Spare, reserved for national use		1	1	1		0	0	1	0
Nonunique national number		1	1	1		0	0	1	1
Nonunique international number		1	1	1		0	1	0	0
Spare		1	1	1		0	1	0	1
Spare		1	1	1		0	1	1	0
Test line code		1	1	1		0	1	1	1
Reserved for network-specific use		1	1	1		1	0	0	0
					to				
		1	1	1		1	1	1	0
Spare		1	1	1		1	1	1	1

(For completion number type of address)

	H	G	F	E		D	C	B	A
Nature of address indicator									
Spare		0	0	0		0	0	0	0
Subscriber number		0	0	0		0	0	0	1
Spare, reserved for national use		0	0	0		0	0	1	0
National significant number		0	0	0		0	0	1	1
International number		0	0	0		0	1	0	0
Spare		0	0	0		0	1	0	1
					to				
		1	1	1		0	0	0	0
Subscriber number, operator requested		1	1	1		0	0	0	1
National number, operator requested		1	1	1		0	0	1	0
International number, operator requested		1	1	1		0	0	1	1
No number present, operator requested		1	1	1		0	1	0	0
No number present, cut-through call to carrier		1	1	1		0	1	0	1
950 + call from local exchange carrier public station, hotel/motel, or nonexchange access end office		1	1	1		0	1	1	0

Test line code		1	1	1		0	1	1	1
Reserved for network-specific use		1	1	1		1	0	0	0
					to				
		1	1	1		1	1	1	0
Spare		1	1	1		1	1	1	1
Odd/even indicator									
Even number of address signals	0								
Odd number of address signals	1								
Octet 3	**H**	**G**	**F**	**E**		**D**	**C**	**B**	**A**
Reserved								0	0
Presentation									
Presentation allowed						0	0		
Presentation restricted						0	1		
Spare						1	0		
Spare						1	1		
Numbering plan									
Unknown numbering plan		0	0	0					
ISDN numbering plan (Rec. E.164, E.163)		0	0	1					
Spare		0	1	0					
Reserved ITU-TS data numbering plan		0	1	1					
Reserved ITU-TS telex numbering plan		1	0	0					
Private numbering plan		1	0	1					
Spare		1	1	0					
Spare		1	1	1					
Spare	0								
Octet 4	**H**	**G**	**F**	**E**		**D**	**C**	**B**	**A**
Address signal—first address									
Digit 0						0	0	0	0
Digit 1						0	0	0	1
Digit 2						0	0	1	0
Digit 3						0	0	1	1
Digit 4						0	1	0	0
Digit 5						0	1	0	1
Digit 6						0	1	1	0
Digit 7						0	1	1	1
Digit 8						1	0	0	0
Digit 9						1	0	0	1
Spare						1	0	1	0
Code 11						1	0	1	1
Code 12						1	1	0	0
Spare						1	1	0	1
Spare						1	1	1	0
End of pulse signal						1	1	1	1
Address signal—second address									
Digit 0	0	0	0	0					
Digit 1	0	0	0	1					
Digit 2	0	0	1	0					

Digit 3	0	0	1	1				
Digit 4	0	1	0	0				
Digit 5	0	1	0	1				
Digit 6	0	1	1	0				
Digit 7	0	1	1	1				
Digit 8	1	0	0	0				
Digit 9	1	0	0	1				
Spare	1	0	1	0				
Code 11	1	0	1	1				
Code 12	1	1	0	0				
Spare	1	1	0	1				
Spare	1	1	1	0				
End of pulse signal	1	1	1	1				
Octets 5–n	**H**	**G**	**F**	**E**	**D**	**C**	**B**	**A**
Address signal—first address								
Digit 0					0	0	0	0
Digit 1					0	0	0	1
Digit 2					0	0	1	0
Digit 3					0	0	1	1
Digit 4					0	1	0	0
Digit 5					0	1	0	1
Digit 6					0	1	1	0
Digit 7					0	1	1	1
Digit 8					1	0	0	0
Digit 9					1	0	0	1
Spare					1	0	1	0
Code 11					1	0	1	1
Code 12					1	1	0	0
Spare					1	1	0	1
Spare					1	1	1	0
End of pulse signal					1	1	1	1
Address signal—second address								
Digit 0	0	0	0	0				
Digit 1	0	0	0	1				
Digit 2	0	0	1	0				
Digit 3	0	0	1	1				
Digit 4	0	1	0	0				
Digit 5	0	1	0	1				
Digit 6	0	1	1	0				
Digit 7	0	1	1	1				
Digit 8	1	0	0	0				
Digit 9	1	0	0	1				
Spare	1	0	1	0				
Code 11	1	0	1	1				
Code 12	1	1	0	0				
Spare	1	1	0	1				
Spare	1	1	1	0				
End of pulse signal	1	1	1	1				

The GAP identifies the type of address (dialed digits and so on) being presented in a call setup. It also indicates the numbering plan used in the address and the actual address. When LNP is provided, the GAP provides the actual dialed digits for a ported number. The called party address is then used for the LRN.

The GAP is also used in LNP applications. When a number has been ported, the dialed digits are placed in the GAP parameter. The called party address contains the location number used to route the call to the proper exchange. See Chapter 10 for more details on LNP.

The nature of address indicator is dependent on the type of address provided. All of the options are previously shown for clarity.

Generic Digits Parameter

Octet 1	**H**	**G**	**F**	**E**		**D**	**C**	**B**	**A**
Type of digits									
Account code				0		0	0	0	0
Authorization code				0		0	0	0	1
Private network traveling class mark				0		0	0	1	0
ANSI spare				0		0	0	1	1
					to				
				0		1	1	0	1
Originating party service provider				0		1	1	0	1
Bill to number				0		1	1	1	1
Reserved for network-specific use				1		0	0	0	0
					to				
				1		1	1	1	0
Reserved for extension				1		1	1	1	1
Encoding scheme									
BCD even	0	0	0						
BCD odd	0	0	1						
IA5	0	1	0						
Binary	0	1	1						
Spare	1	0	0						
		to							
	1	1	1						

Octets 2–n

Digits	(encoded in the format specified previously)

The generic digits parameter provides additional numeric data pertaining to supplementary services such as authorization code, the *personal identification number* (PIN), or account code.

The types of digits are found in the first octet. These types are related to PBX features and/or business group features. The transfer of this data is possible from an ISDN-compatible PBX to another ISDN-compatible PBX through the PSTN using the SS7 ISUP protocol and parameters such as this one.

Generic Name Parameter	**H**	**G**	**F**	**E**	**D**	**C**	**B**	**A**
Presentation								
Presentation allowed							0	0
Presentation restricted							0	1
Blocking toggle							1	0
No indication							1	1
Spare					0	0		
Availability								
Name available/unknown				0				
Name not available				1				
Type of name								
Spare	0	0	0					
Calling name	0	0	1					
Original called name	0	1	0					
Redirecting name	0	1	1					
Connected name	1	0	0					
Spare	1	0	1					
		to						
	1	1	1					

The generic name parameter contains information to be used for name display features. In ANSI networks, *calling name* (CNAM) display requires the terminating exchange to access a database and search for the name information. In some networks (outside the United States), the name information is actually carried in the IAM from the originating exchange. The name information is found in this parameter in the IA5 format. Up to 15 characters may be sent.

Hop Count Parameter	**H**	**G**	**F**	**E**	**D**	**C**	**B**	**A**
Hop counter				x	x	x	x	x
Spare	0	0	0					

The hop counter is used to ensure that ISUP looping does not occur. The initial message is sent in the forward direction with the maximum value allowed (network dependent). As the message is passed through each circuit, the counter is decremented by one. When the counter reaches zero, the message is discarded. The counter value is the number of contiguous SS7 circuits that this message must pass to reach its destination. This is provided in binary form.

Information Indicators

Octet 1	**H**	**G**	**F**	**E**	**D**	**C**	**B**	**A**
Calling party address response indicator								
Calling party address not included							0	0
Calling party address not available							0	1
Spare							1	0
Calling party address included, hold not provided							1	1

Hold-provided indicator								
Hold not provided (default)						0		
*Hold provided						1		
Spare				0	0			

	H	G	F	E	D	C	B	A
Calling party's category response indicator								
Calling party's category not included			0					
*Calling party's category included			1					
Charge information response indicator								
Charge information not included		0						
*Charge information included		1						
Solicited information indicator								
Solicited	0							
Unsolicited	1							

Octet 2	H	G	F	E	D	C	B	A
Spare		0	0	0	0	0	0	0
Multilocation business group info response indicator								
Multilocation business group info not included	0							
Multilocation business group info included	1							

This parameter provides additional information related to a call in progress. It can be sent in either direction and can be a solicited message or unsolicited. Solicited indicates that the distant exchange requested information (billing information, for example) and the receiving exchange is replying to that request.

This parameter does not provide the requested information; it simply indicates that the information is in this message.

Information Request Indicators

Octet 1	H	G	F	E	D	C	B	A
Calling party address request indicator								
Calling party address not requested								0
Calling party address requested								1

	H	G	F	E	D	C	B	A
Holding indicator								
Holding not requested							0	
*Holding requested							1	
Spare						0		

	H	G	F	E	D	C	B	A
Calling party's category request indicator								
Calling party's category not requested					0			
*Calling party's category requested					1			

	H	G	F	E	D	C	B	A
Charge information request indicator								
Charge information not requested				0				
Charge information requested				1				
Spare		0	0					

	H	G	F	E	D	C	B	A
Malicious call identification request indicator								
Malicious call identification not requested	0							
*Malicious call identification requested	1							

Octet 2	H	G	F	E	D	C	B	A
Spare		0	0	0	0	0	0	0
Multilocation business group info indicator								
Multilocation business group info not requested	0							
Multilocation business group info requested	1							

The information request parameter is used to request specific information regarding a call already in progress. The response to the request parameter is the information indicators, with the appropriate parameters providing the actual data.

This information is used in call processing and billing for the call. Not all of these procedures are currently in use in U.S. networks, yet the functionality is defined in both ITU and ANSI standards.

Jurisdiction Information Parameter (JIP)	H	G	F	E	D	C	B	A
Address signal—first address								
Digit 0					0	0	0	0
Digit 1					0	0	0	1
Digit 2					0	0	1	0
Digit 3					0	0	1	1
Digit 4					0	1	0	0
Digit 5					0	1	0	1
Digit 6					0	1	1	0
Digit 7					0	1	1	1
Digit 8					1	0	0	0
Digit 9					1	0	0	1
Spare					1	0	1	0
Code 11					1	0	1	1
Code 12					1	1	0	0
Spare					1	1	0	1
Spare					1	1	1	0
End of pulse signal					1	1	1	1
Address signal—second address								
Digit 0	0	0	0	0				
Digit 1	0	0	0	1				

Digit 2	0	0	1	0
Digit 3	0	0	1	1
Digit 4	0	1	0	0
Digit 5	0	1	0	1
Digit 6	0	1	1	0
Digit 7	0	1	1	1
Digit 8	1	0	0	0
Digit 9	1	0	0	1
Spare	1	0	1	0
Code 11	1	0	1	1
Code 12	1	1	0	0
Spare	1	1	0	1
Spare	1	1	1	0
End of pulse signal	1	1	1	1

The JIP is used in LNP applications to support billing systems. When an originating number has been ported, billing systems may not be able to determine the correct billing for the call (because 1 billion systems are still based on the North American Numbering Plan). The JIP provides the LRN assigned to the originating number, which is then used to determine proper billing for the call. The calling party number is no longer applicable for the billing of a call in the case of ported numbers.

This parameter is a variable-length parameter consisting of address signals only. The parameter provides numerical data indicating the geographic origination of the call.

Message Compatability Information	**H**	**G**	**F**	**E**	**D**	**C**	**B**	**A**
Instruction indicators								
Transit at intermediate exchange indicator								
Transit interpretation								0
End node interpretation								1
Release call indicator								
Do not release call							0	
Release call							1	
	H	**G**	**F**	**E**	**D**	**C**	**B**	**A**
Send notification indicator								
Do not send notification						0		
Send notification						1		
Discard message indicator								
Do not discard message (pass on)					0			
Discard message					1			
Pass on not possible indicator								
Release call				0				
Discard information				1				
Broadband/narrowband interworking indicator								
Pass on		0	0					
Discard message		0	1					
Release call		1	0					
Reserved, assume 00		1	1					

Extension indicator	
Information continues through the next octet	0
Last octet	1

This parameter provides instructions to an exchange processing an ISUP message that it does not understand. This message can be sent in either direction. Additional octets may be added to this parameter at another time, but right now these are the only defined parameter values.

Network-Specific Facility

This field is encoded according to rules set by the network operator.

Service-related information is transparently transferred in either direction between the local exchange and the identified network that provides the service. The information is significant to both the user and identified network.

Normal Call Indicator

Nature of Connection Indicators	**H**	**G**	**F**	**E**	**D**	**C**	**B**	**A**
Satellite indicator								
No satellite circuit in the connection							0	0
One satellite circuit in the connection							0	1
Two satellite circuits in the connection							1	0
Three or more satellite circuits in the connection							1	1
Continuity check indicator								
Continuity check not required					0	0		
Continuity check required on this circuit					0	1		
Continuity check performed on a previous circuit					1	0		
Spare					1	1		
	H	**G**	**F**	**E**	**D**	**C**	**B**	**A**
Echo control device indicator								
Outgoing half-echo control device not included				0				
Outgoing half-echo control device included				1				
Spare	0	0	0					

The nature of connection indicators is sent in the forward direction to provide information regarding the circuit connection specified in the CIC parameter of the message. The values within this parameter enable intermediate exchanges to determine how to handle the processing of this message.

Network Transport

As seen previously, this parameter consists of other ISUP parameters. It is used to send ISUP parameters through the network transparently, without involving a call setup or other mechanism. The objective is to send parameters end to end through the network, without the intermediate exchanges having to process the message.

Notification Indicator	**H**	**G**	**F**	**E**		**D**	**C**	**B**	**A**
Notification indicator									
Spare		0	0	0		0	0	0	0
					to				
		0	0	0		0	0	1	1
Call completion delay		0	0	0		0	1	0	0
Spare		0	0	0		0	1	0	1
					to				
		1	0	0		0	0	0	1
Conference established (multiparty call)		1	0	0		0	0	1	0
Conference disconnected		1	0	0		0	0	1	1
Other party added to conference		1	0	0		0	1	0	0
Isolated		1	0	0		0	1	0	1
Reattached		1	0	0		0	1	1	0
Other part isolated		1	0	0		0	1	1	1
Other party reattached		1	0	0		1	0	0	0
Other part split		1	0	0		1	0	0	1
Other part disconnected		1	0	0		1	0	1	0
Conference floating		1	0	0		1	0	1	1
Spare		1	0	0		1	1	0	0
Spare		1	0	0		1	1	0	1
Spare		1	0	0		1	1	1	0
Conference floating, served user preempted		1	0	0		1	1	1	1
Spare		1	0	1		0	0	0	0
					to				
		1	0	1		1	1	1	1
Call is a waiting call		1	1	0		0	0	0	0
Reserved for transfer in progress		1	1	0		0	0	0	1
Reserved for call isolated from conference call		1	1	0		0	0	1	0
Reserved for call split from conference call		1	1	0		0	0	1	1
Reserved for call reattached to conference call		1	1	0		0	1	0	0
Reserved for call added to conference call		1	1	0		0	1	0	1
Spare		1	1	0		0	1	1	0
					to				
		1	1	0		1	0	0	0
Call transfer, alerting		1	1	0		1	0	0	1
Call transfer, active		1	1	0		1	0	1	0
Spare		1	1	0		1	0	1	1
					to				
		1	1	1		1	0	0	0
Remote hold		1	1	1		1	0	0	1
Remote hold released		1	1	1		1	0	1	0
Call is forwarded/deflected		1	1	1		1	0	1	1
Spare		1	1	1		1	1	0	0
					to				
		1	1	1		1	1	1	0
Reserved		1	1	1		1	1	1	1

Extension indicator	
Octet continues through the next octet	0
Last octet	1

This parameter provides information regarding supplementary services. These services are related to business group services (such as Centrex) that provide PBX-like features to a group of business lines within a group. This parameter provides information regarding the nature of the calling party and its origin (forwarded call, call from hold, and so on).

Optional Backward Call Indicators	**H**	**G**	**F**	**E**	**D**	**C**	**B**	**A**
In-band information indicator								
No indication								0
In-band info or an appropriate pattern is now available								1
Call forwarding may occur indicator								
No indication							0	
*Call forwarding may occur							1	
Spare					0	0		
Reserved for national use					0	0		
Network excessive delay indicator								
No indication		0						
Network excessive delay encountered		1						
User-network interaction indicator								
No indication	0							
User-network interaction occurs, cut-through in both directions	1							

This parameter is sent in the backward direction providing information to another exchange regarding a call in progress. The in-band information indicator is used to alert the distant exchange that in-band information such as a recording or a service tone is present on the voice circuit.

The call forwarding may occur parameter is used to indicate that the call may be forwarded if the called party does not answer. There are currently no national ANSI procedures for this field.

The user-network interaction indicator is sent from the originating exchange to indicate that additional information is being gathered from the caller (such as a PIN number or special code) before routing the call.

Operator Services Information	**H**	**G**	**F**	**E**	**D**	**C**	**B**	**A**
Original access prefix	0	0	0	1				
Unknown					0	0	0	0
1 + or 011 +					0	0	0	1
0 + or 01 +					0	0	1	0
0 –					0	0	1	1
Spare					0	1	0	0
						to		
					1	0	0	1

	H	G	F	E	D	C	B	A
Reserved for national use					1	0	1	0
						to		
					1	1	1	1
Bill to information entry type and handling type	0	0	1	0				
Info entry unknown, unknown handling					0	0	0	0
Info entry manual by operator, station handling					0	0	0	1
Info entry manual by operator, person handling					0	0	1	0
Info entry automated by tone input, station handling					0	0	1	1
Info entry unknown, station handling					0	1	0	0
Info entry unknown, person handling					0	1	0	1
Info entry manual by operator, unknown handling					0	1	1	0
Info entry automated by tone input, unknown handling					0	1	1	1
Info entry automated by tone input, person handling					1	0	0	0
Info entry automated by spoken input, unknown handling					1	0	0	1
Info entry automated by spoken input, station handling					1	0	1	0
Info entry automated by spoken input, person handling					1	0	1	1
Spare					1	1	0	0
						to		
					1	1	0	1
Reserved for network-specific use					1	1	1	0
						to		
					1	1	1	1
Bill-to type	0	0	1	1				
Unknown					0	0	0	0
Calling card—14-digit format					0	0	0	1
Calling card—89C format					0	0	1	0
Calling card—other format					0	0	1	1
Collect					0	1	0	0
Third-party number billing					0	1	0	1
Sent paid (prepaid calling card)					0	1	1	0
Spare					0	1	1	1
						to		
					1	0	1	0
Reserved for network-specific use					1	1	1	1
Bill-to specific information	0	1	0	0				
Spare					0	0	0	0
NIDB authorizes					0	0	0	1
NIDB reports, verify by automated means					0	0	1	0
NIDB reports, verify by operator					0	0	1	1
No NIDB query					0	1	0	0
NIDB reports unavailable					0	1	1	0
No NIDB response—timeout					0	1	1	1

	H	G	F	E	D	C	B	A
No NIDB response—reject component					1	0	0	0
No NIDB response—ACG in effect					1	0	0	1
No NIDB response—SCCP failure					1	0	1	0
Spare					1	0	1	1
Reserved for network-specific use					1	1	0	0
						to		
					1	1	1	1
Special handling	0	1	0	1				
Unknown					0	0	0	0
Call completion					0	0	0	1
Rate information					0	0	1	0
Trouble reporting					0	0	1	1
Time and charges					0	1	0	0
Credit reporting					0	1	0	1
General assistance					0	1	1	0
Spare					0	1	1	1
						to		
					1	0	1	0
Reserved for network-specific use					1	0	1	1
						to		
					1	1	1	1
Spare	0	1	1	0				
Accessing signaling	0	1	1	1				
Unknown					0	0	0	0
Dial pulse					0	0	0	1
Dualtone multifrequency (DTMF)					0	0	1	0
Spare					0	0	1	1
						to		
					1	0	0	1
Reserved for network-specific use					1	0	1	0
						to		
					1	1	1	1

The operator services parameter is used to identify how a caller accessed an operator for assistance and how the call should be billed. For example, if a caller dialed 01 plus the number, special handling may be required. Billing may also be affected based on the caller's interaction with the operator (the caller may request the call to be billed to a calling card, for example). The values in this parameter are used to identify how the call should be handled in the billing system.

Original Called Number	H	G	F	E	D	C	B	A
Octet 1								
Nature of address indicator								
Spare		0	0	0	0	0	0	0
Unique subscriber number		0	0	0	0	0	0	1
Spare, reserved for national use		0	0	0	0	0	1	0
Unique national significant number		0	0	0	0	0	1	1
Unique international number		0	0	0	0	1	0	0

Spare		0	0	0		0	1	0	1
					to				
		1	1	1		0	0	0	0
Nonunique subscriber number		1	1	1		0	0	0	1
Spare, reserved for national use		1	1	1		0	0	1	0
Nonunique national significant number		1	1	1		0	0	1	1
Nonunique international number		1	1	1		0	1	0	0
Spare		1	1	1		0	1	0	1
Spare		1	1	1		0	1	1	0
Test line test code		1	1	1		0	1	1	1
Reserved for network-specific use		1	1	1		1	0	0	0
					to				
		1	1	1		1	1	1	0
Spare		1	1	1		1	1	1	1
Odd/even indicator									
Even number of address signals	0								
Odd number of address signals	1								
Octet 2	**H**	**G**	**F**	**E**		**D**	**C**	**B**	**A**
Reserved								0	0
Address presentation									
Presentation						0	0		
Presentation restricted (default)						0	1		
Spare						1	0		
Spare						1	1		
Numbering plan									
Unknown		0	0	0					
ISDN numbering plan (E.164, E.163)		0	0	1					
Spare		0	1	0					
Reserved (ITU: data numbering plan)		0	1	1					
Reserved (ITU: telex numbering plan)		1	0	0					
Private numbering plan		1	0	1					
Spare		1	1	0					
Spare		1	1	1					
Spare	0								
Octet 3	**H**	**G**	**F**	**E**		**D**	**C**	**B**	**A**
First address									
Digit 0						0	0	0	0
Digit 1						0	0	0	1
Digit 2						0	0	1	0
Digit 3						0	0	1	1
Digit 4						0	1	0	0
Digit 5						0	1	0	1
Digit 6						0	1	1	0
Digit 7						0	1	1	1
Digit 8						1	0	0	0
Digit 9						1	0	0	1
Spare						1	0	1	0

Code 11					1	0	1	1
Code 12					1	1	0	0
Spare					1	1	0	1
Spare					1	1	1	0
End of pulse signal					1	1	1	1
Second address								
Digit 0	0	0	0	0				
Digit 1	0	0	0	1				
Digit 2	0	0	1	0				
Digit 3	0	0	1	1				
Digit 4	0	1	0	0				
Digit 5	0	1	0	1				
Digit 6	0	1	1	0				
Digit 7	0	1	1	1				
Digit 8	1	0	0	0				
Digit 9	1	0	0	1				
Spare	1	0	1	0				
Code 11	1	0	1	1				
Code 12	1	1	0	0				
Spare	1	1	0	1				
Spare	1	1	1	0				
End of pulse signal	1	1	1	1				

This parameter is used when call redirecting (forwarding) occurs. The parameter identifies the address of the party that initiated the redirection. This parameter also provides information regarding the presentation of the calling party number, which is used by the end exchange when connecting the call to its destination.

Only the first and second address signals are shown here, but there can be additional address signals. The address signal is typically the telephone number assigned to the subscriber who initiated the redirecting.

Originating Line Information	**H**	**G**	**F**	**E**		**D**	**C**	**B**	**A**
Binary equivalent of the II digits	0	0	0	0		0	0	0	0
					to				
	0	1	1	0		0	0	1	1
Reserved for future expansion	0	1	1	0		0	1	0	0
					to				
	1	1	1	1		1	1	1	1

This information is sent in the forward direction representing a toll class of service for the call.

Outgoing Trunk Group Number

This parameter provides the trunk number used for an interworking call. *Interworking* means that the call was sent to another exchange in another network. The trunk number represents the circuit used at the gateway switch into the other network. This parameter is only found in instances where internetworking occurs.

Precedence

Octet 1	**H**	**G**	**F**	**E**	**D**	**C**	**B**	**A**
Precedence level								
Flash override (0)					0	0	0	0
Flash (1)					0	0	0	1
Immediate (2)					0	0	1	0
Priority (3)					0	0	1	1
Routine (4)					0	1	0	0
Spare					0	1	0	1
						to		
					1	1	1	1
Spare				0				
Look ahead for busy (LFB)								
Look ahead for busy allowed		0	0					
Look ahead for busy not allowed		1	0					
Path reserved		0	1					
Spare		1	1					
Extension bit								
Octet continues through the next octet	0							
Last octet	1							

Network identity (coded in BCD), the telephone country code is placed in the second to fourth digits of this parameter. The first digit is set to zero.

Octet 2	**H**	**G**	**F**	**E**	**D**	**C**	**B**	**A**
Multilevel Precedence and Preemption (MLPP)								
Defense switched network		0	0	0	0	0	0	0
Spare		0	0	0	0	0	0	1
					to			
		1	1	1	1	1	1	1
Extension								
Octet continues through the next octet	0							
Last octet	1							

Precedence is a feature provided in defense networks, where an individual of higher rank is given priority for outgoing trunks over someone of lower rank. This feature is typically found in AUTOVON systems, but is now being offered through the central office services.

This parameter identifies the level of precedence allowed and whether or not the *look ahead for busy* (LFB) feature is allowed.

Range and Status

Octet 2	**H**	**G**	**F**	**E**	**D**	**C**	**B**	**A**
Range								

This field of the parameter is a binary representation of the range of circuits that are affected by the status field that follows it. Because this is a zero-based number, the actual circuit number is the binary representation plus one.

In national circuits (ANSI only), the range is from 0 to 23. International circuits range from 0 to 31.

Octet 2
Status

The status field provides the status of the circuits indicated in the range parameter. The status bits are numbered from 0 to 23, or 0 to 31 in international circuits. There is a direct correlation between the range and status fields. The number of status bits is equal to the value of the range field plus 1.

Status bit 0 is located in the first bit position of the status octet. Other status bits follow in numerical order. More than one octet can exist. In the range field, if the range is coded as zero, the status bit is not provided.

The status bits also depend on the message type for their value. For instance, in a CGB message, the status bits are different than in a CGU message. The status bit values are as follows:

In circuit group blocking (CGB) messages:	
No blocking	0
Blocking	1
In circuit group blocking acknowledgment (CGBA) messages:	
No blocking acknowledgment	0
Blocking acknowledgment	1
In circuit group unblocking (CGU) messages:	
No unblocking	0
Unblocking	1
In circuit group unblocking acknowledgment (CGUA) messages:	
No unblocking acknowledgment	0
Unblocking acknowledgment	1
In circuit group reset acknowledgment (CGRA) messages:	
No blocking	0
Blocked	1

The range and status parameter is found in circuit group supervision messages to indicate the range of circuits that are affected by the status indicator and the status of those circuits. The status field correlates to the type of circuit group message. The status bits are provided in numerical order, with each status bit corresponding with the CIC of the affected circuit.

As seen in the range field, the range field identifies the actual circuit number (or range of circuits), which is used to also identify which status bits are required for each circuit. Status bit 0 is the first status bit, which is found in the first bit location of the first octet in the status field.

Redirecting Number

Octet 1	H	G	F	E	D	C	B	A
Nature of address indicator								
Spare		0	0	0	0	0	0	0
Subscriber number		0	0	0	0	0	0	1

	H	G	F	E		D	C	B	A
Spare, reserved for national use		0	0	0		0	0	1	0
National significant number		0	0	0		0	0	1	1
International number		0	0	0		0	1	0	0
Spare		0	0	0		0	1	0	1
					to				
		1	1	1		0	0	0	0
Subscriber number, operator requested		1	1	1		0	0	0	1
National number, operator requested		1	1	1		0	0	1	0
International number, operator requested		1	1	1		0	0	1	1
No number present, operator requested		1	1	1		0	1	0	0
No number present, cut-through call to carrier		1	1	1		0	1	0	1
950 1 call from local exchange carrier public station, hotel/motel, or nonexchange access end office		1	1	1		0	1	1	0
Test line code		1	1	1		0	1	1	1
Reserved for network-specific use		1	1	1		1	0	0	0
					to				
		1	1	1		1	1	1	0
Spare		1	1	1		1	1	1	1
Odd/even bits									
Even number of address signals	0								
Odd number of address signals	1								
Octet 2	**H**	**G**	**F**	**E**		**D**	**C**	**B**	**A**
Reserved								0	0
Address presentation									
Presentation allowed						0	0		
Presentation restricted						0	1		
Spare						1	0		
Spare						1	1		
Numbering plan									
Unknown numbering plan		0	0	0					
ISDN numbering plan (Rec. E.164, E.163)		0	0	1					
Spare		0	1	0					
Reserved ITU-TS data numbering plan		0	1	1					
Reserved ITU-TS telex numbering plan		1	0	0					
Private numbering plan		1	0	1					
Spare		1	1	0					
Spare		1	1	1					
Spare	0								
Octet 3	**H**	**G**	**F**	**E**		**D**	**C**	**B**	**A**
Address signal—first address									
Digit 0						0	0	0	0
Digit 1						0	0	0	1
Digit 2						0	0	1	0
Digit 3						0	0	1	1
Digit 4						0	1	0	0
Digit 5						0	1	0	1

	H	G	F	E	D	C	B	A
Digit 6					0	1	1	0
Digit 7					0	1	1	1
Digit 8					1	0	0	0
Digit 9					1	0	0	1
Spare					1	0	1	0
Code 11					1	0	1	1
Code 12					1	1	0	0
Spare					1	1	0	1
Spare					1	1	1	0
End of pulse signal					1	1	1	1
Address signal—second address								
Digit 0	0	0	0	0				
Digit 1	0	0	0	1				
Digit 2	0	0	1	0				
Digit 3	0	0	1	1				
Digit 4	0	1	0	0				
Digit 5	0	1	0	1				
Digit 6	0	1	1	0				
Digit 7	0	1	1	1				
Digit 8	1	0	0	0				
Digit 9	1	0	0	1				
Spare	1	0	1	0				
Code 11	1	0	1	1				
Code 12	1	1	0	0				
Spare	1	1	0	1				
Spare	1	1	1	0				
End of pulse signal	1	1	1	1				

When call forwarding is applied to a call, this parameter is used to indicate the telephone number from which the called number was last forwarded. Call forwarding can be invoked from any telephone, hence the need to identify from where the telephone number was last forwarded.

The address signals, as is the case in all parameters with this field, can consist of several digits, even though only one octet is shown here. At least four octets are needed to represent a telephone number. Any odd number of address signals requires a filler for the last half of the octet. The odd/even address indicator identifies those addresses that do not require a full octet and contain a filler at the end.

Redirect Capability	H	G	F	E		D	C	B	A
Redirection possibility indicator									
Not used							0	0	0
Redirection possible before ACM							0	0	1
Redirection possible before ANM							0	1	0
Redirection possible at any time during the call							0	1	1
Spare							1	0	0
					to				
							1	1	1
Spare		0	0	0		0			

	H	G	F	E	D	C	B	A
Extension indicator								
Next octet	0							
Last octet	1							

This information is sent in the forward direction. It indicates that the succeeding exchange is allowed to initiate the redirection of a call and indicates when the redirection may take place (in relation to the call-processing procedures).

Redirect Counter	**H**	**G**	**F**	**E**	**D**	**C**	**B**	**A**
Counter				x	x	x	x	x

This parameter indicates the number of times a call has been redirected within the network. This may be used to control the maximum number of times a number can be redirected as part of a service offering. The number is represented in binary.

Redirection Information	**H**	**G**	**F**	**E**	**D**	**C**	**B**	**A**
Octet 1								
Spare						0	0	0
Reserved					0			
Original redirecting reason								
Unknown/not available (default)	0	0	0	0				
User busy	0	0	0	1				
No reply	0	0	1	0				
Unconditional	0	0	1	1				
Deflection	0	1	0	0				
Spare	0	1	0	1				
		to						
	1	1	1	0				
Reserved	1	1	1	1				
Octet 2	**H**	**G**	**F**	**E**	**D**	**C**	**B**	**A**
Redirection counter								
No redirection has occurred					0	0	0	0
Redirected 1 time					0	0	0	1
Redirected 2 times					0	0	1	0
Redirected 3 times					0	0	1	1
Redirected 4 times					0	1	0	0
Redirected 5 times					0	1	0	1
Redirected 6 times					0	1	1	0
Redirected 7 times					0	1	1	1
Redirected 8 times					1	0	0	0
Redirected 9 times					1	0	0	1
Redirected 10 times					1	0	1	0
Redirected 11 times					1	0	1	1
Redirected 12 times					1	1	0	0
Redirected 13 times					1	1	0	1
Redirected 14 times					1	1	1	0
Redirected 15 times					1	1	1	1
Redirecting reason								
Unknown/not available (default)	0	0	0	0				

User busy	0	0	0	1
No reply	0	0	1	0
Unconditional	0	0	1	1
Spare	0	1	0	0
		to		
	1	1	1	1

This parameter is used with calls where call forwarding has been invoked. The information provided indicates the original reason for the forwarding and, in the case where the call has undergone more than one forward, the reason for the subsequent forwardings.

It is quite possible for a call to be forwarded a number of times without the caller's knowledge. For example, a telephone number may be forwarded to another telephone number. When a caller tries to reach that number, he or she is forwarded to the second number. The second number may also be forwarded, resulting in the call being redirected again. This parameter indicates the number of times the call was redirected and the reason for the redirection.

Redirection Number

This parameter identifies the number to which the called number is to be redirected. The values contained in this parameter include the nature of address indicator, the numbering plan, and the telephone number that the call will be redirected to. The actual values have not been shown here because they are repeated in several other places in this book.

Remote Operations	**H**	**G**	**F**	**E**		**D**	**C**	**B**	**A**
Protocol profile									
Spare				0		0	0	0	0
					to				
				1		0	0	0	0
Remote operations protocol				1		0	0	0	1
Spare				1		0	0	1	0
					to				
				1		1	1	1	1
Spare		0	0						
Extension bit									
Next octet	0								
Last octet	1								
Component (follows same format as TCAP component)									

Service Activation	**H**	**G**	**F**	**E**		**D**	**C**	**B**	**A**
Feature code indicators									
Reserved for international use	0	0	0	0		0	0	0	0
					to				
	0	1	1	1		1	0	1	1
Call waiting originating invoked	0	1	1	1		1	1	0	0
Dial call waiting invoked	0	1	1	1		1	1	0	1

Complete call req, ISUP used all the way	0	1	1	1		1	1	1	0
Complete call req, ISUP not used all the way	0	1	1	1		1	1	1	1
Network service attached	1	0	0	0		0	0	0	0
Network service released	1	0	0	0		0	0	0	1
Coin collect	1	0	0	0		0	0	1	0
Coin return	1	0	0	0		0	0	1	1
Network service recall	1	0	0	0		0	1	0	0
Billing verification	1	0	0	0		0	1	0	1
Hold available	1	0	0	0		0	1	1	0
Hold not available	1	0	0	0		0	1	1	1
Hold request	1	0	0	0		1	0	0	0
Hold acknowledge	1	0	0	0		1	0	0	1
Hold release request	1	0	0	0		1	0	1	0
Hold release acknowledge	1	0	0	0		1	0	1	1
Hold continuation request	1	0	0	0		1	1	0	0
Disconnect request	1	0	0	0		1	1	0	1
Reconnect request	1	0	0	0		1	1	1	0
Spare	1	0	0	0		1	1	1	1
					to				
	1	0	0	1		0	0	1	0
Resume operator services	1	0	0	1		0	0	1	1
Spare	1	0	0	1		0	1	0	0
					to				
	1	0	1	1		1	1	1	1
Network-specific use	1	1	0	0		0	0	0	0
					to				
	1	1	1	1		1	1	1	0
Reserved for network-specific use	1	1	1	1		1	1	1	1

This parameter is used to invoke supplementary services from another exchange. Currently, there are not many features called for in this parameter, but there is room for expansion.

Service Code

The service code field is a one-octet field representing the service code as assigned by the North American Numbering Plan Administration at Bellcore. Currently, this parameter is under further study, but it can be used to identify a specific type of service, which a subscriber can invoke either in real time or otherwise. The number in this parameter is a binary number representing the decimal equivalent of the service code.

Special Processing Request

	H	G	F	E	D	C	B	A
Special processing request								
Spare	0	0	0	0	0	0	0	0
Reserved for international use	0	0	0	0	0	0	0	1

					to				
	0	0	0	0		1	1	1	1
Reserved for national use	0	0	0	1		0	0	0	0
					to				
	0	1	1	1		1	1	1	0
Service processing requested	0	1	1	1		1	1	1	1
Reserved for network-specific use	1	0	0	0		0	0	0	0
					to				
	1	1	1	1		1	1	1	0
Spare	1	1	1	1		1	1	1	1

In the event that a call originates on a private network, a special number translation or authorization code verification may be necessary. This parameter indicates the special processing requirements of such a call. The receiver of this message is a service node in the PSTN from a service node in the private network.

Suspend/Resume Indicators

	H	G	F	E	D	C	B	A
Suspend/resume indicator								
ISDN subscriber initiated								0
Network initiated (default)								1
Spare	0	0	0	0	0	0	0	

The suspend/resume indicator is sent in the forward direction to indicate the originator of a suspend or resume. There are only two options—an ISDN subscriber or the network that initiated the message.

Transaction Request

This parameter follows the same message structure as seen in the TCAP, providing the transaction ID and the SCCP address for those messages used to carry information regarding a call in progress. This parameter can only be used for calls that are already in progress and enables the ISUP protocol to use the services of the TCAP protocol to deliver service information relating to a call.

This parameter is carried in the IAM during the circuit connection establishment. The receiving exchange then uses this parameter for all subsequent messages related to the call on that circuit, such as feature invocation or call hand-off procedures.

Transit Network Selection

Octet 1	H	G	F	E	D	C	B	A
Network identification plan (national ANSI networks)								
Unknown					0	0	0	0
Three-digit carrier identification with circuit code					0	0	0	1
Four-digit carrier identification with circuit code					0	0	1	0
Reserved					0	0	1	1
						to		

					0	1	1	1
Reserved for network-specific use					1	0	0	0
						to		
					1	1	1	1
Network identification plan (international networks)								
Unknown					0	0	0	0
Public data network identification code					0	0	1	1
Public land mobile network (PLMN) ID code					0	1	1	0
Type of network identification								
ITU standardized identification		0	0	0				
National network identification		0	1	0				
Spare	0							
Octet 2	**H**	**G**	**F**	**E**	**D**	**C**	**B**	**A**
Digit one								
Digit 0					0	0	0	0
Digit 1					0	0	0	1
Digit 2					0	0	1	0
Digit 3					0	0	1	1
Digit 4					0	1	0	0
Digit 5					0	1	0	1
Digit 6					0	1	1	0
Digit 7					0	1	1	1
Digit 8					1	0	0	0
Digit 9					1	0	0	1
Spare					1	0	1	0
Code 11					1	0	1	1
Code 12					1	1	0	0
Spare					1	1	0	1
Spare					1	1	1	0
End of pulse signal					1	1	1	1
Digit two								
Digit 0	0	0	0	0				
Digit 1	0	0	0	1				
Digit 2	0	0	1	0				
Digit 3	0	0	1	1				
Digit 4	0	1	0	0				
Digit 5	0	1	0	1				
Digit 6	0	1	1	0				
Digit 7	0	1	1	1				
Digit 8	1	0	0	0				
Digit 9	1	0	0	1				
Spare	1	0	1	0				
Code 11	1	0	1	1				
Code 12	1	1	0	0				
Spare	1	1	0	1				
Spare	1	1	1	0				
End of pulse signal	1	1	1	1				

Octet 3	H	G	F	E	D	C	B	A
Digit three								
Digit 0					0	0	0	0
Digit 1					0	0	0	1
Digit 2					0	0	1	0
Digit 3					0	0	1	1
Digit 4					0	1	0	0
Digit 5					0	1	0	1
Digit 6					0	1	1	0
Digit 7					0	1	1	1
Digit 8					1	0	0	0
Digit 9					1	0	0	1
Spare					1	0	1	0
Code 11					1	0	1	1
Code 12					1	1	0	0
Spare					1	1	0	1
Spare					1	1	1	0
End of pulse signal					1	1	1	1
Circuit code								
Unspecified	0	0	0	0				
International call, no operator requested	0	0	0	1				
International call, operator requested	0	0	1	0				
Spare	0	0	1	1				
			to					
	0	1	1	1				
Reserved for network-specific use	1	0	0	0				
			to					
	1	1	1	1				

This parameter is sent in the forward direction to indicate the long-distance carrier or transit network to be used to carry this call. This is used whenever the call is an inter-LATA call or international call.

The carrier ID is a three- or four-digit code administered by Bellcore that uniquely identifies each of the long-distance carriers. This is the same code dialed when using calling cards (10xxx, where xxx equals the carrier code).

Currently, there are no implications allowing for the use of ISUP to international networks such as the public data network or the *public land mobile network* (PLMN). These are for further study.

Transmission Medium Used

Octet 1	H	G	F	E	D	C	B	A
Transmission medium used								
Speech	0	0	0	0	0	0	0	0
Spare	0	0	0	0	0	0	0	1
Reserved for 64 kbps unrestricted	0	0	0	0	0	0	1	0
3.1 kHz audio	0	0	0	0	0	0	1	1
Reserved	0	0	0	0	0	1	0	0

Reserved	0	0	0	0		0	1	0	1
Reserved for 64 kbps preferred	0	0	0	0		0	1	1	0
Reserved	0	0	0	0		0	1	1	1
					to				
	0	0	0	0		1	0	1	0
Spare	0	0	0	0		1	0	1	1
					to				
	1	1	1	1		1	1	1	1

The transmission used in a call setup is sent in the backward direction in the event that the original circuit requested could not be used. This parameter then identifies the circuit type and is carried in the ANM, ACM, and CPG messages.

User Service Information

Octet 1	**H**	**G**	**F**	**E**	**D**	**C**	**B**	**A**
Information transfer capability								
Speech				0	0	0	0	0
Unrestricted digital information				0	1	0	0	0
Restricted digital information				0	1	0	0	1
3.1 kHz audio				1	0	0	0	0
7 kHz audio				1	0	0	0	1
Coding standard								
ITU standardized coding		0	0					
National standard		1	0					
Extension bit								
Octet continues through the next octet	0							
Last octet	1							

*Note: Only permitted when the 64 kbps information transfer rate is used.

Octet 2	**H**	**G**	**F**	**E**	**D**	**C**	**B**	**A**
Information transfer rate								
Code for packet mode calls				0	0	0	0	0
64 kbps				1	0	0	0	0
384 kbps				1	0	0	1	1
1,472 kbps (national ANSI only)				1	0	1	0	0
1,536 kbps				1	0	1	0	1
1,920 kbps				1	0	1	1	1
Multirate (64-kbps based)				1	1	0	0	0
Transfer mode								
Circuit mode		0	0					
Packet mode		1	0					
Extension bit								
Octet continues through the next octet	0							
Last octet	1							

Octet 2a	**H**	**G**	**F**	**E**		**D**	**C**	**B**	**A**
Establishment									
Demand (default)								0	0
Configuration									
Point-to-point (default)						0	0		
Structure									
Default (see description)		0	0	0					
8 kHz integrity		0	0	1					
Service data unit integrity		1	0	0					
Unstructured		1	1	1					
Extension bit									
Octet continues through the next octet	0								
Last octet	1								

Octet 2b	**H**	**G**	**F**	**E**		**D**	**C**	**B**	**A**
Information transfer rate (destination to origination)									
Code for packet mode calls				0		0	0	0	0
64 kbps				1		0	0	0	0
384 kbps				1		0	0	1	1
1,472 kbps				1		0	1	0	0
1,536 kbps				1		0	1	0	1
1,920 kbps				1		0	1	1	1
Multirate (64 kbps-based)				1		1	0	0	0
Symmetry									
Bidirectional symmetric (default)		0	0						
Extension bit									
Octet continues through the next octet	0								
Last octet	1								

Octet 2.1	**H**	**G**	**F**	**E**		**D**	**C**	**B**	**A**
(present if octet 2 indicates multirate 64 kbps base rate)									
Rate multiplier									
Reserved (0)		0	0	0		0	0	0	0
1 to 30		0	0	0		0	0	0	1
					to				
		0	0	1		1	1	1	0
31 to 127		0	0	1		1	1	1	1
					to				
		1	1	1		1	1	1	1
Extension bit									
Octet continues through the next octet	0								
Last octet	1								

Octet 3	**H**	**G**	**F**	**E**		**D**	**C**	**B**	**A**
User information layer 1 protocol									
ITU standardized rate adaption V.110/X.30				0		0	0	0	1
Recommendation G.711 µ-law speech				0		0	0	1	0
Recommendation G.722 and G.725 7 kHz audio				0		0	1	0	1

Non-ITU standardized rate adaption				0	0	1	1	1
ITU standardized rate adaption V.120				0	1	0	0	0
ITU standardized rate adaption X.31 HDLC flag stuffing				0	1	0	0	1
Layer 1 identification		0	1					
Extension bit								
Octet continues through the next octet	0							
Last octet	1							

Octet 3a	**H**	**G**	**F**	**E**	**D**	**C**	**B**	**A**
(present if ITU standardized rate adaption V110/V120)								
User rate				x	x	x	x	x
Rate is indicated by E-bits specified in Rec. I.460				0	0	0	0	0
0.6 kbps Recommendations V.6 & X.1				0	0	0	0	1
1.2 kbps Recommendations V.6				0	0	0	1	0
2.4 kbps Recommendations V.6 & X.1				0	0	0	1	1
3.6 kbps Recommendations V.6				0	0	1	0	0
4.8 kbps Recommendations V.6 & X.1				0	0	1	0	1
7.2 kbps Recommendations V.6				0	0	1	1	0
8.0 kbps Recommendations I.460				0	0	1	1	1
9.6 kbps Recommendations V.6 & X.1				0	1	0	0	0
14.4 kbps Recommendations V.6				0	1	0	0	1
16.0 kbps Recommendations I.460				0	1	0	1	0
19.2 kbps Recommendations V.6				0	1	0	1	1
32.0 kbps Recommendations I.460				0	1	1	0	0
48.0 kbps Recommendations V.6 & X.1				0	1	1	1	0
56.0 kbps Recommendations V.6				0	1	1	1	1
64.0 kbps Recommendations X.1				1	0	0	0	0
0.1345 kbps Recommendations X.1				1	0	1	0	1
0.100 kbps Recommendations X.1				1	0	1	1	0
0.075/1.2 kbps Recommendations V.6 & X.1				1	0	1	1	1
1.2/0.075 kbps Recommendations V.6 & X.1				1	1	0	0	0
0.050 kbps Recommendations V.6 & X.1				1	1	0	0	1
0.075 kbps Recommendations V.6 & X.1				1	1	0	1	0
0.110 kbps Recommendations V.6 & X.1				1	1	0	1	1
0.150 kbps Recommendations V.6 & X.1				1	1	1	0	0
0.200 kbps Recommendations V.6 & X.1				1	1	1	0	1
0.300 kbps Recommendations V.6 & X.1				1	1	1	1	0
12 kbps Recommendations V.6				1	1	1	1	1
Negotiation								
In-band negotiation not possible			0					
In-band negotiation possible			1					
Synchronous/asynchronous								
Synchronous at the R interface		0						
Asynchronous at the R interface		1						
Extension bit								
Octet continues through the next octet	0							
Last octet	1							

Octet 3b	**H**	**G**	**F**	**E**	**D**	**C**	**B**	**A**
(present if ITU standardized rate adaption V110)								
Spare								0
Flow control on receive								
Cannot accept data with flow control mechanism							0	
Can accept data with flow control mechanism							1	
Flow control on transmit								
Not required to send data with flow control mechanism						0		
Required to send data with flow control mechanism						1		
Network-independent clock on receive								
Cannot accept data with independent clock					0			
Can accept data with independent clock					1			
Network-independent clock on transmit								
Not required to send data with network-independent clock				0				
Required to send data with network-independent clock				1				
Intermediate rate								
Not used		0	0					
8 kbps		0	1					
16 kbps		1	0					
32 kbps		1	1					
Extension bit								
Octet continues through the next octet	0							
Last octet	1							

Octet 3c	**H**	**G**	**F**	**E**	**D**	**C**	**B**	**A**
(present if ITU standardized rate adaption V.120)								
Spare								0
In-band/out-band								
Not applicable to this standard							0	
Negotiation is done in-band using logical link zero							1	
Assignor/assignee								
Message originator is "Default Assignee"						0		
Message originator is "Assignor Only"						1		
Logical link identifier (LLI) negotiation								
Default LLI = 256					0			
LLI negotiation					1			
Mode of operation								
Bit-transparent mode of operation				0				
Protocol-sensitive mode of operation				1				
Multiframe								
Multiframe establishment not supported			0					
Multiframe establishment supported			1					
Rate adaption header/no header								
Rate adaption header not included		0						
Rate adaption header included		1						

Extension bit								
Octet continues through the next octet	0							
Last octet	1							
Octet 3d	**H**	**G**	**F**	**E**	**D**	**C**	**B**	**A**
(present if ITU standardized rate adaption V110/V120)								
Parity						x	x	x
Odd						0	0	0
Even						0	1	0
None						0	1	1
Forced to 0						1	0	0
Forced to 1						1	0	1
Number of data bits excluding parity bit								
Not used				0	0			
5 data bits				0	1			
7 data bits				1	0			
8 data bits				1	1			
Number of stop bits								
Not used		0	0					
1 stop bit		0	1					
1.5 stop bits		1	0					
2 stop bits		1	1					
Extension bit								
Octet continues through the next octet	0							
Last octet	1							
Octet 3e	**H**	**G**	**F**	**E**	**D**	**C**	**B**	**A**
(present if ITU standardized rate adaption V110/V120)								
Modem type	x	x	x	x	x	x		
Coded according to network-specific rules								
Duplex mode								
Half-duplex		0						
Full-duplex		1						
Extension bit								
Octet continues through the next octet	0							
Last octet	1							
Octet 4	**H**	**G**	**F**	**E**	**D**	**C**	**B**	**A**
User information (layer 2 protocol)								
ANSI T1.602				0	0	0	1	0
Recommendation X.25 link level				0	0	1	1	0
Layer 2 identifier		1	0					
Extension bit								
Octet continues through the next octet	0							
Last octet	1							

Octet 5	**H**	**G**	**F**	**E**	**D**	**C**	**B**	**A**
User information (layer 3 protocol)								
ANSI T1.607				0	0	0	1	0
Recommendation X.25 packet layer				0	0	1	1	0
Layer 3 identifier		1	1					
Extension bit								
Octet continues through the next octet	0							
Last octet	1							

The user service information parameter is used when the subscriber is requesting data transmission on the voice facility without use of a modem. In this case, the subscriber is using either ISDN or X.25 packet switching as an interface to the PSTN.

The purpose of this parameter is to send the data transmission parameters to the distant exchange so that the called party can receive the same parameters and establish the proper connection for receipt of the data. This parameter does not address the requirements of ATM or BISDN. These are addressed through another protocol—BISUP.

Many notes and prerequisites are listed in the ANSI standard regarding this parameter. Not all of these are listed here. The use of several of the octets is dependent on the use of other octets. For example, octet 3b is only used when octet 3 indicates ITU standardized rate adaptation V.110/X.30. Many of the notes are provided, but the best source for this parameter is the ANSI publication of T1.113-1992.

BISUP Message Types

BISUP has been developed to support the use of ATM voice circuits on the PSTN. The ISUP protocol was developed to support the connection and teardown of digital channelized circuits, such as DS1 and DS3. The ISUP protocol provides the data necessary to connect and control these channels in digital circuits, but ATM is based on virtual paths and virtual circuits, not on channels.

The BISUP protocol is based on ISUP and uses the same signaling procedures when possible. It is important to understand the role of BISUP so as not to confuse its procedures with those used when ATM SS7 links are used. The BISUP protocol is intended for the setup and teardown of voice transmissions over ATM, not the management of SS7 over ATM links in the same node.

Rather than duplicate the efforts from the previous section, this section will serve to identify the message types and their parameters used for BISUP signaling within the SS7 network. These message types and their parameters are published as they are currently defined and could change as the standards continue to evolve.

Figure 9.61 reflects the message types that are defined for use in the broadband networks. They are defined in the previous section for the most part with the exception of those that are new. The new message types are defined next.

Message Type	HGFE	DCBA
Address Complete	0 0 0 0	0 1 1 0
Answer	0 0 0 0	1 0 0 1
Blocking	0 0 0 1	0 0 1 1
Blocking Acknowledgment	0 0 0 1	0 1 0 1
Call Progress	0 0 1 0	1 1 0 0
Confusion	0 0 1 0	1 1 1 1
Consistency Check End	0 0 0 1	0 1 1 1
Consistency Check End Ack	0 0 0 1	1 0 0 0
Consistency Check Request	0 0 0 0	0 1 0 1
Consistency Check Request Ack	0 0 0 1	0 0 0 1
Forward Transfer	0 0 0 0	1 0 0 0
IAM Acknowledgment	0 0 0 0	1 0 1 0
IAM Reject	0 0 0 0	1 0 1 1
Initial Address Message	0 0 0 0	0 0 0 1
Network Resource Management	0 0 1 1	0 0 1 0
Release	0 0 0 0	1 1 0 0
Release Complete	0 0 0 1	0 0 0 0
Reset	0 0 0 1	0 0 1 0
Reset Acknowledgment	0 0 0 0	1 1 1 1
Resume	0 0 0 0	1 1 1 0
Segmentation	0 0 1 1	1 0 0 0
Subsequent Address	0 0 0 0	0 0 1 0
Suspend	0 0 0 0	1 1 0 1
Unblocking	0 0 0 1	0 1 0 0
Unblocking Acknowledgment	0 0 0 1	0 1 1 0
User Part Available	0 0 1 1	0 1 0 1
User Part Test	0 0 1 1	0 1 0 0
User-to-user Information	0 0 1 0	1 1 0 1
Reserved for Narrowband ISDN	0 0 0 0	0 0 1 1
	to	
	0 0 1 1	0 1 1 1
Reserved for Code Extension	1 1 1 1	1 1 1 1

Figure 9.61 BISUP requires some new message types in addition to those already supported in ISUP. This figure shows all of the BISUP message types and their indicator values.

In addition to some changes in message types, some additional parameters have been added to support ATM and BISDN circuits. These parameters are illustrated with their respective message types and described in the following section. The values of the various parameters are defined in the Bellcore and ANSI standards, as well as the ITU-TS Recommendation Q.2931.

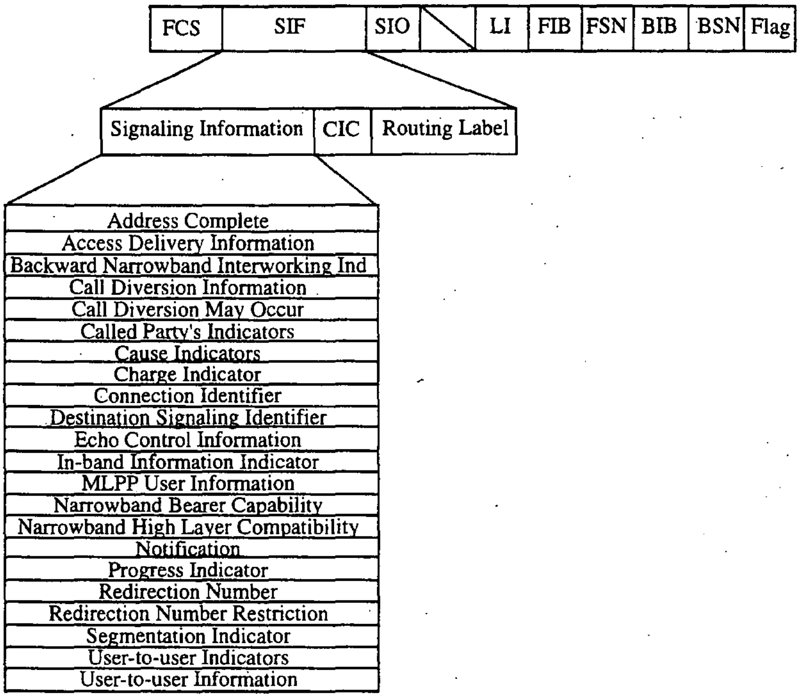


Figure 9.62

Address Complete (ACM)

The ACM has added several new parameters to support the VPCIs and additional broadband parameters. These can be seen in Figure 9.62. The function has not changed, however.

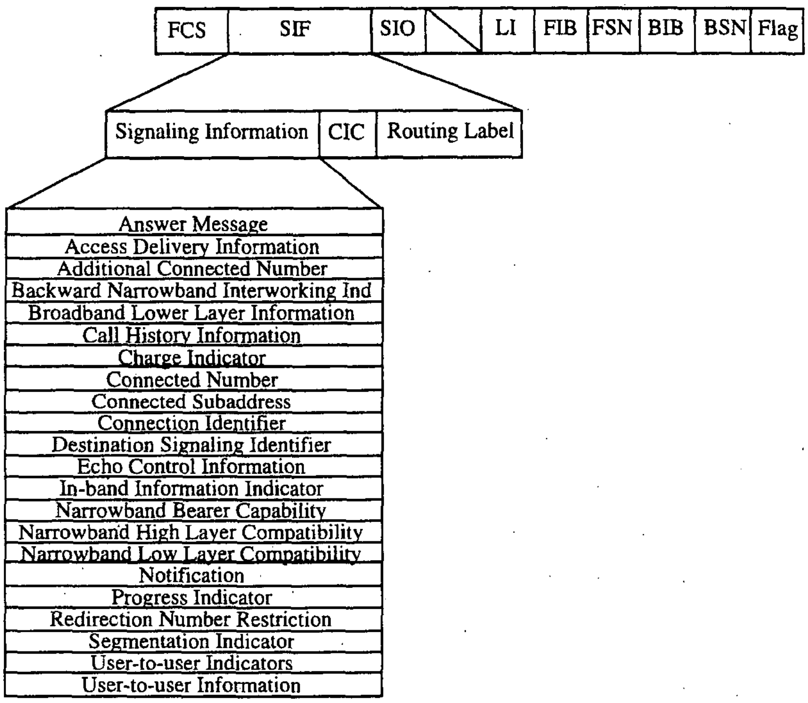

FCS
SIF
SIO
LI
FIB
FSN
BIB
BSN
Flag
Signaling Information
CIC
Routing Label
Answer Message
Access Delivery Information
Additional Connected Number
Backward Narrowband Interworking Ind
Broadband Lower Layer Information
Call History Information
Charge Indicator
Connected Number
Connected Subaddress
Connection Identifier
Destination Signaling Identifier
Echo Control Information
In-band Information Indicator
Narrowband Bearer Capability
Narrowband High Layer Compatibility
Narrowband Low Layer Compatibility
Notification
Progress Indicator
Redirection Number Restriction
Segmentation Indicator
User-to-user Indicators
User-to-user Information

Figure 9.63

Answer (ANM)

The ANM also remains the same in function. The cut-through on a voice circuit (VPCI in the case of broadband) does not take place in both directions in ANSI networks until the called party goes off-hook, and the ANM has been received. Tones are passed over the voice circuit in one direction only, meaning the cut-through takes place in the backward direction.

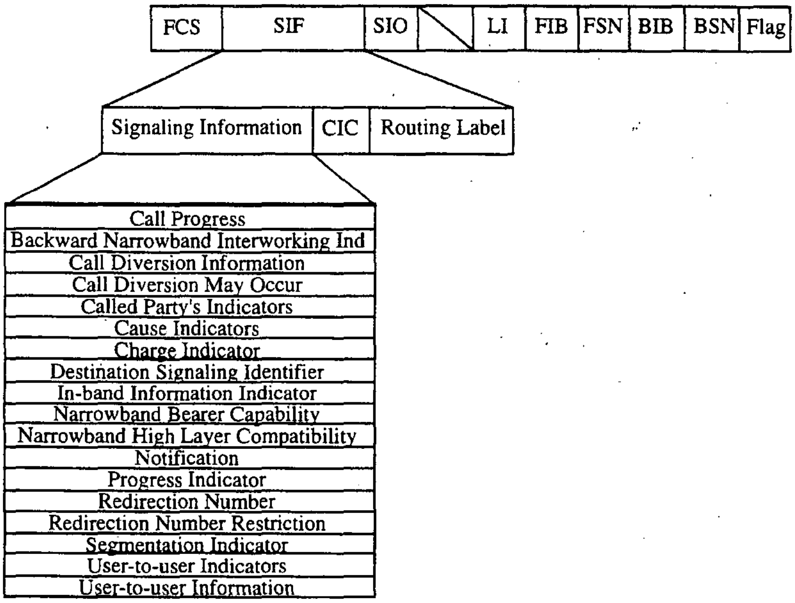


Figure 9.64

Call Progress (CPG)

This message enables additional information regarding the status of a call to be sent to a remote exchange. Used only in broadband, this enables exchanges to send event information in either direction while a call is in progress.

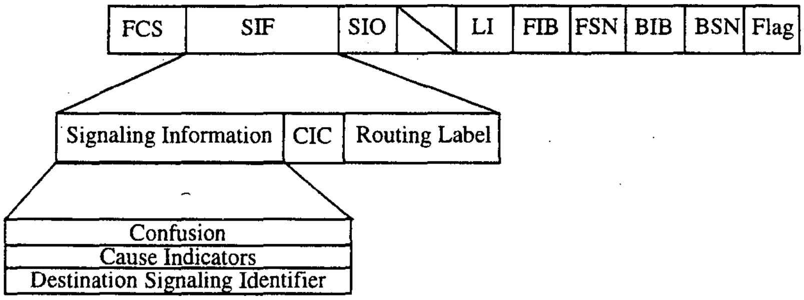
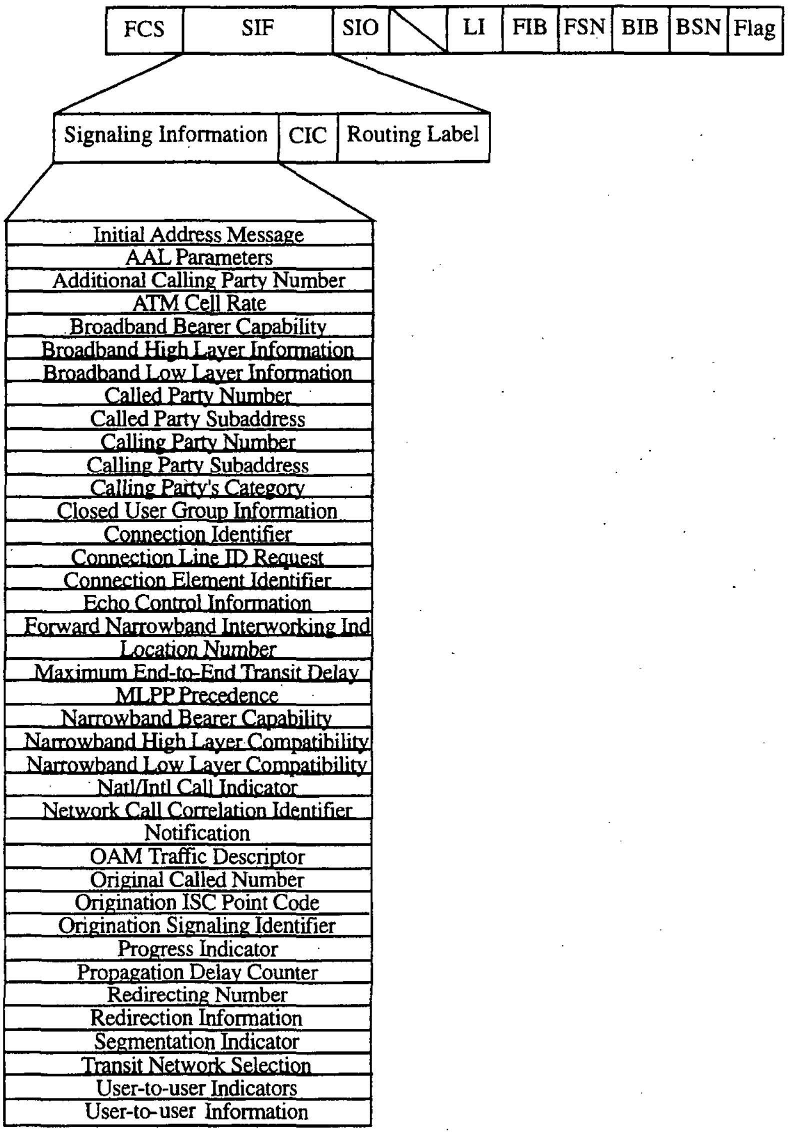


Figure 9.65

Confusion (CFN)

This message is the same as in normal ISUP. When a message is received that the exchange cannot identify, it returns the CFN message type.



Initial Address Message
AAL Parameters
Additional Calling Party Number
ATM Cell Rate
Broadband Bearer Capability
Broadband High Layer Information
Broadband Low Layer Information
Called Party Number
Called Party Subaddress
Calling Party Number
Calling Party Subaddress
Calling Party's Category
Closed User Group Information
Connection Identifier
Connection Line ID Request
Connection Element Identifier
Echo Control Information
Forward Narrowband Interworking Ind
Location Number
Maximum End-to-End Transit Delay
MLPP Precedence
Narrowband Bearer Capability
Narrowband High Layer Compatibility
Narrowband Low Layer Compatibility
Natl/Intl Call Indicator
Network Call Correlation Identifier
Notification
OAM Traffic Descriptor
Original Called Number
Origination ISC Point Code
Origination Signaling Identifier
Progress Indicator
Propagation Delay Counter
Redirecting Number
Redirection Information
Segmentation Indicator
Transit Network Selection
User-to-user Indicators
User-to-user Information

Figure 9.66

Exit (EXM)

This message is virtually the same as in ISUP. It is used when interworking with other networks to indicate that a message has been successfully passed to another network.

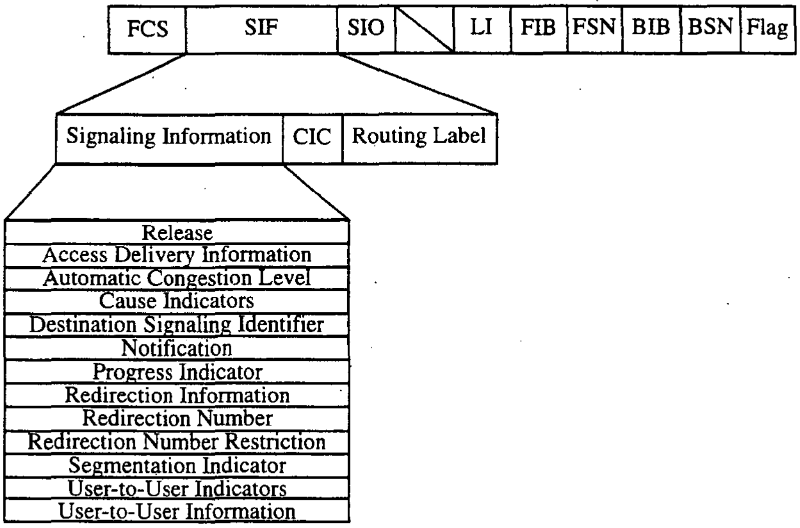


Figure 9.67

Initial Address Message (IAM)

This is also the same as in normal ISUP. The main difference between the two protocols and their IAMs is in the procedures. Normally, the routing label would identify which trunk circuit is to be used. In broadband, it identifies the VPCI, if there is one available at the originating exchange that will accommodate the call being requested. Each exchange (originating and destination) is responsible for half of the virtual path connections, which prevents the possibility of glare.

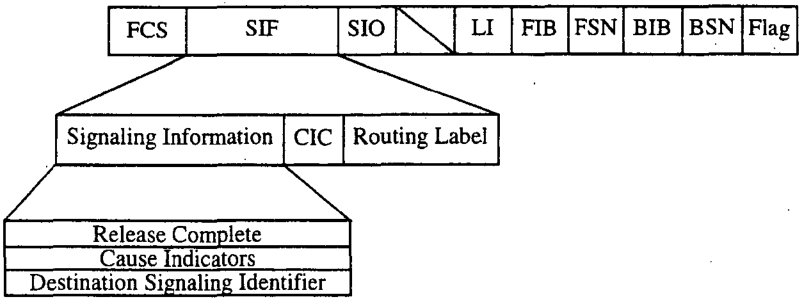


Figure 9.68

Release (REL)

The REL has several new parameters. All pertain to identifying the circuit that is to be released.

Release Complete (RLC)

The RLC remains the same and does not change with the exception of the destination signaling identifier.

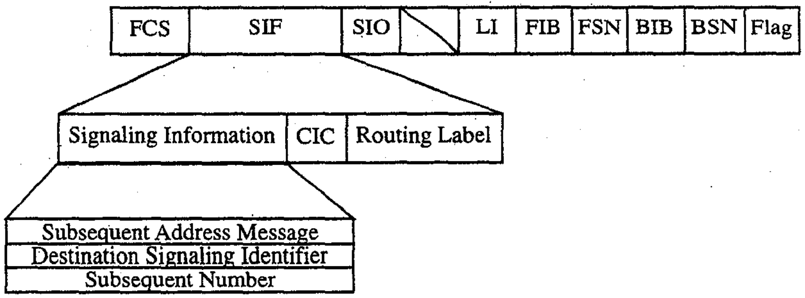


Figure 9.69

Subsequent Address (SAM)

This message type is used in international networks only and provides additional addressing information. This does not apply to U.S. networks.

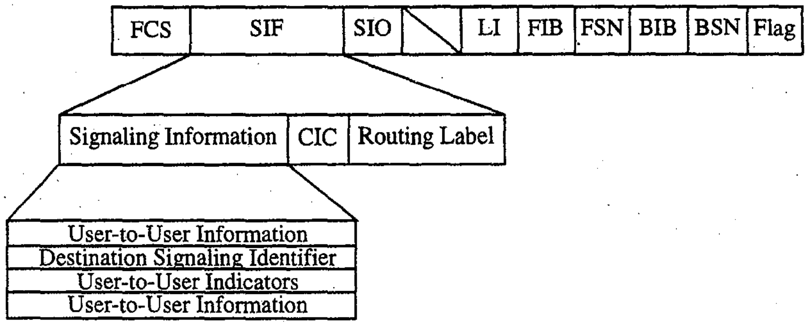


Figure 9.70

User-to-User Information (USIS)

There are no procedures currently defined for this parameter in U.S. networks. The purpose of this message is to provide additional information to a user part from a remote user part in relation to a call already in progress. The term "user-to-user" refers to two user parts sending information to one another.

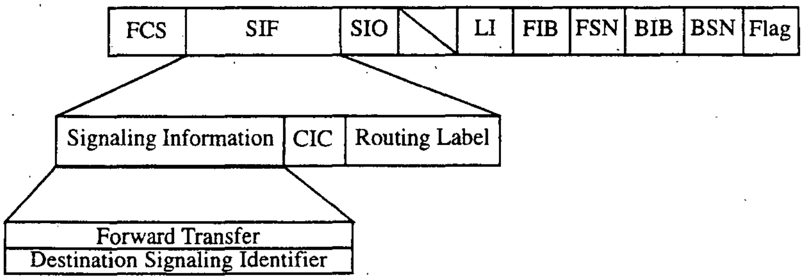


Figure 9.71

Forward Transfer (FOT)

This message type is used when an operator is requested during direct dial to an international number. The operator is also recalled when the call is terminated to provide additional assistance. Currently, this is only used in international networks.

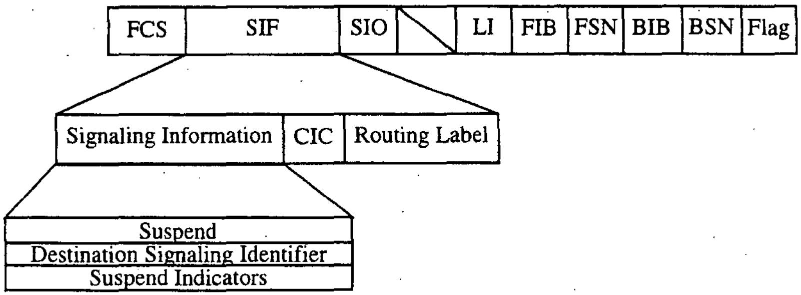


Figure 9.72

Suspend (SUS)

This message is sent in either direction to indicate that the called party has been disconnected. This would be used only if the connection had first been established and then the called party was disconnected. Rather than send a REL, the SUS enables the call circuit to be maintained for a period of time before initiating the release procedure. For example, if a called party is using call waiting, they may issue a flash hook to answer the other party. The SUS would be sent to the first calling exchange to hold the call circuit until a timeout or until the called party came back to the original call.

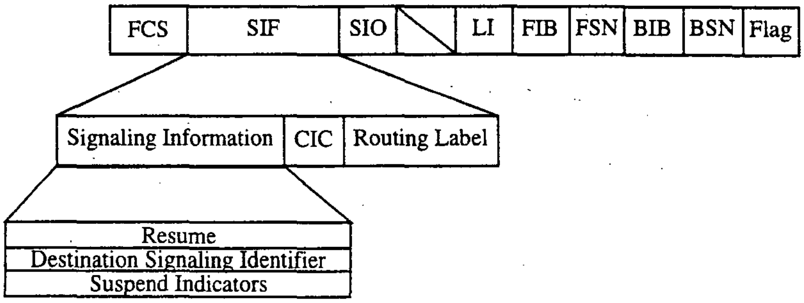


Figure 9.73

Resume (RES)

This is used along with the SUS type. When the called party comes back to the call pre· viously suspended, RES is sent to indicate that the call progress can continue aı normal.

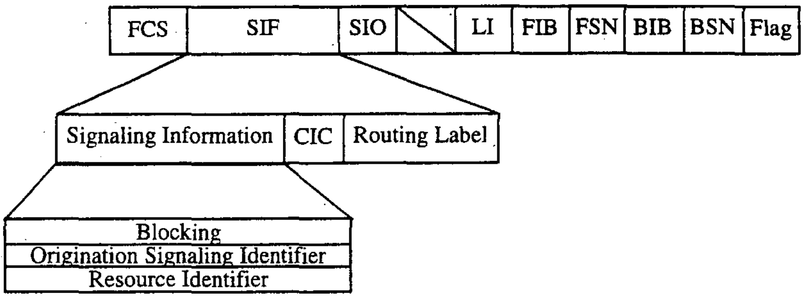


Figure 9.74

Blocking (BLO)

This is a maintenance message used for blocking a particular circuit from a connection It can be initiated manually or automatically by network management. It enables a cir· cuit to be removed from service while still being able to send traffic over the circuit Maintenance personnel may choose to block a circuit while they send test message: over that circuit.

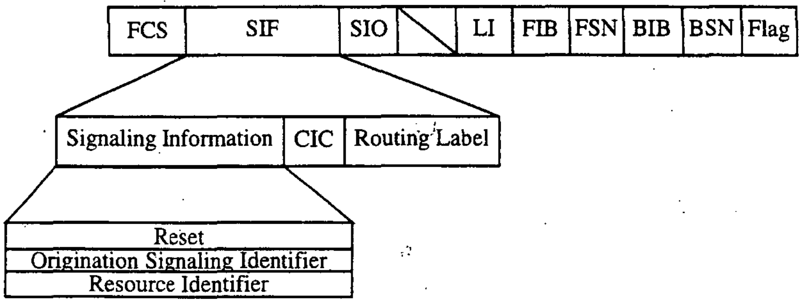


Figure 9.75

Reset (RSM)

This is used when memory allocated to a specific VPCI gets corrupted and can no longer remember the state of the connection. The reset is used to start both ends in the same known state. All resources are released and counters are reset, yet the connection is maintained.

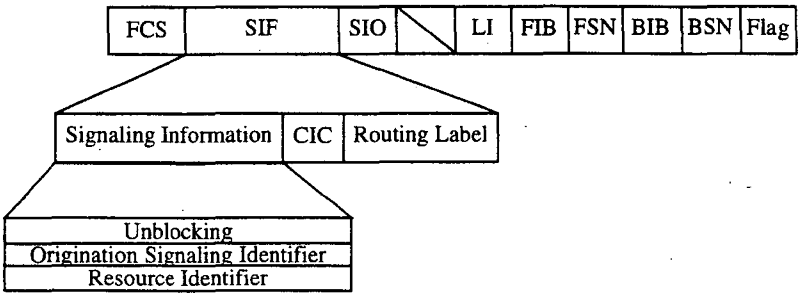


Figure 9.76

Unblocking (UBL)

This is the opposite of the blocking message and is used to unblock a circuit.

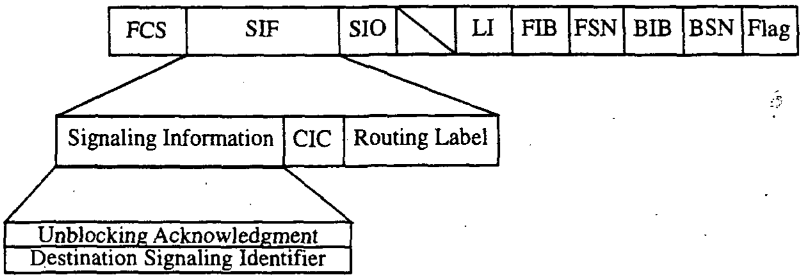


Figure 9.77

Blocking Acknowledgment (BLA)

This message type is sent in acknowledgment to a blocking message. It indicates that the blocking message was received and the user parts have been blocked from using the indicated circuit.

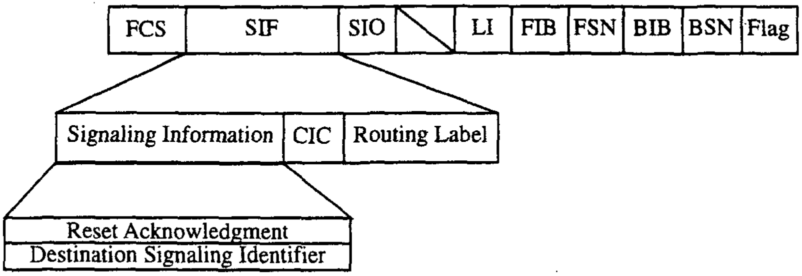


Figure 9.78

Reset Acknowledgment Message (RAM)

When memory at either end of a connection becomes corrupted, the signaling point may lose track of the state of a connection. In this case, a reset is requested on the specified VPCI. This reset will release all resources associated with the connection.

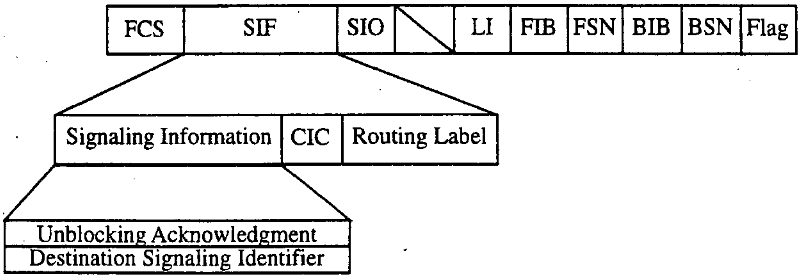


Figure 9.79

Unblocking Acknowledgment (UBA)

This message indicates receipt of an unblocking message. This acknowledges receipt of the unblocking message and confirms that the circuit identified has been unblocked from the user parts.

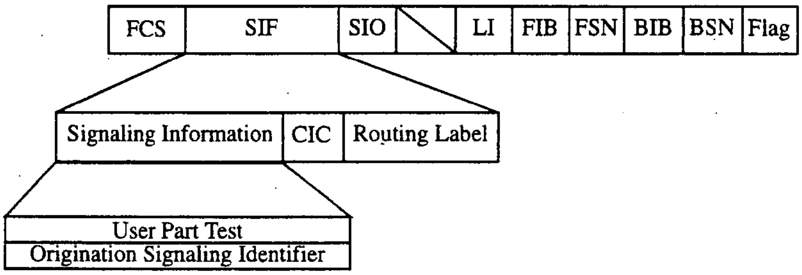


Figure 9.80

User Part Test (UPT)

This test message is used to verify the status of the specified user part. The purpose is to ensure that a user part marked as available or prohibited at the sending exchange is an accurate representation of the true state of the user part at the destination signaling point.

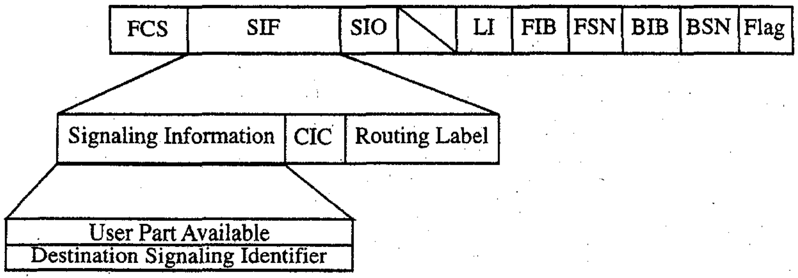


Figure 9.81

User Part Available (UPA)

This message is sent in response to a UPT message, indicating that the specified user part is available. UPT messages are used to verify the status of a given user part.

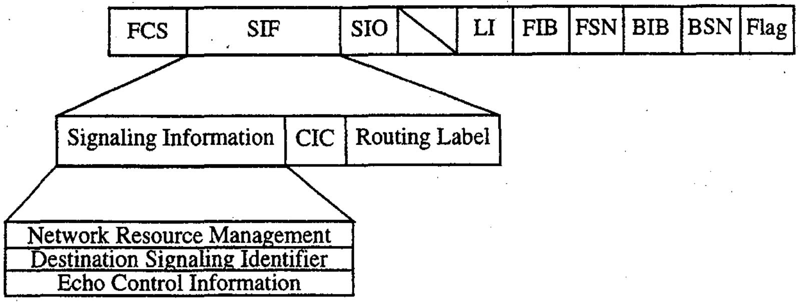


Figure 9.82

Network Resource Management (NRM)

This message is sent in the backward direction whenever the resources allocated to an established call need to be modified (that is, additional resources are allocated for additional bandwidth requirements).

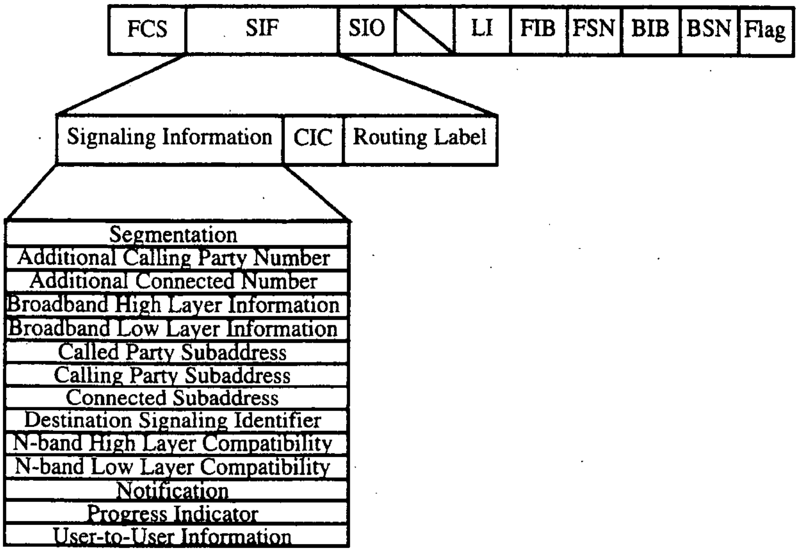

FCS
SIF
SIO
LI
FIB
FSN
BIB
BSN
Flag
Signaling Information
CIC
Routing Label
Segmentation
Additional Calling Party Number
Additional Connected Number
Broadband High Layer Information
Broadband Low Layer Information
Called Party Subaddress
Calling Party Subaddress
Connected Subaddress
Destination Signaling Identifier
N-band High Layer Compatibility
N-band Low Layer Compatibility
Notification
Progress Indicator
User-to-User Information

Figure 9.83

Segmentation Message (SGM)

When messages exceed the maximum size of the SS7 packet (272 octets), the message must be segmented. This message type is used to send an additional segment to the destination signaling point.

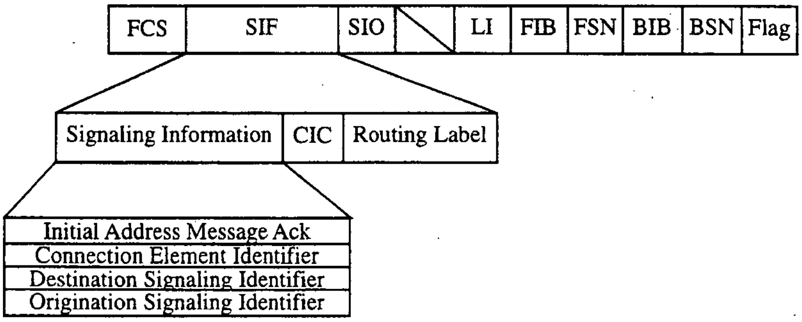

FCS
SIF
SIO
LI
FIB
FSN
BIB
BSN
Flag
Signaling Information
CIC
Routing Label
Initial Address Message Ack
Connection Element Identifier
Destination Signaling Identifier
Origination Signaling Identifier

Figure 9.84

IAM Acknowledgment (IAA)

Unlike the normal ISUP procedures, BISUP requires an acknowledgment to an IAM The purpose is twofold. In the event that the requesting exchange has assigned th VPCI, then the acknowledgment is used to confirm that the connection has been re served at the remote exchange. However, if the originating exchange is unable to assig a VPCI (because there is not ample bandwidth available within the range of circuit controlled by the originating exchange), then the acknowledgment is used to notify th originating exchange, which the VPCI has been assigned by the remote exchange.

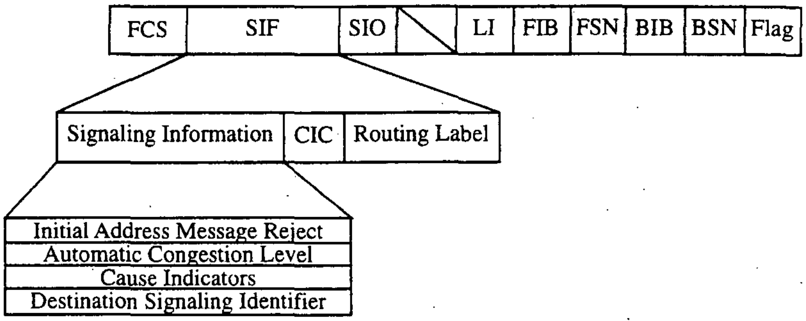

FCS
SIF
SIO
LI
FIB
FSN
BIB
BSN
Flag
Signaling Information
CIC
Routing Label
Initial Address Message Reject
Automatic Congestion Level
Cause Indicators
Destination Signaling Identifier

Figure 9.85

IAM Reject (IAR)

This is used to notify the originator of an IAM that the requested bandwidth is no available on any of the VPCIs within the control of the remote exchange. Therefore, connection cannot be established. Since the originating exchange has not assigned th VPCI, it is assumed that the originating exchange does not have ample bandwidt within its range of circuits either. The call connection attempt is aborted.

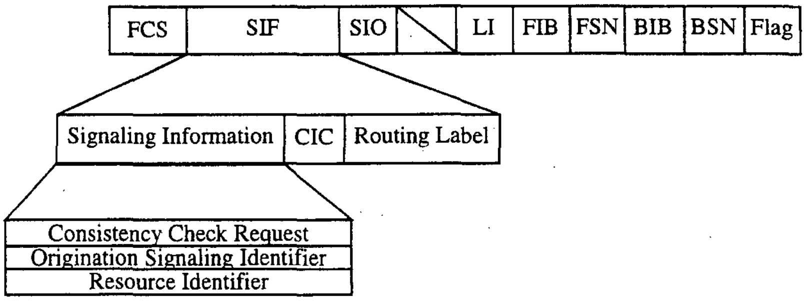


Figure 9.86

Consistency Check Request (CCR)

The CCR is sent to request the beginning of a continuity test on a specified VPCI within a virtual path. This is much like the continuity check used in the ISUP between two exchanges. The purpose is to ensure that a virtual connection can be established on the user plane between two exchanges. The test can be initiated by either exchange.

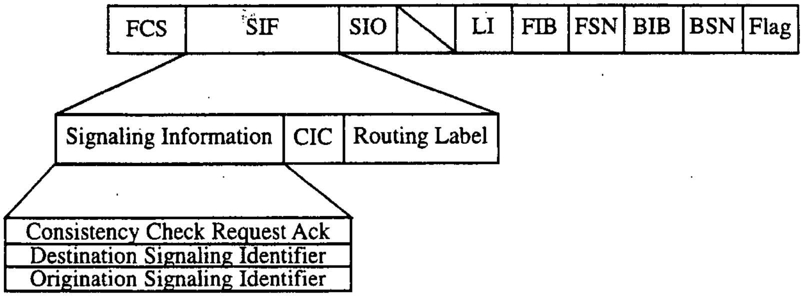

FCS
SIF
SIO
LI
FIB
FSN
BIB
BSN
Flag
Signaling Information
CIC
Routing Label
Consistency Check Request Ack
Destination Signaling Identifier
Origination Signaling Identifier

Figure 9.87

Consistency Check Request Acknowledgment (CCRA)

This is sent in response to the CCR and establishes the connections necessary to perform the test. Once the acknowledgment has been received, a test pattern can be sent over the proposed VPCI.

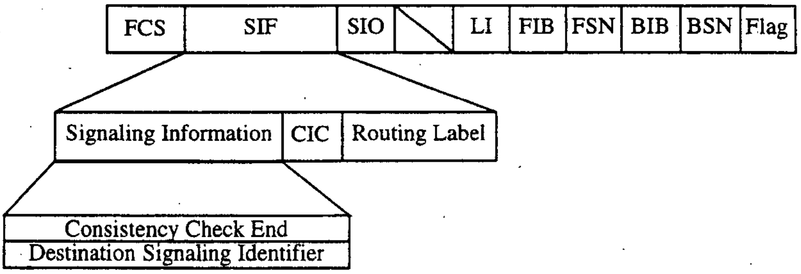


Figure 9.88

Consistency Check End (CCE)

This parameter indicates the end of a consistency check. It can only be sent when the consistency check has been completed. The *consistency check end* (CCE) is treated as an acknowledgment that the test has been completed.

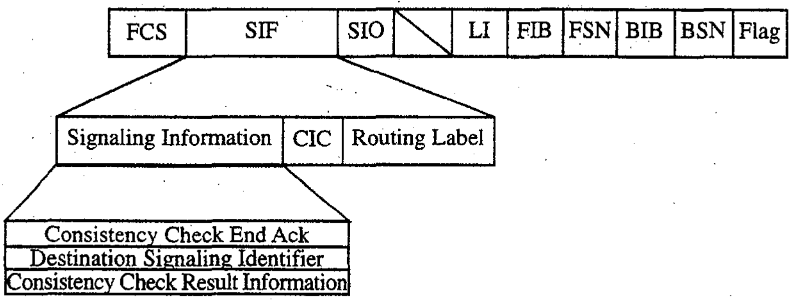

FCS
SIF
SIO
LI
FIB
FSN
BIB
BSN
Flag
Signaling Information
CIC
Routing Label
Consistency Check End Ack
Destination Signaling Identifier
Consistency Check Result Information

Figure 9.89

Consistency Check End Acknowledgment (CCEA)

This is sent in response to a CCE message. It indicates the end of a test. The acknowledgment cannot be sent until the test flow has been stopped.

Broadband Parameters

The following is a description of the parameters that have been identified for usage in BISUP. The association of these parameters to specific message types is shown in the previous section.

Access Delivery Information. This information indicates that a setup message (ISDN protocol) was generated by the destination. Only 1 bit in this octet is used; the rest is for future definition. This parameter is defined in the ITU-TS standards but not in the ANSI standards.

Additional Calling Party Number. This parameter provides the additional address information required for certain supplementary services. Typically, another exchange or another network entity will be providing the supplementary service, requiring additional calling party address information from the other entity. The same format as the calling party address is used. This parameter is defined in the ITU-TS standards but not in the ANSI standards.

Additional Connected Number. This parameter is sent in the backward direction and is used in the same manner as the additional calling party number. The difference is that this parameter is associated with an additional *connected* party number. The same format as the additional calling party number is used. This parameter is defined in the ITU-TS standards but not in the ANSI standards.

ATM Adaptation Layer (AAL) Parameter. This parameter is used to indicate which adaptation services will be required for the call. This information is of use to the local exchanges, who will be providing the services, as well as the subscriber, who will be providing the endpoint connection. The originating subscriber will be providing this information based on the type of data that they will be sending and depending on the type of source network being used at the originating subscriber's premise. Adaptation enables protocols such as Ethernet and Token Ring to interface to BISDN or ATM circuits seamlessly and transparently. The header information is stripped and encapsulated into an ATM header and then transported across the network to its destination, where it can then be reassembled to the destination network.

ATM Cell Rate. This information is used by the receiving exchange in setting up the ATM circuit. The receiving exchange must know how many cells per second are needed for this particular call to be successful. The originator will specify the transmission rate through the BISDN interface using the BISDN setup message, which maps to the SS7 IAM.

Automatic Congestion Level. When an exchange reaches a specified level of congestion, it will use this parameter to indicate which level of congestion it has reached to the opposite exchange. The message is not propagated through the network and is addressed to the adjacent exchange with which a connection has been established.

Backward Narrowband Interworking Indicators. This parameter identifies whether or not ISDN is used as the interface by the subscriber and whether or not ISUP is available end to end within the SS7 network.

Broadband Bearer Capability. This parameter is used to indicate to the adjacent exchange how much bandwidth will be needed on the subscriber interface (BISDN interface) to accommodate the call. This applies to the bearer channel only and has no meaning to the signaling network.

Broadband Low-Layer Information. The parameters within this parameter are defined in the Q.2931 ITU-TS standards. This is used to ensure compatibility at the lower layers of the protocol stack at the subscriber interface. They are sent to the distant exchange to ensure that the distant exchange is compliant.

Broadband High-Layer Information. Like the low-layer information parameter, this parameter and its contents are defined by ITU-TS Q.2931. The purpose is to ensure compatibility at the higher layers of the protocol stack at the subscriber interface at the distant end.

Call Diversion Information. When a call attempt is unsuccessful, this parameter is used to determine what type of treatment to provide for an attempted call. In some cases, an announcement may be requested, whereas in others, a busy tone or some other service tone may be used. This parameter is defined in the ITU-TS standards but not in the ANSI standards.

Call Diversion May Occur. This parameter is used to indicate that a call diversion may be indicated in a later message. This is necessary when interworking with *narrowband ISDN* (NISDN) networks. This parameter is defined in the ITU-TS standards but not in the ANSI standards.

Call History Information. The purpose of this parameter is to provide a means for advising the distant exchange as to how long a call took to reach its destination. This is accomplished by providing a time value, sending it to the distant exchange, and the distant exchange comparing the time to its own real-time clock. The difference is considered propagation delay, which is sent to the originating exchange for processing. This parameter is defined in the ITU-TS standards but not in the ANSI standards.

Called Party's Indicators. This is used to identify the type of called party, such as a normal subscriber or pay phone, and is used to determine if special treatment should be given to the call (for example, in the case of a pay phone).

Called Party Number. This parameter provides the called party number just as it does in normal ISUP. The called party number now includes a screening parameter, which is used to indicate whether or not the called party number will be presented or screened from view. The telephone company is allowed access to the number for routing purposes, but if the number is to be screened, they cannot provide the number to the subscriber. This is used with certain services (such as 800 services) where the called party may not know what number the calling party dialed. The number would then be displayed to them, enabling them to determine how the call should be answered.

Called Party Subaddress. This information is also defined in ITU-TS Q.2931. This parameter carries the information defined in that standard through the SS7 network for delivery to the distant exchange.

Calling Party Number. This is sent in the forward direction during a call setup procedure to identify the calling party. The presentation parameter determines whether or not the calling party number may be displayed to the called party. If not, the calling party number cannot be transferred to the BISDN interface.

Calling Party's Category. This parameter identifies the origin of the call, that is, a pay phone, a data terminal, or an ordinary subscriber. The parameter also provides operators with a language indicator to indicate which language they should use when answering the call. Spanish, Russian, French, English, and German are currently defined within the protocol. Other languages would have to be defined by the agencies using the network and be agreed upon mutually. The protocol provides additional codes to support network-defined languages.

Calling Party's Subaddress. This is the same as the called party's subaddress, but is sent in the opposite direction.

Carrier Identification Code. This parameter is used to identify the carrier that a subscriber has selected. It is sent in the forward direction during call setup.

Carrier Selection Information. This identifies how the carrier selection code was selected: by the subscriber dialing digits or through preselection.

Cause Indicator. This is used when a call has failed or has been cleared for any reason.

Charge Indicator. This is sent in the backward direction to indicate to the originating exchange whether or not a call is chargeable or not chargeable based on the destination and the called party number.

Charge Number. It provides the number to be billed for a call and is sent only in the forward direction.

Closed User Group Information. Currently, there are no procedures defined in ANSI or Bellcore networks for this parameter. A closed user group treats a group of numbers as if they were PBX extensions, much like the way Centrex treats a group of numbers as belonging to a business group. The user group can then be assigned specific features and privileges, enabling them to call within the group, but possibly blocking them from outside access.

Connected Line Identity Request Indicator. This is sent in the forward direction to indicate a request for the identity of the connected number rather than the called number (such as in the case of a forwarded call). Currently, there are no procedures defined in ANSI or Bellcore networks.

Connected Number. Although there are no procedures defined in U.S. networks for this parameter, its intended use is to identify the number to which a forwarded call was actually connected.

Connected Subaddress. This parameter is used along with the connected number and provides subaddress information according to the ITU-TS Recommendation I.330. As mentioned in the connected number description, these two parameters are used to identify which number a call was eventually terminated to, such as in the case of a forwarded or a transferred call.

Connection Element Identifier. This information is sent in the forward direction to identify the ATM virtual connection. In the event that the originating exchange does not have any virtual connections available within its control, this parameter would be sent by the destination exchange indicating which virtual circuit it has assigned for the requested call.

Consistency Check Request Information. This information is used to indicate the result of the consistency check. The consistency check is like the continuity check used in normal ISUP procedures, but because of the nature of broadband, continuity tests can no longer apply. The consistency check is used instead and is capable of testing the assignment of the virtual connection as well as the capability to transmit data through that virtual connection.

Destination Signaling Identifier. This parameter is used to associate the signaling connection with a virtual connection. This will most likely be used when BISUP is used in a fully associated signaling configuration.

Echo Control Information. As in the normal ISUP procedures, this parameter indicates whether or not echo control is required for half of the circuit or all of it.

Egress Service. This parameter provides information about the network of the terminating exchange.

Forward Narrowband Interworking Indicator. When interworking with an NISDN interface, this parameter provides information in the forward direction regarding the signaling capabilities on the connection.

Generic Address. This is used in supplementary services and can identify a destination number (dialed number).

Generic Digits. These digits can be account codes, authorization codes, or any type of number used in supplementary services.

Generic Name. This parameter provides specific name-related information used in supplementary services.

In-band Information Indicator. This parameter is used to indicate that in-band signaling information or an appropriate pattern is available on the connection specified.

Jurisdiction Information. This is the same as the JIP parameter used in ISUP. It provides the LRN of the originating party to be used by billing systems when the calling party number has been ported to another service provider.

Location Number. This parameter provides the same information as the called or calling party number, but is associated with a user within an NISDN connection. This is only used when interworking with NISDN networks.

Maximum End-to-End Transit Delay. This parameter indicates the maximum transit delay allowed for a message traveling through the network. The maximum delay is for an end-to-end transmission.

MLPP Precedence. Used primarily with military networks, this parameter indicates the level of precedence supported for the connection. The MLPP supplementary service is used in military installations to enable officers of higher rank to seize a trunk in use based on priority. This feature used to be limited to large AUTOVON systems installed on military bases, but is now offered through telephone service providers.

MLPP User Information. This is a one-octet parameter that uses only 1 bit in the entire octet. That 1 bit is used to indicate whether or not the called party is an MLPP user.

Narrowband Bearer Capability. During the setup phase of a connection, the originator may request a specific bandwidth. In some cases, the originator may provide a lower bandwidth for the connection only if the requested bandwidth is not available. This is referred to as the *fallback* bandwidth. This parameter is used to indicate the fallback bandwidth that has been allocated for the call connection.

Narrowband High-Layer Compatibility. This parameter is used to ensure compatibility between two exchanges. When an exchange requests service to a specific exchange, it may request a specific level of service or provide fallback to an available service. In the event the requested service is not available and the fallback service is assigned, this parameter notifies the distant exchange of the fallback service assignment so that it may set up its end to be compatible.

Narrowband Lower-Layer Compatibility. This parameter is like the previous parameter, but is concerned with compatibility at the lower layers rather than the upper layers. This also ensures compatibility at both ends of the exchange.

National/International Call Indicator. This is used to indicate to a national exchange that the origin of the call is from a national or an international network. The call-handling procedures for an international call are somewhat different than they are for a national call.

Notification. This parameter may be sent in either direction and is used to notify the other exchange of supplementary services such as call diversion procedures. Call diversion procedures encompass the use of service tones and announcements if a call cannot be completed (such as in the case of a wireless subscriber being away from his or her car and the wireless phone being turned off).

Notification Indicator. This is used with supplementary services to send notification to the user.

OAM Traffic Descriptor. This parameter indicates the cell rate that is required by *Operations, Administration, and Maintenance* (OAM) traffic on a specified virtual connection.

Original Called Number. When a call has been redirected, whether through forwarding or a transfer, this parameter identifies the original called party number. The connected called number will also be provided through that parameter. This is used when interworking with NISDN connections.

Originating Line Information. This information provides the toll class of service for a call. It is sent only in the forward direction.

Origination ISC Point Code. This information is provided in an IAM to indicate the originating point code of an international ISC.

Origination Signaling Identifier. This is sent by the originator of a signaling or control message to identify the association with the distant signaling connection.

Originating Facility Identifier. This information is only used when intranetworking and identifies the outgoing facility selected to reach the adjacent network.

Parameter Compatibility Information Parameter. This information is used to inform the receiver how to interact in the event that it receives an incorrect parameter it cannot recognize.

Progress Indicator. This is used to indicate an event that has taken place during the lifetime of a call. This is defined in Q.9231.

Propagation Delay Counter. While transferring through the network, this parameter is accumulating timer information. Each time it passes through a network entity, the timer information is updated in 1 ms increments. By the time it reaches its final destination, the destination exchange can determine the amount of propagation delay experienced by the message.

Redirecting Number. When a call is forwarded to another number, this parameter identifies the number that forwarded the call. For example, a number may be forwarded to another number, which in turn has diverted the call to yet another number. The number that provided the diversion would be indicated in this parameter.

Redirection Information. This parameter is also related to call diverting, or call forwarding. When a call is redirected to another number, this parameter indicates the nature of the redirection. For example, if a mobile subscriber is not available, then the call is diverted to an announcement. This parameter would then provide information as to why the call was diverted and the nature of the call.

Redirection Number. When a call is being diverted to another number, this parameter identifies the number to which the call has been diverted. This parameter is sent in the backward direction.

Redirection Number Restriction. Also related to diverted calls, this parameter identifies whether or not the diverted user allows the presentation of the telephone number. Because of legislature in many states, telephone companies must provide the option of screening the calling party number identification to the called party. This parameter indicates whether or not that is the case.

Resource Identifier. When a resource has been blocked or unblocked (such as an announcement), this parameter indicates which resource it applies to.

Signaling Identifier. The signaling identifier enables each exchange to assign a signaling association independently of one another and correlate messages with each signaling association. A signaling association is received over a signaling identifier.

Segmentation Indication. When a message is segmented into several messages, this parameter is used to indicate that the message is segmented and that there is additional segmentation information.

Special Processing Request. This parameter is sent only in the forward direction when special processing is required at the terminating exchange. Special processing includes everything from verifying authorization codes to translating private network numbers.

Subsequent Number. When the called party requires additional information, this parameter is used to provide the additional numbers or subsequent addresses.

Suspend/Resume Indicators. This parameter is sent in either the SUS or RES message to indicate whether or not the SUS or RES message was initiated by an ISDN subscriber or by the network. The SUS is used to temporarily place a circuit into a hold state, such as when the subscriber invokes another feature such as call waiting by performing a hook flash. The RES is used to indicate that the circuit is now reconnected and transmission can resume.

Transit Network Selection. When an IAM is sent, this parameter indicates which transit network, if any, is to be used to carry the message. A transit network is one that is used to access another carrier's network when network boundaries must be crossed.

User-Network Interaction Indicator. This parameter is sent in the backward direction when additional information is needed from the calling party to process a call.

User-to-User Indicators. This parameter carries the information defined in ITU-TS Q.2931 when the users at each end of the connection have information to share. The indicators are used to indicate that such information exists, whereas the following information parameter provides the actual information. The indicators also specify whether or not a reply is to be sent.

User-to-User Information. This parameter is used to send information from a user to the user at the remote exchange. The information is passed transparently through all intermediate exchanges.

Chapter

10

Local Number Portability (LNP)

By far the biggest project to ever hit the telephone industry has been *Local Number Portability* (LNP). Never before has anything this big been attempted in such a short time frame. There are a lot of questions and issues surrounding LNP, as well as a lot of uncertainties.

LNP was defined in the Telecommunications Act of 1996 as the "ability of users of telecommunications services to retain, at the same location, existing telecommunications numbers without impairment of quality, reliability, or convenience when switching from one telecommunications carrier to another."

The Telecommunications Act mandated that all telecommunications service providers provide, to the extent technically feasible, number portability in accordance with the requirements prescribed by the Commission. LNP got little attention until the FCC issued a mandate in June of 1996, requiring the implementation of LNP according to a very aggressive schedule. Rather than cite the specifics of the FCC mandate (FCC Docket 95-116), I will leave it to the reader to review the mandate itself, which is being published in three different phases. However, it should be understood that LNP affects everyone involved in wireline and wireless industries.

The following are some highlights from the FCC docket:

- The solution must support existing services and features.
- LNP must use the existing numbering resources efficiently.
- LNP cannot require subscribers to change their telephone numbers.
- There can be no unreasonable degradation in service (such as call setup delays) or network reliability degradation when subscribers switch carriers.
- No carrier can have a proprietary interest.
- The LNP solution must be able to accommodate location and service portability in the future.
- There can be no significant adverse impact outside areas where number portability is deployed.

The intent of LNP is to open up local telephone service to competition. The authors of the Telecommunications Act feel that the biggest roadblock to competition is the ownership of telephone numbers. Subscribers are reluctant to switch to a new service provider because they have to give up their telephone numbers when they switch to a new service provider. LNP enables subscribers to switch to a new provider while keeping their existing telephone numbers.

This presents a huge challenge to the telephone industry. Until now, the routing of telephone calls has been based on the first six digits of the telephone number (NPA-NXX, or area code and office code). If a subscriber moves to a new area or elects to change service providers, this is no longer possible. The telephone switches once identified by the old numbering plan are suddenly faced with servicing numbers from other service providers.

If a subscriber moves across the country, the problem becomes more complex. Telephone equipment and software throughout the network have been designed to use the telephone number to determine the geographical location of subscribers. For example, everyone knows that the 212 area code is Manhattan. However, with LNP, a subscriber with a 212 area code could live in California. This problem is compounded when one looks at billing systems, *operations support systems* (OSSs), and other network subsystems, which all rely on the numbering plan for determining a caller's geographic location and service provider.

This chapter will outline the impact of LNP on the telecommunications industry, how it works, and how *Signaling System 7* (SS7) is used to implement LNP for both wireline and wireless networks.

Introduction

There are three phases to LNP. The first phase, *service provider portability*, is being implemented now. This enables a subscriber to select a new local service provider while keeping his or her existing telephone number. It also enables a subscriber to move within his or her rate center while maintaining the same telephone number.

The next phase of LNP is *service portability*. This enables subscribers to change the type of service they have while keeping their telephone numbers. For example, if a subscriber changes from a *Plain Old Telephone Service* (POTS) line to an *Integrated Services Digital Network* (ISDN) service, he or she has to obtain a new telephone number, because the switching equipment used to provide the ISDN service supports a different block of numbers. With LNP, the subscriber does not have to give up the telephone number when changing the type of service.

The most difficult phase of LNP is *location portability*. This will enable a subscriber to move from city to city, or even state to state, while maintaining the same telephone number. This has a much more global impact. Even subscribers are accustomed to associating geographical areas with telephone numbers. It will be difficult for anyone to determine where he or she is calling once location portability is implemented.

Currently, porting is only supported within a rate center. A rate center is a geographic area, usually within a *Local Access and Transport Area* (LATA), which is used for determining the time and distance used in the billing of phone calls. Rate centers are determined by using vertical (V) and horizontal (H) coordinates. Porting a number outside of a rate center will present many technological challenges in itself and is being addressed in later implementations of number portability.

The Telecommunications Act of 1996 and the FCC mandate (docket 95-116) do not specify exactly how LNP is to be implemented. They simply outline the ground rules to be used when implementing it in the network. The first trial of LNP took place in Illinois under the direction of the *Illinois Commerce Commission* (ICC). The ICC specified some rules regarding the actual implementation, such as using a *location routing number* (LRN). These have been published in the AT&T specification FSD 30-12-0001. These implementation requirements have become the de facto standard for LNP and have been widely accepted throughout the industry.

LNP Impact

There have been several proposals for providing LNP without implementing a database solution. The first solution was call forwarding. This was quickly rejected by the FCC as an interim solution because of the delay imposed on the calling party while the carriers tried to route the call. The FCC did not want subscribers punished for changing providers, and it has stringent requirements regarding the level of service provided to subscribers when they switch carriers.

Another approach was *Query-on-Release* (QoR). When a call is routed to a number that has been ported, the receiving switch identifies the number as being vacant and returns an SS7 REL with an appropriate cause code. The originating switch would then initiate a database query to determine if the number had been ported. This approach certainly reduces the traffic across the SS7 network and lessens the impact of the database queries, but again it places unnecessary delays on setting up telephone calls to subscribers who have changed carriers. QoR was also rejected as an interim method.

The solution that was chosen was the LRN method. The end-office switches in the rate center have a table identifying all NPA-NXXs, which have numbers in them that have been ported. The specific number is not provided in the database, so the switch must initiate a query if it is determined that the number dialed was to an NPA-NXX considered as ported.

The database provides a LRN, which is explained later when we discuss call flows. The LRN method places a higher demand on the SS7 network, but ensures there is no degradation of quality or service for the subscriber who changes carriers. Unfortunately, the LRN also imposes huge unrecoverable costs on telephone companies.

There has to be some event that causes a query to take place. The industry has agreed that *Intelligent Network/Advanced Intelligent Network* (IN/AIN) triggers should be used to initiate queries. A trigger expands the call-processing capabilities of switches by *triggering* defined events to take place (like initiating an LNP query) when specific criteria are met. For example, if received dialed digits equal a specific value, a query is sent to obtain additional routing instructions. This will require software upgrades in all switching equipment to support IN/AIN triggering.

To understand the impact that LNP has on the telephone network routing, you must first understand how networks treat telephone numbers. The first usage of telephone numbers came with the early manual switchboards. Local town operators were finding it difficult keeping up with all of the town's citizens by name. In those days, if you wanted to call someone, the operator had to connect the call for you. There were no telephone numbers; everyone was known by name. It was a doctor who first suggested using a numbering system instead of names, using his own practice as an example (doctors maintain an elaborate and numbered filing system to maintain patient records).

The early telephone numbers were sometimes given names for the "exchange" they served. The first three numbers after the area code were given letters; those letters identified the exchange the telephone number resided in. The exchange was the area that a particular telephone switch (or group of telephone switches) served. For example, in my study is an antique telephone from an office in Los Angeles, with the exchange MAD for Madison exchange still on its dial.

Over time this changed, and the names were lost, but numbers continue on. Telephone switches are assigned blocks of numbers, with the first three digits (office code) identifying the particular switch or central office the subscriber number is served by. When calls are routed, only the first six digits are used (area code/office code, or NPA/NXX). When the call is delivered to the correct destination central office, the switches recognize the office code as their own and route the call by the last four digits (the subscriber number).

Likewise, billing systems use the telephone numbers in much the same way. Telephone companies established rate centers by dividing the exchanges into geographical areas. These areas could be measured for distance by using H/V coordinates. When a call is placed, the area/office code is used to determine if the calling party is making a local or a long-distance call. If it is long distance, the charges are applied according to a complex formula using the H/V coordinates. What happens if you take that telephone number and use it in another city? You can begin to see the complexity now of LNP.

This complexity is even more severe when you examine the wireless network. Wireless providers face a greater challenge ahead because of the very nature of their networks. Subscribers are already "portable," moving from cell site to cell site. Their billing is determined by a completely different plan and does not match the same system used by wireline providers.

Each subscriber is assigned to a home *mobile switching center* (MSC), which falls within a specific wireline rate center. If a mobile subscriber calls a wireline number, the billing is determined by the distance from the MSC to the wireline number. However, that same subscriber could be roaming (outside their home area), making the billing more difficult. Previous to LNP, the wireless subscriber's *mobile identification number* (MIN) has been used for determining the home MSC and how they would be billed for calls when roaming. If they take that MIN to another wireless provider, it becomes even more difficult to determine what charges should be applied. The MIN is also used for identifying the carrier providing the wireless services. The first six digits of the MIN identify the service provider for that subscriber. This means the MIN can no longer be used for call processing, because the MIN cannot be ported.

Wireless networks rely on the MIN for call processing, billing, and virtually every transaction related to a mobile subscriber. However, use of the MIN to address portability would require too many database queries and impacts global title databases. The wireless industry has elected to change the identification of mobile subscribers by assigning two numbers; the *mobile directory number* (MDN) and the *mobile station identifier* (MSID). The MSID can use the MIN format, but the MSID is not portable. The MDN, however, is portable. This changes the way call processing takes place in the wireless networks, impacting virtually every element in the wireless network.

In international wireless networks, the MIN is not recognized. Instead, networks use what is called the *international mobile station identifier* (IMSI), which is recognized in any network overseas. These networks are usually based on *Global System for Mobile Communications* (GSM) technology. Wireless providers in the United States have been preparing to change from the MIN format to the ISMI format, so they could support international roaming. LNP offers an opportunity to use the IMSI format when assigning MSIDs to mobile subscribers.

Another impact on wireless networks will be the method used to query databases. Wireline networks have agreed on the IN/AIN triggers for querying databases; however, wireless networks do not necessarily support IN/AIN. The industry is looking at IS-41 and GSM protocols for querying the LNP database. This will place a burden on developers of LNP products, who must be able to support all solutions for both wireline and wireless. Both the IS-41 and GSM protocols are being modified to support additional parameters for LNP.

Suffice it to say, there are a number of issues surrounding the implementation of LNP in both the wireline networks and the wireless networks. A lot of these issues have been resolved; some are still being investigated. What is clear is that the networks can no longer rely on the telephone number for routing, billing, and network operations. This means that millions of dollars will be spent changing the software in existing equipment and adding new hardware (such as databases and data links to access those databases) to make LNP work.

In addition to changes in the network equipment, SS7 has been modified as well. The solution to LNP has required new parameters to the protocol, mostly in the area of the *ISDN User Part* (ISUP). Additional information is now required to route calls. The solution agreed upon by the industry was the assignment of a new number, called the LRN. The LRN is assigned to each end office, using the same existing numbering plan in place today. In most cases, if an end office serves, say, the 919-460 area, NPA-NXX, its LRN would be 919-460-0000. This preserves the concept of associating routing with geography, but does require changes to routing tables and software. We'll discuss how this all works a little later.

Not all networks support SS7. Some networks still rely on *multifrequency* (MF) signaling. There are interworking procedures defined when SS7 networks must internetwork with MF networks; however, there is an efficiency problem introduced with LNP.

If an MF network becomes part of the call path, the information contained in the ISUP *initial address message* (IAM) cannot be transferred across the network. This means all LNP data will be lost. If there is an SS7 network on the other side (in other words, the MF network is between two SS7 networks), then the SS7 network receiving calls routed from the MF network must initiate another query to an LNP database to determine how the call is to be routed.

This introduces an additional delay on processing the call and introduces the risk that the call may be handled improperly if for some reason the query cannot be performed at the receiving SS7 network. These conditions are still under study, and procedures are being established for handling calls with MF signaling networks.

Databases and OSSs used throughout the SS7 network face changes. Many of these systems rely on specific fields within the SS7 ISUP messages, which have now been modified to support LNP. These systems will have to be upgraded to understand the new message format. For example, calling number delivery depends on the ISUP IAM calling party field. This may no longer be the actual directory number of the person calling, but the LRN of the calling party's exchange. There are many similar scenarios, all of which have been defined in the Bellcore specification GR-2936-CORE.

Congestion within a *service control point* (SCP) is a real threat, especially with so many queries being made for LNP. *Automatic call gapping* (ACG) has been identified as the method for preventing SCP congestion. ACG defines how long (duration) and how often (intervals) transactions should be placed under control. ACG will be supported in both wireline and wireless networks.

The 1996 FCC mandate provided a schedule for the implementation of LNP. This schedule was extremely aggressive, and by the time this book was published, most of the LNP milestones will have been met. The mandate required LNP to be demonstrated within the top 100 MSAs by December 1998. However, the FCC issued a deployment schedule starting as early as October 1997. LNP was fully operational in the top 100 MSAs by June 1999.

Wireless service providers were issued a reprieve and are still granted extensions to have LNP implemented in the top 100 MSAs. They have nine months to implement LNP when they are requested by another carrier to provide number portability. This means that if a carrier loses a subscriber to another carrier, and that carrier requests the number to be ported, the wireless provider has nine months to make that happen. Wireline service providers were given six months after a request to implement LNP.

Smaller service providers are allowed to file for exemption from LNP under certain conditions. If the carrier has fewer than 50,000 access lines, they can be exempt from the LNP requirements. There are other conditions that must be met as well, all published in the FCC mandate.

Of major concern to all carriers is the responsibility of sending a query to an LNP database. A carrier could have a subscriber place a call to a number that has been ported, and if they do not have LNP in their network, route the call as normal. The receiving network would be unable to connect the call, because the number would be found as vacant in the switch's database. The receiving network would then be left with the responsibility of sending a query to an LNP database to see if the number had been ported.

This impacts other networks and shifts the fiscal responsibility on other carriers, which the FCC mandate clearly cited as a violation. To prevent this type of "dumping," the industry has agreed on an N-1 scheme. The next-to-last network in a call route is responsible for making the database query. For example, if only two carriers are involved in routing a call, the originating network is responsible for sending the database query.

If several networks are involved in routing and connecting the call, the next-to-last network (which would be a long-distance carrier) is responsible for sending the database query and routing the call to the appropriate network. This approach is the most equitable way to prevent carriers from dumping calls on other carriers and makes everyone responsible for implementing LNP.

The LNP Elements

Several elements are required to support LNP on the network. Figure 10.1 shows the general architecture of the LNP network. Each of the elements is described in the following text.

Number Portability Administration Centers (NPACs)

A *Number Portability Administration Center* (NPAC) is managed by a third-party company with no interest in the telephone business. Presently, two corporations are responsible for the management of NPACs, Martin Marietta and Perot Systems. An NPAC is responsible for receiving requests from carriers (recipient carriers) for the

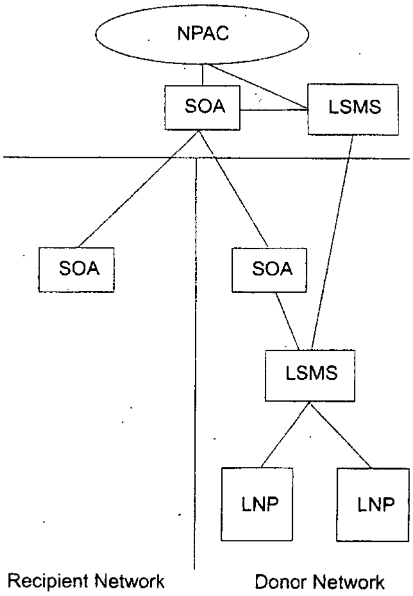


Figure 10.1

porting of telephone numbers (from donor carriers). They then coordinate the porting of the number by sending the data to the donor network, confirming the request has been accepted, and then downloading the ported number data (which is the new LRN for the telephone number and other routing information) to all of the other networks connected to that NPAC.

There are seven NPAC regions, roughly aligned with the regions designated for the Bell Operating Companies. Perot Systems manages the Southeast, West, and West Coast regions. Martin Marietta manages the Midwest, Southwest, Mid-Atlantic, and New England regions. The NPAC network includes a *service order administration* (SOA) system as well as a *service management system* (SMS). Their functions are the same as those described next.

Local Service Order Administration (LSOA)

All order information regarding a number being ported is sent to the SOA system. The SOA is used to process a subscriber's order and track the order through completion. It provides all departments with a single record location regarding a service order and is used to coordinate and track service order activities.

In the case of LNP, the subscriber data includes the LRN of the serving (donor) carrier, the date/time the number is to be ported to the new carrier (recipient), and other pertinent information. The SOA communicates this information to the SOA in the NPAC, which in turn passes the information to the SOA in the donor network.

It is important to understand that the only purpose of the SOA is to track the activities of an order and, in the case of LNP, to provide the specifics about when a number is to be ported and who the donor network is. Some of this information will be used by the billing center, while some of it is passed to provisioning systems to be configured in switches and databases in the new network.

Local Service Management System (LSMS)

Each carrier has a *Local Service Management System* (LSMS), which serves as the interface between the carrier networks and the NPAC. The LSMS is responsible for collecting porting data and downloading it to the LNP databases. The LSMS is usually a computer system with database storage and must be able to verify the data within the database with the data stored at the NPAC. This is accomplished through periodic audits between the LSMS and the NPAC.

Likewise, the LSMS also audits the LNP databases within its own network. It is crucial that the data is accurate. If discrepancies exist in the data, then telephone calls cannot be completed to ported numbers properly. The computer systems used for LSMS must have high reliability, and in many networks, they are deployed as mirror systems.

A mirrored system is one where a duplicate LSMS is placed in a different geographical location, but contains a mirrored image of the data of its mate system. This configuration provides diversity within the network. Diversity is as crucial with LNP as it is within other parts of the SS7 network.

The industry standard interface used between the LSMS and the NPAC is a Q3 interface, using the *Open Systems Interconnections* (OSI) stack for communications. The protocol used to communicate between the two entities is CMISE and ROSE. The communications protocol is used at the network layers of Q3 and *Transmission Control Protocol / Internet Protocol* (TCP/IP).

The interface between the LSMS and the LNP database has not yet been standardized, but Bellcore is actively working on defining Q3 as the interface between these two entities as well. Most systems being implemented today are using this Q3 interface between the LSMS and the LNP database.

LNP Database

The actual database used to maintain LNP data can be one of two types: an SCP or an integrated *signal transfer point* (STP). The integrated STP solution is more favorable in many cases because of the throughput that modern STPs can provide in comparison to SCPs. The top-of-the-line SCP is only capable of 850 transactions per second, while some newer STPs are capable of 20,000 transactions per second and higher.

The actual throughput requirement of the database itself will depend on the calculated number of database queries to be performed. To calculate the number of queries (transactions) per second, remember that every call made to an NPA-NXX with a directory number that has been ported will require a database query. This can be significant in some areas.

You also have to factor in the average message size of an LNP query. LNP queries average around 120 bytes in length, so if you are using 56 kbps links, you can transmit approximately 58 messages per second if your link is transmitting at 100 percent of its capacity (56,000/[120 × 8]). This, of course, is never the case since links are engineered to transmit at 40 percent capacity. Actual throughput of a 56 kbps link with an LNP message size of 120 bytes would be more like 23 LNP transactions per second.

Summary

This chapter covered the basic elements used in an LNP network. Notice that nothing was mentioned about STPs, or SCPs for that matter. The database function itself can reside in either an STP or an SCP. The only connection to the SS7 network provided is the database itself. The LSMS does not connect to the SS7 network, and neither does the NPAC. All communications between these entities are through a private communications link using Ethernet and TCP/IP protocols.

The SS7 network uses the information provided by the LNP database to route calls through the network. This function is much like the database function used in 800 services. Figure 10.2 illustrates the relationship between the SS7 network elements and LNP elements.

Now that we understand the elements needed to implement LNP, let's look at how LNP works, first in the wireline network and then in the wireless network. We will review the role of each of the elements in these discussions as well.

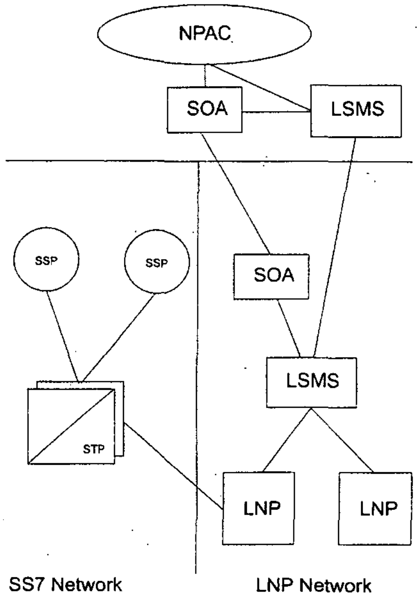


Figure 10.2

The Wireline Solution

In the wireline network, LNP is simpler. This is because the subscribers are all fixed in location. In the wireless network, the subscribers are mobile, making billing applications and OSSs more complicated. Before discussing the specifics about how a number is ported and then routed, let's first look at the changes made within the SS7 ISUP protocol to support LNP.

The ISUP IAM message has been modified to include information regarding a ported number. The most significant change is the placement of the LRN. The called party number field is no longer the directory number of the called party, but now is the LRN of the called party if the called party directory number has been ported. The directory number of the called party is moved to the *generic access parameter* (GAP) of the IAM message. The type of address parameter in the IAM message is the *ported number*.

The *forward call indicators* (FCI) field of the IAM now includes a translated called number field. This 1-bit parameter in the 8-bit field is used to indicate whether or not a number has been translated. This is used by switches to determine if a query is needed, and whether or not routing should be based on the GAP or the called party number field. Even if a number has not been ported, the forward call indicators will still indicate "number translated."

The *jurisdiction information parameter* (JIP) is used for billing purposes. The LRN of the originating end office is placed in the JIP field. Billing systems can then determine how the call is to be processed. Only the first six digits of the LRN are needed.

Now that we understand the modifications to the SS7 protocol, let's look at how telephone numbers are ported and how calls are routed to ported numbers.

Porting a Number in the Wireline Network

When a subscriber places an order with a competitive local service provider (a recipient carrier), it is up to the new carrier to send a request to have the subscriber's telephone number ported. The request is sent from the recipient network LSOA to the NPAC SOA. The NPAC SOA is then responsible for notifying the original carrier (the donor carrier) of the request.

The NPAC sends the request down to the donor carriers' LSOA, which then verifies the number, is currently assigned in the network, and confirms receipt of the porting data. The porting data consists of the subscriber's telephone number, the date and time the number is to be ported, and the LRN of the recipient carrier. The NPAC then forwards the confirmation back to the recipient carrier's LSOA.

The NPAC then sends the porting information to its own LSMS, which then distributes the information immediately down to all the LNP databases within its region. The number has now been ported, and the actual switch from one carrier's switching equipment to the other takes place according to the porting data. Obviously, coordination between the two carriers is critical in order to prevent the subscriber from being disconnected by his or her original carrier before the recipient carrier can activate service.

The donor carrier is not responsible for notifying the recipient carrier of the services in the subscriber's profile. It is still the responsibility of the subscriber to notify the recipient carrier of which services he or she wants to have when he or she ports. If the subscriber disconnects his or her service at any time, the telephone number is returned to the donor carrier (until the LNP service matures and number assignment becomes more universal).

Routing a Call with LNP in the Wireline Network

Each end-office switch is assigned a 10-digit LRN. Each switch is required to support at least two LRNs (in the event that the switch previously serviced more than one NPA-NXX). This LRN consists of the same area/office code previously assigned to the end office. The last four digits are typically all zeros. By using an LRN, the existing numbering plan can be utilized. Routing is based on the LRN instead of the dialed telephone number when a number has been ported.

When a number is dialed, the originating switch looks up the NPA-NXX of the number in a table. This table will indicate whether or not the NPA-NXX is considered to be ported or not. If even one telephone number within an NPA-NXX is ported to another carrier, then the entire NPA-NXX is considered ported. This means every phone call to that NPA-NXX requires a database query to determine whether or not the dialed number has actually been ported to another carrier and, if so, what the LRN of the new carrier is.

Before we continue discussing routing, we need to understand why the end-office switch only knows about the NPA-NXX and not the subscriber number. It would be easy to store routing information about every number in the end-office switch, alleviating the requirement to access a database. However, propagating this information to many switches and maintaining that data is a different story altogether.

The alternative is to query a database on each and every call, but the impact on SS7 is too large, and the capacities of almost every network would seriously be taxed. It was for these reasons the industry elected to use the LRN method for routing, and to propagate only the NPA-NXX information in the individual switches. The impact on networks is still high, but the solution is preferred over more costly methods.

If an end-office switch determines that a number belongs to a ported NPA-NXX, it originates a query to the LNP database (provided the switch has been equipped with that capability). In the event the switch has not been equipped to perform LNP queries, the call is routed as normal through the network to the donor carrier. When the call reaches the donor network's end office, the switch will determine that the call cannot be completed, because that switch no longer services the dialed number. The directory number will appear as vacant in the end-office switches database.

The question is then raised as to who is responsible for originating an LNP query. It would be very easy for carriers to simply route calls to neighboring networks without performing queries, thereby alleviating the need to implement any method of LNP on their own networks. Simply let the neighbors bear the cost of implementation and pass all your traffic to them. The FCC recognized the potential of call dumping and passed a ruling that the next-to-last carrier (or network) was responsible for originating the LNP query. This is known as the N-1 method. If only two carriers are involved in the call setup, the original carrier is responsible for originating the LNP query.

The LNP database will provide the LRN of the servicing carrier if a number has been ported. This information is placed in the SS7 ISUP message using the called party parameter. The actual called party directory number is moved to the GAP. This is one of the reasons why upgrades to virtually every network node in the SS7 network are required to support LNP; ISUP messages have been modified to support this feature. The ISUP parameters that have been modified have been discussed.

Let's look at a typical call to a number that has not been ported, but is in an NPA-NXX considered "portable." Figure 10.3 shows an example of the call flow in the case of a nonported number.

In Figure 10.3, the calling party dials the number 919-460-2100. The end-office switch servicing the called party identifies that particular NPA-NXX as one that has been ported. The end-office switch then originates a database query to the LNP database.

The query is sent to the STP, which then examines the address information in the SCCP portion of the message to determine how to handle the query. The SCCP portion of the message identifies the type of database to direct the query to, if that is known. If the end-office switch does not know the network address of the database, the STP can perform a global title on the SCCP portion to determine the type of query and the database to send the query to.

The LNP database checks the database, determines that the called number 919-460-2100 has not been ported to another carrier, and returns the query results to the end-

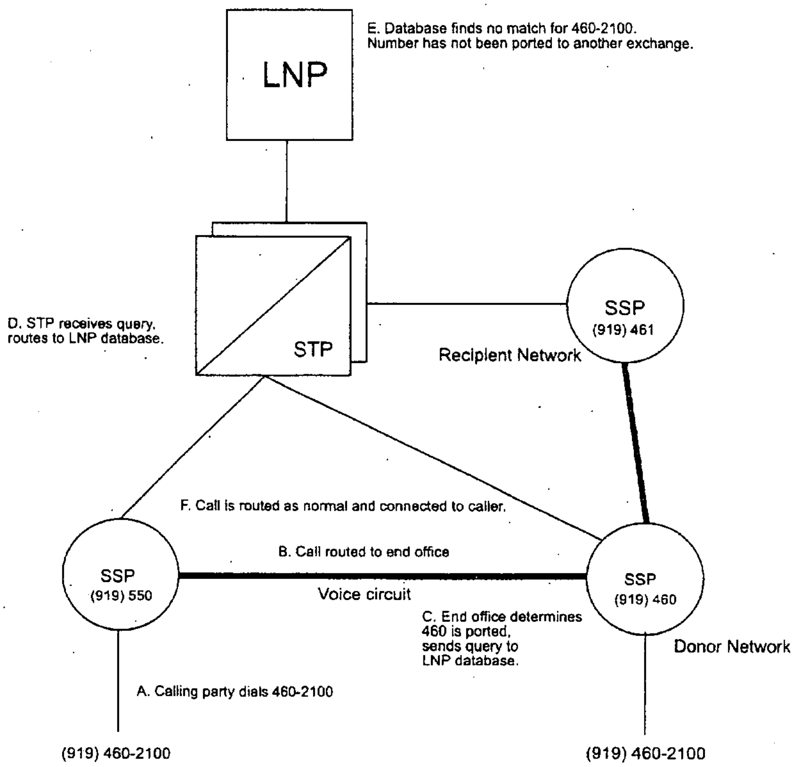


Figure 10.3

office switch. The end-office switch then routes the call based on the called party address field in the ISUP message. The call is connected.

This is an example of how a call is connected to a nonported number. Now let's look at how a call gets connected to a ported number. In Figure 10.4, a call is made to 919-460-2100, but this time the called party has switched carriers, and the number has been ported to the new carrier.

Again, the end office has determined that 919-460 is a ported NPA-NXX and initiates a query to the LNP database. The LNP database looks for the dialed number in its database. A match is found, and the LNP returns the results back to the end office.

The end office now has the information needed to route the call. The call is routed based on the LRN instead of the called party number. The originating end office generates an ISUP IAM message with the LRN placed in the called party address field and the dialed number in the GAP. The JIP identifies the NPA-NXX to be used for a billing reference.

The call is routed to the recipient network, which routes the call to the subscriber using the dialed digits in the GAP. Billing systems that determine the origin of the call

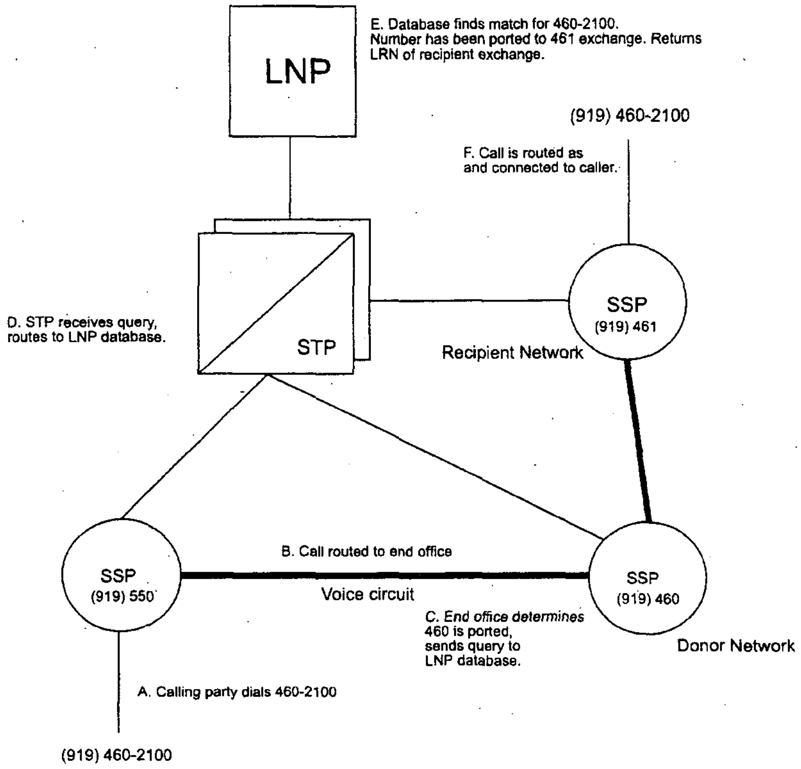

LNP
E. Database finds match for 460-2100. Number has been ported to 461 exchange. Returns LRN of recipient exchange.
(919) 460-2100
F. Call is routed as and connected to caller.
D. STP receives query, routes to LNP database.
STP
SSP
(919) 461
Recipient Network
B. Call routed to end office
SSP
(919) 550
Voice circuit
SSP
(919) 460
C. End office determines 460 is ported, sends query to LNP database.
Donor Network
A. Calling party dials 460-2100
(919) 460-2100

Figure 10.4

use the LRN in the JIP. The calling party number was used to determine the geographic-specific digits needed by billing systems previous to LNP.

This explains the basics of routing a call in the wireline network using SS7 to deliver the necessary information to the network elements. The wireless network operates differently than the wireline network and presents challenges of its own.

The Wireless Solution

In the wireless network, the information needed to route calls to mobile subscribers is somewhat different than what is required in the wireline network. For beginners, the number itself does not render enough information about the location of the called party. They can be anywhere in the network.

Each mobile subscriber has a MIN that is used to identify the individual subscriber to the network. The wireless network relies on databases to find the mobile subscribers' locations, which change as they move through the network. We won't go into details about the routing of calls in the wireless network, as that was discussed in the beginning of this book. We will now look at how calls are routed using LRN in the wireless network and the impact LNP has on the wireless network.

A wireless telephone is a radio device. The phone must be compatible with the cell site equipment in the areas where it operates. Cases occur when a subscriber will not be able to port his or her wireless phone. In other words, subscribers may change wireless service providers and maintain their existing telephone numbers, but they will have to buy a new wireless telephone.

Another fundamental difference between wireline and wireless is in the billing and network boundaries. In the wireline network, boundaries are set by LATAs. The wireless industry does not recognize LATAs, and the service provider determines the boundaries. This can cause difficulty in determining how LNP is to be supported, because the ruling says that LNP must be supported within a rate center, which is not recognized by the wireless industry. The *Cellular Telecommunications Industry Association* (CTIA) is currently defining this. Some of the fundamentals of porting a number in the wireless network are discussed in the next section.

Porting a Number in the Wireless Network

When a subscriber ports to a new wireless service provider, the new carrier must provide service in the same area as the donor network. This is one of the principal requirements, since rate centers are not recognized in wireless networks. In the wireline network, the service provider has to be providing local service within the same rate center as the subscriber.

Other requirements must be met before porting can be supported. The wireline rate center associated with the directory number assigned to the mobile subscriber must be in the same geographic area as the home service area of the subscriber. This is also true of wireline subscribers wishing to port their numbers to a wireless service provider.

Routing a Call with LNP in the Wireless Network

Routing in wireless networks is different than in wireline networks because the subscriber is mobile. Databases must be used to locate the mobile subscriber, which results in delays in the call routing. With LNP, we are now introducing yet another database query prior to even locating the proper network.

There are several call scenarios in wireless: land-to-mobile, mobile-to-mobile, and mobile-to-land. All are handled somewhat differently. We will start by looking at a simple mobile-to-land call.

Figure 10.5 shows the call flow for a mobile-to-land call. When a mobile subscriber dials a number, the receiving MSC establishes a trunk connection with the PSTN. The receiving switch in the PSTN examines the dialed digits and determines that the dialed number is in a portable NPA-NXX. The switch sends a query to the LNP database using IN/AIN triggers to obtain additional routing instructions. The MSC in the serving wireless network can just as easily send this query if the wireless provider has the capability.

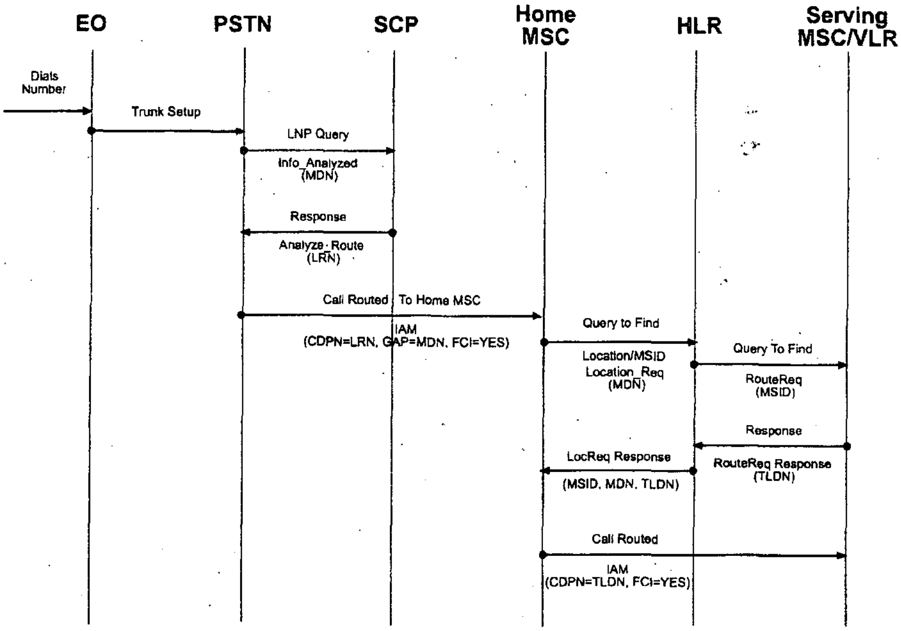


Figure 10.5

The query to the LNP database results in a response, providing the LRN to be used in routing the call. This is no different than the routing procedures used in the wireline network. The actual IN/AIN triggers have been added (Info_Analyze and Analyze_Response) in Figure 10.5.

Once the switch has the routing information it needs, it generates an ISUP IAM message containing the LRN, the dialed number, and the mobile directory number that originated the call. This call flow is not the same when a call is made land-to-mobile, as you are about to see. The location of the subscriber must be obtained first.

In Figure 10.6, we see the events that take place in a land-to-mobile call. The end-office switch initiates a query to the LNP database, sending the MDN. The LRN is returned (if the mobile number has been ported), and the end office establishes a trunk connection with the mobile's home MSC. Notice the IAM and the MDN that Figure 10.6 provides, but not the MSID.

The home MSC searches its own *home location register* (HLR) to determine the location of the mobile station. The HLR shows that the mobile station is in another network and provides the identity of the serving MSC. The HLR then sends a query to the serving MSC's *visitor location register* (VLR), which will provide location information.

The serving VLR returns a response to the HLR, providing the *temporary local directory number* (TLDN) used in roaming. The TLDN is a temporary number assigned to the mobile station as long as it is served in the visited network. This number is then assigned to a new mobile station when the previously assigned mobile leaves the network.

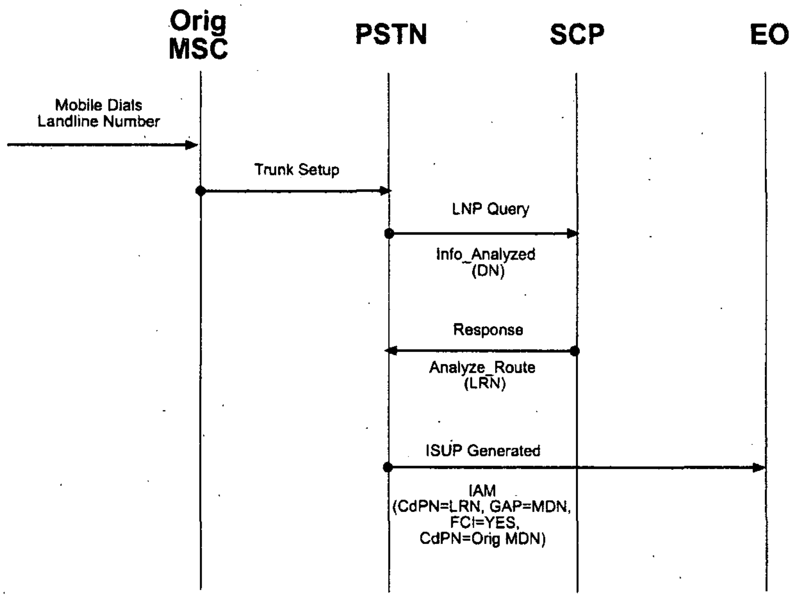


Figure 10.6

The HLR then sends a response to the home MSC providing the MSID, MDN, and TLDN. The home MSC can now generate the ISUP IAM. In this case, the called party number contains the TLDN, not the MDN. If the mobile station were still in its home network, this parameter would contain the MDN.

These call flows have been simplified considerably for ease of reading. Many other events must take place to route calls in the wireless network. This should give you an idea how the protocol works in obtaining the necessary routing information when the numbers have been ported in wireless networks.

Summary

LNP will impact everyone. The conventional methods of routing calls are gone, and new procedures will require many modifications to our nation's networks. These changes are not just impacting U.S. networks. The international telephone companies are also looking at implementing LNP in their networks. Many international carriers have already begun choosing solutions for their networks and have identified deployment schedules.

Developers of telecommunications products will also be impacted, but in a positive way. Their products must be modified to support LNP solutions. As mentioned in our previous discussions about wireline and wireless portability, these solutions will vary. Equipment vendors must support all solutions to market products in both the wireline and wireless markets. This means supporting IN/AIN triggers as well as IS-41 and GSM protocol changes.

As LNP moves forward, the subscriber will begin to feel the impact as well, but hopefully in a positive way. As subscribers, we will have many more choices available to us. We will now be able to choose who we want to provide our local telephone service as well as our long-distance service. In many cases, service providers will provide bundled services. These bundled services will include local, long-distance, wireless, paging, and even Internet access. Hopefully, as more competition enters the market, we will also see reduced prices for these services. Don't count on substantial savings, however; these network changes cost millions of dollars, and LNP alone is costing some telephone companies billions of dollars to deploy. These costs have to be recouped somehow, and always trickle back to the subscriber.

Chapter

11

IP Telephony

This chapter provides an overview of IP telephony and the protocols developed for signaling within IP telephony networks. Because many of the standards are still under development, I have chosen not to provide the details of these protocols. They continue to change on a regular basis. As the standards evolve, those that are specific to *Signaling System* 7 (SS7) will be documented in later editions.

What started as a fad has suddenly redefined traditional networks around the world. Although experts agree that *Transmission Control Protocol / Internet Protocol* (TCP/IP) networks are not favorable for telephony applications because of latency issues, companies are eagerly finding ways to circumvent the issues and take advantage of the Internet as a means for transporting voice and data around the world. Although TCP/IP may not be the ideal transport for real-time applications, it is certainly more efficient and much more economical than conventional networks used today.

One of the drivers is the cost of TCP/IP-based switches. Although they currently do not provide the capacities of traditional circuit-switching platforms, they are much cheaper, allowing small startups to deploy networks much quicker for a fraction of the cost. This has caused existing carriers to review their network plans and implement new deployments of IP-based equipment, in many cases replacing their existing legacy switches.

This, of course, introduces new challenges to the industry. The TCP/IP protocols must be modified to support telephony applications. The *Internet Engineering Task Force* (IETF), as well as the *International Telecommunication Union* (ITU), is actively modifying these standards today. Although voice transmission over IP networks is of primary concern, much is also to be done in the area of signaling.

This book will only focus on the work being done for signaling. The voice and data transmissions are really outside the scope of this book and are best served by other publications. We will review the architecture, however, because this is vital to understanding how signaling works in IP-based networks.

The timeframe for these networks is now. Already companies are deploying systems and transmitting voice traffic through the public Internet as well as specially engineered IP networks (the preferred approach). Many of the *Regional Bell Operating Companies* (RBOCs) have taken advantage of some of these IP carriers, although they would never admit that their traffic is being sent through the public Internet. In the next few years, we will continue to see widespread deployment of IP-based telephony networks with both old and new carriers. This is especially true of the wireless carriers, who are eager to explore the possibilities of IP-based services.

What Is IP Telephony?

IP telephony encompasses the use of IP-based networks for the purpose of transporting both data and voice. They depend on server-based systems as switches (sometimes referred to as *softswitches*). The general concept is to eliminate the intelligence found in many legacy switches today on the edge of the network and rely on more centralized servers to provide call control to the edge devices.

The subscriber is served by an IP-based device, which replaces the end-office switch used in conventional networks. The purpose of this device is to provide connections (via IP, *asynchronous transfer mode* [ATM], or whatever medium is required) to the subscriber. Ideally, this medium is IP. The voice is packetized using *customer-provided equipment* (CPE). All the switching intelligence is provided by a controller, which sits in a more regional location. The controller supports all of the devices within its defined zone.

The controller provides call control functions, instructing the devices on how to handle a specific call. Controllers use a signaling protocol to communicate with these devices as well as a signaling protocol to communicate with one another. All of these protocols are based on IP as the transport.

In order for these devices to communicate with devices in another network, there must be some device that provides a bridge. This requires yet another level of signaling. If calls are to be sent to the *Public Switched Telephone Network* (PSTN), SS7 is the only accepted form of signaling. A *signaling gateway* (SG) provides the bridge between IP networks and the PSTN.

What makes these networks different from conventional telephony networks is the capability to connect to an IP "cloud," rather than dedicated circuits between each of the entities. This is, of course, the primary advantage of using packet-switching technology rather than circuit-switched. While signaling system #7 is a packet-switched protocol, some adaptations are necessary to work in this environment. SS7 uses many timers at levels 2 and 3 to maintain the integrity of each "link" or network element and to guarantee the delivery of all messages. We will discuss this further later in the chapter.

By eliminating the direct connections between all entities, and by utilizing IP as the transport, carriers can now realize substantial savings in their network infrastructures. The devices used in these networks are substantially cheaper than circuit-switched equipment, and carriers can eliminate other expensive equipment such as multiplexers and transmission equipment based on *Time Division Multiplexing* (TDM).

Still, there is a need to connect to traditional telephony networks. This will be the case for many years to come because there is a huge investment in networks of this type, and IP telephony networks are still far too immature in design to replace TDM networks. As previously mentioned, in order to connect into the TDM networks, SS7 is required.

It is important to understand that SS7 is not necessarily being replaced by new IP signaling protocols. Rather, SS7 is being modified to meet the requirements of IP networks. When one looks at the *ISDN User Part* (ISUP) and *Transaction Capabilities Application Part* (TCAP) protocols, it becomes obvious that the services brought by these two protocols are not easily duplicated by any other protocols. They have been developed over many years to support telephony services. TCP/IP does not have any comparable protocols to support these functions. Creating new IP-based protocols to replace ISUP and TCAP would be far too complex and time-consuming. It makes far more sense to modify SS7 to support IP networks than to create new replacement protocols.

This is not to say that SS7 may not someday be replaced entirely. The telephony network as we know it today is undergoing dramatic changes. These changes may render SS7 obsolete some years from now. In the meantime, work continues on modifying the SS7 network to interoperate with IP telephony networks.

Support of SS7 transport over IP facilities is being addressed through both the ITU and the IETF. In the meantime, vendors are deploying proprietary solutions to get their products to market.

Perhaps the only vendor who has had any level of success with this is Tekelec. This company has developed a protocol interface called *Transport Adaption Layer Interface* (TALI) that is designed to transport SS7 over pure IP networks. To date, more than 140 vendors have downloaded this software (www.tekelec.com) and registered with Tekelec to implement this interface on their solutions.

Although I may be a little biased as an employee of Tekelec, I believe the TALI interface deserves discussion here because of its already widespread industry support. Although many vendors have tried solutions of their own, TALI is the only protocol in use today in live networks transporting SS7 over IP facilities.

Although TALI is receiving widespread acceptance, it is not meant to replace the work of the IETF. In order for vendors to interoperate with one another, they must be compliant with whatever industry standard is adopted. TALI is viewed as an enabler to allow vendors and carriers to begin their deployment of SS7 over IP today, rather than wait on the standard committees to complete their work.

The Signaling Challenge in IP

The challenge in SS7 is replacing or emulating the procedures found in a *Message Transfer Part* (MTP) in an IP network. TCP/IP does not provide the same functions as MTP; specifically, it provides link management, traffic management, and route management. These functions must be emulated in the IP world to enable SS7 to work. They are vital to maintaining the *quality of service* (QoS) we have become accustomed to here in the United States.

Standards are now focusing on emulating the services of SS7 levels 2 and 3 so these three functions can be provided seamlessly, regardless if signaling messages come from the PSTN or IP networks.

To understand the roles of the standards being developed, let's first look at the architecture of IP telephony networks and the elements providing call functions.

IP Telephony Network Architectures

IP telephony networks are not as complicated as they may seem. If you look at the basic functions of a legacy switching system, you will find three basic functions: call control, the switching matrix, and the physical transmission interface. The idea in IP telephony is to break these different functions out and distribute them through the network using inexpensive servers.

In Figure 11.1, the physical transmission interface resides at the edge of the network. The switching matrix is really a series of routers and hubs, providing input and output ports. Depending on the address of the call (determined by the call control function), the packetized voice is routed through these devices to the destination point.

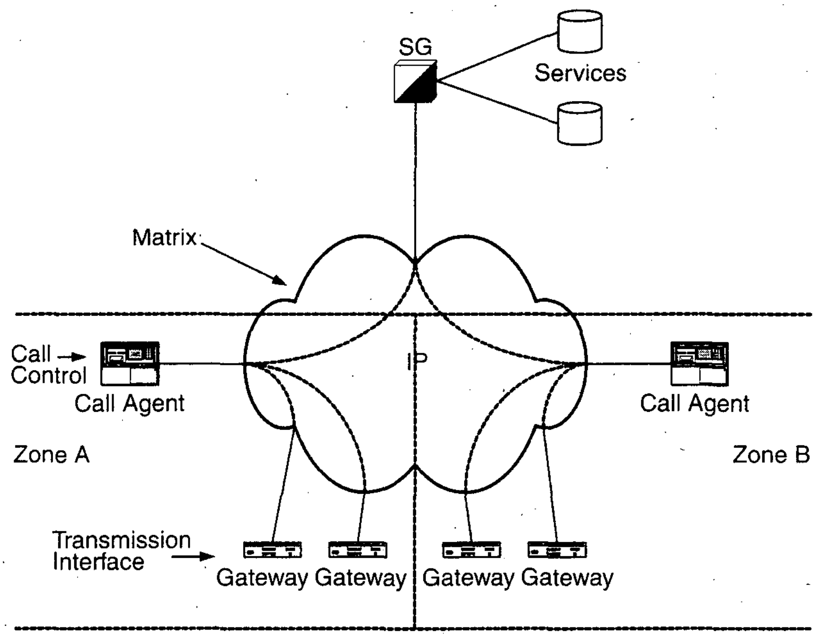


Figure 11.1 The basic functions of a circuit-switching system can be found in an IP telephone network, the difference being the functions have been distributed around the network using low-cost computer equipment rather than expensive switching equipment.

The call control function is deployed in zones. Each call controller (sometimes referred to as a call agent) manages the devices within its own zone. This is depicted in Figure 11.1.

The SS7 interface is more centralized in most networks. There are a couple of different thoughts about how SS7 is deployed in these networks. Some vendors use software stacks enabling SS7 in their solutions (such as controllers). Many have found out that these software stacks (implemented on PCs and servers) are not dependable and often result in failures. When these elements fail, calls do not pass through the network.

For this reason, many vendors have begun to migrate away from software stacks and partner with vendors to provide more telco-hardened platforms for SS7. *Signal transfer points* (STPs) that support the function of SGs can now be found. These are also being used as front ends to controllers and other entities in the IP network.

Figure 11.2 depicts an ideal network architecture. By centralizing the SG (or deploying in regions), carriers can control their most precious network resources more closely. As depicted in Figure 11.2, the intelligence of the network is provided by the SG, which acts as an interface to these resources.

Another concept still in its infancy is billing through SS7, rather than creating billing records at the edge of the network. In legacy networks, this practice is already gaining popularity. SS7 provides a much more efficient method for collecting call data used for

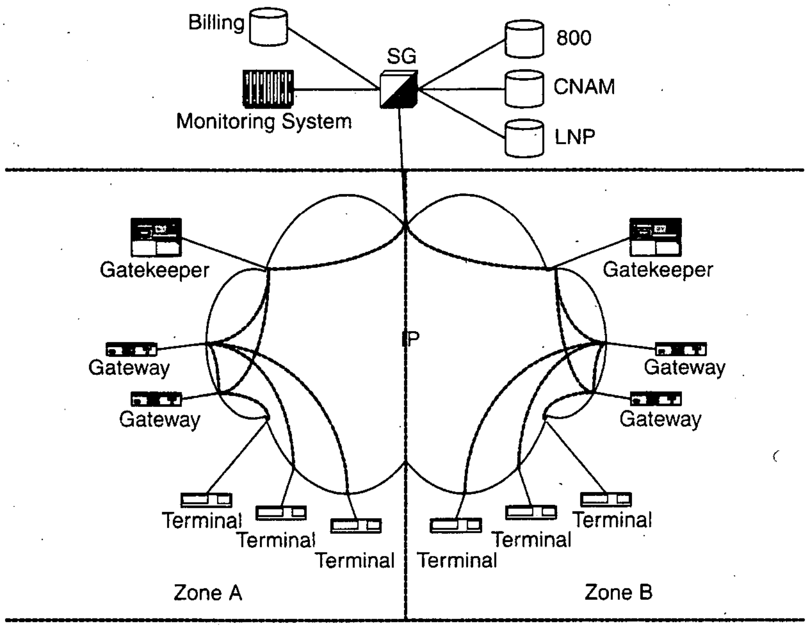


Figure 11.2 A more ideal IP telephony network architecture is illustrated here using ITU architecture. The same approach can be taken with IETF architecture. By centralizing the network intelligence, the carrier has much more control and flexibility over its network.

billing purposes. It eliminates the need for generating this data in switching equipment, collecting the data from these switches, and processing the data at some central billing center.

The same advantages exist in IP telephony networks. By centralizing this process using SS7, the edge devices responsible for actually generating phone calls and routing them through the network do not need software to generate call detail records.

Monitoring the entire network is critical, and several approaches can be used. One is to deploy probes throughout the network that then report back to a central monitoring system. Another is to actually connect to the SG and monitor the signaling messages as they pass through the SG.

Regardless of the architecture used, there is one critical factor to consider when designing signaling over IP networks. Latency is a critical issue in IP networks, and the round-trip time for messages to traverse the network must be closely engineered. An SS7 message should experience no more than a 100 ms round-trip time to prevent the expiration of level 2 and level 3 timers.

Standardization

The biggest issue with IP telephony is a lack of standardization. Both the ITU and the IETF are working diligently to develop the necessary standards to ensure interoperability between vendors' systems as well as to address the many QoS issues brought about by latency factors in TCP/IP networks. There is no clear indication when these standards will be completed, but suffice it to say that it won't happen quickly.

In the meantime, vendors are finding ways to deploy solutions using existing technologies as a workaround to the lack of standards. Q.9231 (also known as *broadband ISDN* [BISDN]), ATM, Frame Relay, and other older technologies are being used to take the place of nonexistent standards. ATM has been the biggest success story because carriers can use this as the network backbone and ATM provides QoS as part of the protocol.

Signaling, however, is another issue. Although some preliminary standards are being adopted and deployed in IP networks, there still is not a standard for SS7 transport through an IP network. The signaling protocols currently being used enable carriers to communicate within their own networks, but when connections to the PSTN must be made, they are left with no solution.

Basically, two functions have been defined for signaling protocol work: adaptation and transport. Adaptation functions act like *application programming interfaces* (APIs) interfacing between telephony and IP protocols. This is where Tekelec's TALI plays a major role, along with the protocols currently under development in the ITU and IETF.

The purpose of the adaptation protocols is to emulate the services provided by the MTP, maintaining link management, traffic management, and route management.

The transport function provides a reliable transfer of signaling traffic across the network. The IETF is currently defining a new protocol that will be a peer to IP called the *Simple Control Transmission Protocol* (SCTP).

To better understand the role the standards being developed play in IP telephony, we need to look at the architecture of IP telephony networks and the elements defined in these networks.

ITU H.323

The ITU is expanding on their H.323 Recommendations, which originally were to cover facsimile and video conferencing, but have since been expanded to include IP telephony.

The purpose of H.323 is to enable users to run multimedia applications, such as voice and video, using existing *local area network* (LAN) infrastructures without concern for latency factors.

The original specification for H.323 was for visual telephone systems as well as LAN equipment that provided a nonguaranteed QoS. Version 1 of this standard was accepted in October of 1996.

Version 2 was modified to accommodate packet-based multimedia communications, specifically *Voice over IP* (VoIP) and *Voice over ATM* (VoATM). Security was added in this version. Version 2 was accepted in January of 1998. Version 3 is now under development.

Figure 11.3 shows the functions defined in H.323 for IP telephony networks. This is where some confusion sets in because the ITU uses different terminology than the IETF. I have adopted the IETF terminology for simplicity, but also defined the ITU terminology here for clarity. It should also be noted that there is no one-for-one mapping between the functions defined by the ITU and those defined by the IETF.

The function at the customer premise is referred to as a terminal. This could be a computer, telephone, or any other device capable of supporting voice, data, and possibly video. As shown in Figure 11.4, terminals use H.245 to negotiate channel usage and capabilities (call control). The terminals communicate with gatekeepers for call control.

Call signaling (call setup and teardown) is provided by H.225. This is communicated from terminal to terminal (through the gatekeeper).

Gatekeepers are the central points for call control. They are deployed in "zones," controlling all terminals and gateways within their zone. Gatekeepers provide call control signaling, call authorization (based on the Q.921 specification), bandwidth management, and call management.

Figure 11.5 shows the functions provided by the gatekeeper. Many of the functions shown in the figure can be moved to a more central point, such as an SG, or they can be distributed throughout the network at the gatekeepers (as depicted in Figure 11.5).

Call signaling is accomplished using Q.921 signaling between endpoints. These signaling messages can be sent directly to the endpoint (provided they are within the same zone/network) or sent to the gatekeeper. The gatekeeper can either process the signaling message or simply pass it on to the appropriate endpoint.

Terminals register with their gatekeeper using *Registration Admission/Status* (RAS). This protocol enables the gatekeeper to maintain a status of all terminals within its zone. The gatekeeper also provides address resolution from LAN aliases to IP

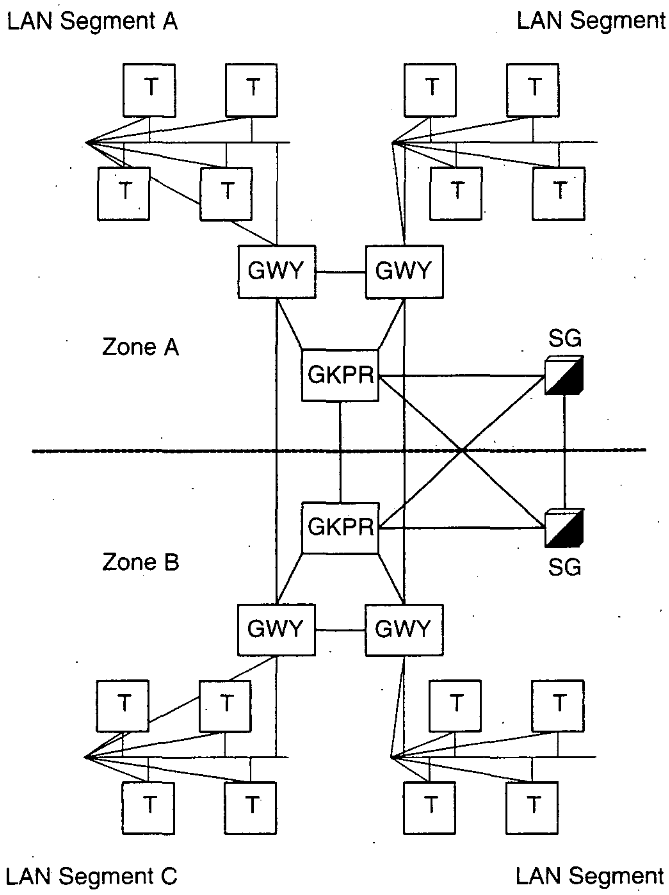


Figure 11.3 The ITU has defined a different set of functions than the IETF. The entities shown in this illustration are functions defined by the ITU. The connections shown between each function are not meant to be inerpreted as point-to-point physical connections, but as logical connections.

addresses. Three functions are provided by RAS. Registration is an endpoint announcing its presence and providing its LAN address. Admission controls LAN access based on call authorization, available bandwidth (and the bandwidth required for the call), and other criteria.

The gatekeeper can also provide call routing. The advantage of allowing the gatekeeper to provide this function is once again the centralization of intelligence within each zone. This also makes billing simpler within each zone. Call management uses a state machine to maintain the status of all active calls within the gatekeepers' zone.

In the case of bandwidth management, the gatekeeper can reject call connections based on bandwidth availability. This provides additional flexibility to network admin-

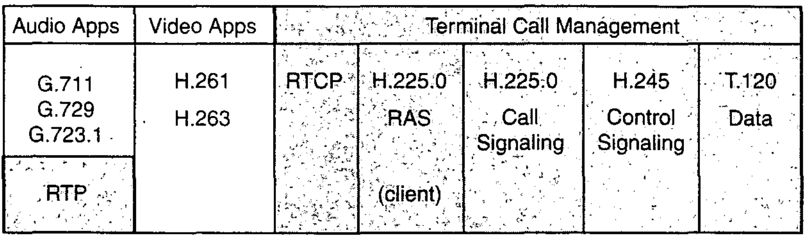


Figure 11.4 H.323 Terminal-side protocol stack. The shaded areas represent terminal resident standards.

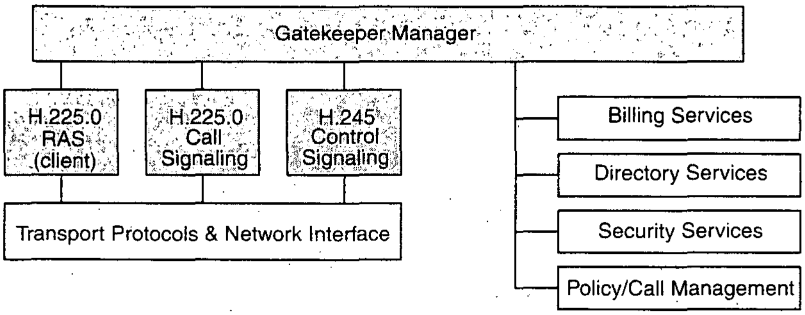


Figure 11.5 H.323 Gatekeeper protocol stack. The shaded areas represent gatekeeper resident standards.

istrators who can set thresholds in the gatekeepers, preventing multimedia applications from hogging all the bandwidth in networks shared by data applications. Call management maintains the status of all calls in progress through a state machine.

The gateway is an optional function in ITU networks. They are used when connecting to another network. Gateways provide protocol conversion and establish links between the PSTN and ISDN networks (Figure 11.6).

Protocol conversion may be necessary when connecting to dissimilar networks. For example, in the case of international networks, each country may be using its own version of SS7 as well as call signaling. In addition, the voice is encoded differently than here in the U.S. The gateway provides conversion of the signaling as well as the voice. Figure 11.7 shows the protocol stack by application, as defined in H.323.

IETF Sigtran

This working group within the IETF has been commissioned with defining a set of protocols to be used for signaling within IP telephony networks. Each of these protocols has distinct functions, performed in different segments of the network. Figure 11.8 illustrates a typical IP telephony network (high-level view) and where these protocols are being used.

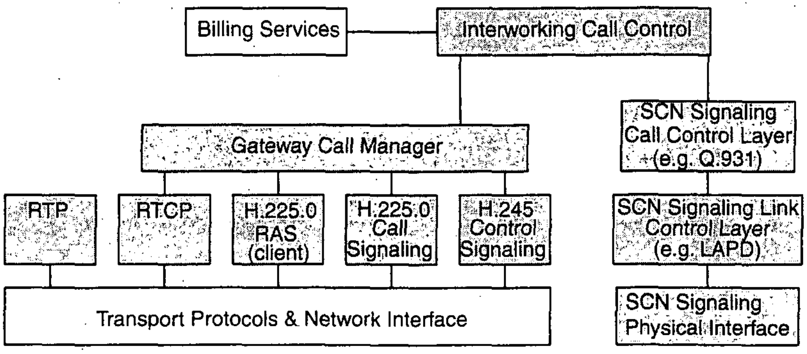
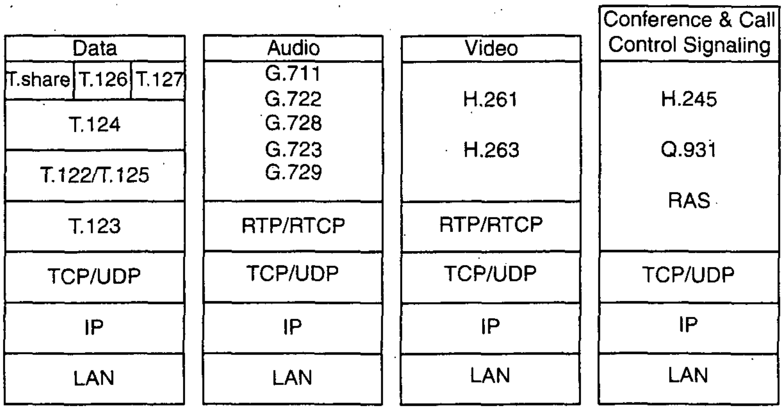

Billing Services
Interworking Call Control
Gateway Call Manager
SCN Signaling Call Control Layer (e.g. Q.931)
RTP
RTCP
H.225.0 RAS (client)
H.225.0 Call Signaling
H.245 Control Signaling
SCN Signaling Link Control Layer (e.g. LAPD)
Transport Protocols & Network Interface
SCN Signaling Physical Interface

Figure 11.6 H.323 Gateway protocol stack. The shaded areas represent gateway resident standards.



Data	Audio	Video	Conference & Call Control Signaling
T.share T.126 T.127	G.711 G.722 G.728 G.723 G.729	H.261 H.263	H.245 Q.931 RAS
T.124			
T.122/T.125			
T.123	RTP/RTCP	RTP/RTCP	
TCP/UDP	TCP/UDP	TCP/UDP	TCP/UDP
IP	IP	IP	IP
LAN	LAN	LAN	LAN

Figure 11.7 The ITU protocol stack for data, voice, audio, and video applications.

Media Gateway (MG)

The *media gateway* (MG) shown in Figure 11.8 supports packetized voice and the interface to whatever medium the voice is to be transported on. The MG performs the task of packetizing voice and providing connections from switched circuits (TDM) to packetized circuits (IP, Frame Relay, or ATM). The MG does not necessarily reside at a subscriber premise, but interfaces to equipment located at the subscriber premise.

The MG is responsible for media conversion, resource allocation and resource management, and event notifications. It is responsible for reporting events to the *media gateway controller* (MGC) within its zone.

Media Conversion

The MG must be able to provide conversion from TDM circuit-switched connections to ATM, IP, or Frame Relay connections. This includes the packetization of the voice itself.

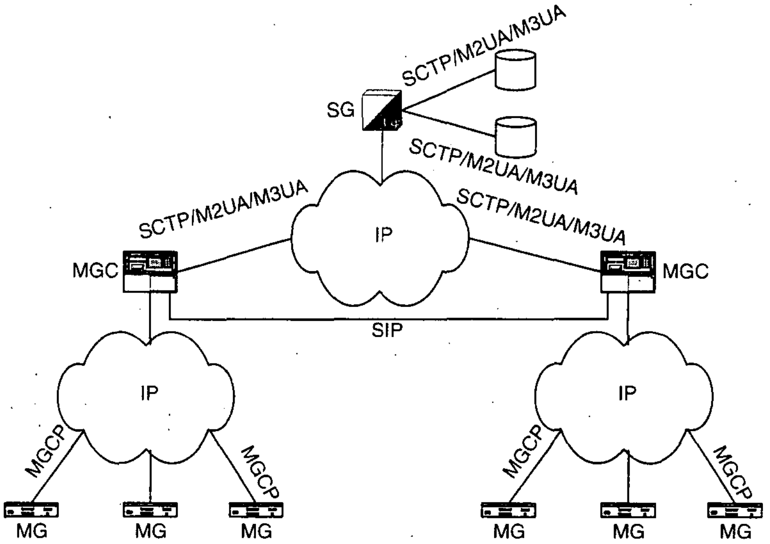


Figure 11.8 A high-level view of what an IETF network might look like, and what protocol standards would be used in this network.

Media processing includes transcoding, conferencing, interactive voice recognition, and other audio resource functions.

Resource Allocation

Resource allocation includes the reservation and release of all resources. It is important to understand that although the MG is responsible for resource allocation and management, it does so under the direction of the MGC. The MGC holds the ultimate responsibility of defining what resources are to be allocated for any one call.

The MG is capable of providing either point-to-point connections or point-to-multipoint connections (such as in a conference call). The MG must also support voice, data, video, and facsimile.

Event Notification

The MG must also maintain the state of all resources and report the state to the MGC. If a particular resource fails, the MG reports the failure to the MGC. The MGC maintains a state table for all resources within the MGs in its zone.

Media Gateway Controller (MGC)

The MGC is responsible for controlling all the MGs within its zone. As previously mentioned in the description of the MG, the MGC performs a number of call control functions that are typically found in the call control portion of a legacy switch.

This includes determining and controlling signals used on specific circuits, such as ringing and busy tones. The MGC also determines the QoS required for any one connection at the MG.

In short, the call handling is performed at the MGC, rather than at the MG itself. This enables carriers to deploy less expensive equipment at the edge of their networks.

Signaling Gateway (SG)

The IETF defines the *signaling gateway* (SG) as being the bridge to the PSTN. But it also supports the function of the SG at the MG level. When the SG function is part of the MG, the role is somewhat diluted from that of a SG providing STP functions to the network.

Keep in mind that global title translation and gateway screening are best accomplished at a different level than the MG. These STP functions should be more centralized, as depicted in Figure 11.8.

The SG should also be capable of providing conversions between SS7 addresses (point codes) and IP addresses. One method of address resolution utilizes the point code of an incoming message from the PSTN, along with the *circuit identification code* (CIC), to determine the appropriate IP address of the receiving MG. If a message is TCAP, the address resolution is provided using the point code and subsystem number.

Using IP in signaling networks rather than TDM point-to-point connections has a number of advantages. One of the most significant advantages is a more efficient (and less expensive) use of bandwidth. With TDM circuits, a lot of ancillary equipment is required to maintain the circuit. With IP networks, the equipment is much less expensive.

The addressing in IP signaling networks provides far more flexibility than in conventional SS7 networks. All entities in the SS7 environment are addressed through the use of point codes. The point code administrator in each country (in the United States, the *Science Applications International Corp* [SAIC]) issues point codes. In some countries, point codes are difficult to come by (due to rationing by the ITU).

When a carrier uses an IP network, the entities in the IP network are addressed by IP addresses rather than point codes. This requires the use of an SG to resolve the addresses from the SS7 network to the addresses in the IP network. The routing is based usually on a destination point code and, in the case of ISUP messages, the CIC range.

This means that MGs and MGCs can be added to the network and assigned IP addresses. The routing in the SG changes to provide routes to the new entities. The connection to other networks is through the SG, so any network additions do not necessarily have to be advertised to other networks. The carrier can maintain transparency to outside networks.

This is accomplished by assigning ranges of unique CIC codes to the gateways within the network. The SG uses one point code assignment for all the gateways and uses the unique CIC ranges to route the messages to the appropriate gateways.

If a carrier wishes to maintain even more flexibility in its network routing, a point code can be assigned (again within the SG) to regions with unique CIC combinations within each region (enabling for CIC duplication in different regions).

When TALI is used, automatic CIC registration is supported. This enables carriers to deploy additional gateways without the need to change the routing tables in the SG. The gateways advertise (register) their CIC ranges to the SG automatically.

The same scheme is used when routing TCAP messages. The routing is based on a point code and *subsystem number* (SSN). If a carrier is deploying a number of services, each service is assigned a unique subsystem number. The outside networks need only to know the one point code and subsystem number (or the point code alone when *Global Title Translation* (GTT) is provided).

When a database requires additional servers to increase capacity, an additional point code is not required (as is the case in conventional SS7 networks). The new server can be added using IP addressing, and the routing assignments can be made in the SG to ensure load sharing and subsystem management.

One of the biggest advantages to launching services in IP networks is the flexibility provided to the carrier. Rather than depend on expensive legacy SCP solutions, carriers can develop their own applications deployed on carrier-grade servers. Already many carriers have begun developing creative new services themselves, rather than depending on vendors to provide "canned" applications.

This factor alone will greatly enhance the concept of IP networking and provide great benefits to new carriers looking to differentiate themselves from the competition. Of course, as is the case with traditional signaling networks, the signaling network should be deployed first, providing an infrastructure that will enable the carrier to expand into new markets quickly and launch exciting new services to address the needs of their customers. Figure 11.9 depicts an example (the addresses are purely for example) of how IP provides additional flexibility in routing for signaling networks and valuable resources.

Three protocols are under development by the IETF for SS7 transport in IP signaling networks. These are described later in this chapter. The purpose of these protocols is to provide the transport of ISUP and TCAP messages across IP networks without the use of the MTP protocols.

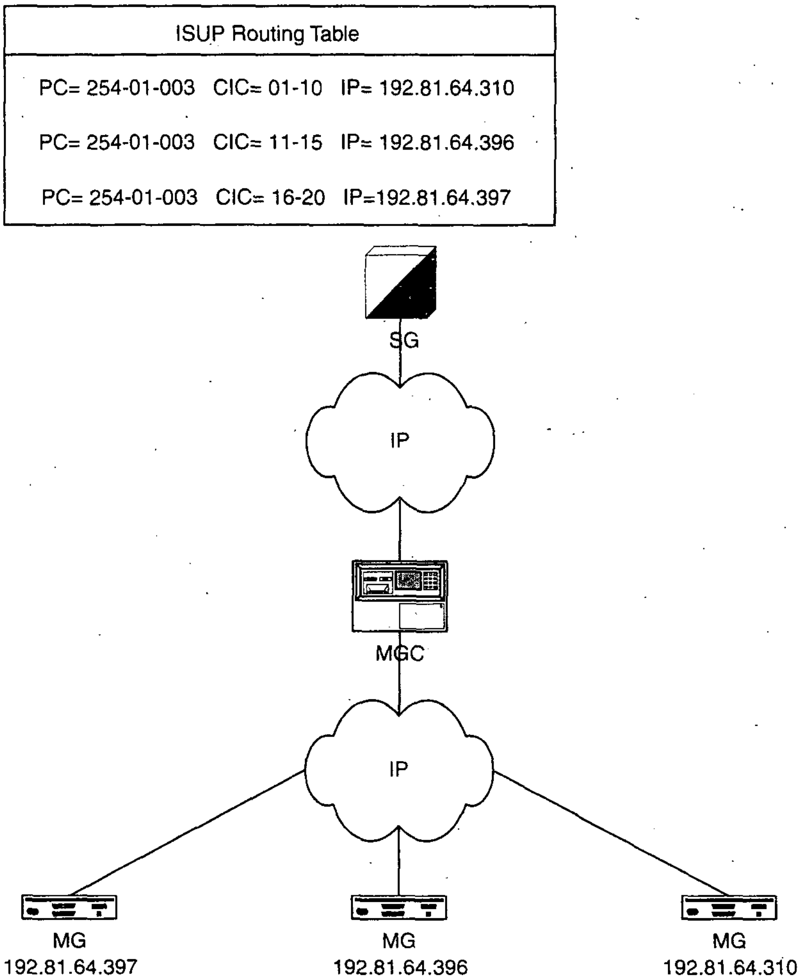
MTP is not eliminated entirely in IP networks. Parts of MTP are still required for routing purposes and for the destination entities to properly process the message. In these cases, the MTP portions are "tunneled" in IP or encapsulated into a TCP header for transmission to the destination.

Before we discuss these three new protocols, we must first look at the other signaling protocols within the IP telephony network and understand how they are used.

IP Telephony Signaling

Figure 11.10 shows the typical IP telephony network and the protocols used for signaling (or proposed for signaling) between the various entities. The ITU architecture or the IETF architecture has basically three entities: a gateway, a call agent responsible for call control, and an SG. The ITU network actually adds a third device for providing gateway functions between networks.

The signaling for these architectures varies depending on whether you are looking at IETF or ITU. This chapter focuses on IETF solutions, although ITU standards are shown here as well (in overview form).



ISUP Routing Table

PC= 254-01-003	CIC= 01-10	IP= 192.81.64.310
PC= 254-01-003	CIC= 11-15	IP= 192.81.64.396
PC= 254-01-003	CIC= 16-20	IP=192.81.64.397

Figure 11.9 SS7 point codes are used within the signaling gateway for routing purposes, but the true routing keys are the CIC for ISUP and the subsystem number for TCAP. A wide variety of combinations can be mixed and matched, though, using *origination point code* (OPC), *destination point code* (DPC), CIC, and SSN combinations.

Signaling between the various entities is not necessarily SS7. Several protocols are being developed for this function. Figure 11.10 shows the IETF protocols proposed for IP telephony networks and where they are used within the network.

Between gateways and gateway controllers (call agents) are two competing protocols: the *media gateway controller protocol* (MGCP) and Megaco. The Megaco protocol began as MGCP and later evolved into an entirely separate protocol under the working group Megaco.

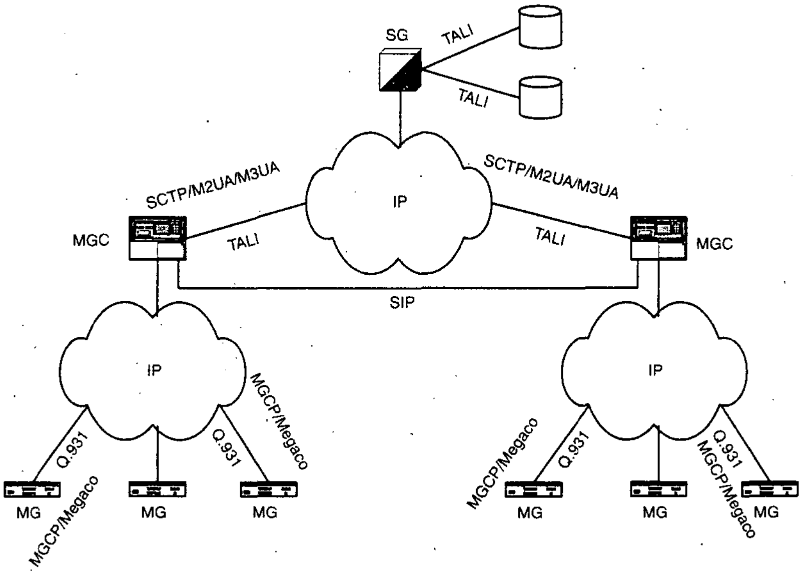


Figure 11.10 The signaling protocols proposed for IP telephony networks and where they are used in the network.

MGCP also continues to evolve with the endorsement of the International Softswitch Consortium. Either one may eventually become the finalized IETF standard after development has been completed. In the interim, vendors continue to implement both protocols in their solutions, selecting one over the other depending on their own opinions. The ITU has endorsed the Megaco protocol and is using it as the base for their H.245 protocol.

The purpose of either one of these protocols is to provide communications for the MG back to the MGC, and vice versa, for call control and call signaling. In some cases, even Q.931 can be used as an interim solution (as is the case with many vendor solutions). Keep in mind that these are for the purpose of signaling only. The packetized bearer traffic (voice, data, and so on) uses the IP *Real Time Protocol* (RTP).

The *Session Initiation Protocol* (SIP) was designed for communications between two MGCs. This is somewhat analogous to SS7 in a conventional telephony network. For SS7 transport within an IP telephony network, three new protocols are under development by the IETF: the *Simple Control Transmission Protocol* (SCTP), *MTP2 User Adaptation* (M2UA), and *MTP3 User Adaptation* (M3UA).

Simple Control Transmission Protocol (SCTP)

A number of limitations within TCP impede the performance of SS7 messaging. The nature of TCP and how messages are prioritized requires unnecessary delays in processing TCP packets. To ensure that a complete packet is received, the push function of TCP must be used.

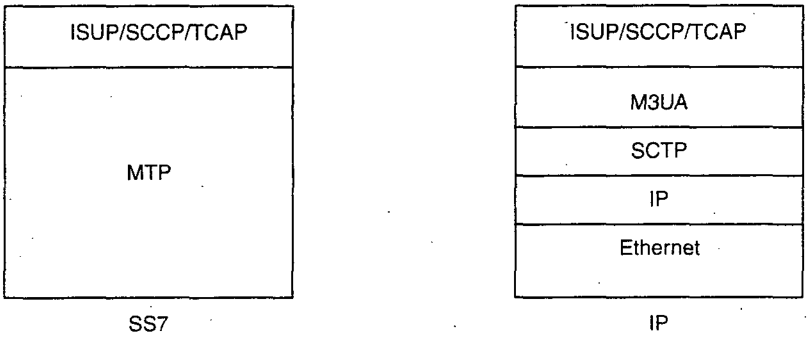


Figure 11.11 The protocol stack used for SS7 transport in an IP telephony network involves quite a few more protocols than in a conventional SS7 network, as shown in comparison here.

These limitations were taken into consideration when SCTP was developed. Initially, SCTP was to use the services of the *User Datagram Protocol* (UDP) for delivery. This was soon changed, however, when it was determined that UDP was not reliable enough a transport and in itself created problems. SCTP was then changed to a peer protocol to TCP and UDP.

SCTP is a connection-oriented protocol providing a number of functions. Besides segmentation and reassembly, SCTP also provides path management. When SCTP determines that a particular path to an endpoint is unreachable, SCTP reports to the application (ISUP, TCAP, and so on) the change in the state of the path. (Figure 11.11)

Likewise, if for some reason a number of retransmissions occur for any given session, SCTP considers the destination as unreachable and reports the fault to the application. A counter is used by SCTP to determine the threshold of retransmissions allowed. Periodic heartbeats are sent to all established sessions to ensure their reachability.

SCTP is capable of releasing a session gracefully in the event of a fault, reporting the cause of the release to the application using cause codes. This helps in troubleshooting network trouble. Both IPv4 addressing (32-bit) and IPv6 (128-bit) are supported.

MTP3 User Adaptation Layer (M3UA)

As shown in Figure 11.12, M3UA uses the services of SCTP for delivery and provides services to the application level (ISUP, TCAP, and so on). When compared with conventional SS7, the combination of SCTP and M3UA is equivalent to the services provided by MTP.

Although Figure 11.12 shows IP and Ethernet, both SCTP and M3UA are designed to be transport independent. ATM or Frame Relay can be used just as easily. The purpose of M3UA is to provide the same services as MTP3, with some exceptions.

Conventional SS7 has a boundary limitation of 256 octets (MTP3 limitation). This limitation does not exist with M3UA. However, although M3UA does not have the same

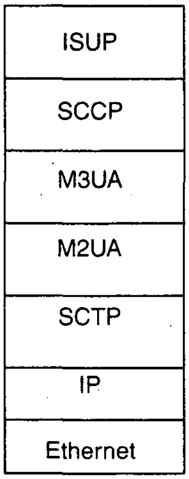


Figure 11.12 The full protocol stack for ISUP or TCAP transport using IETF standards.

boundary limitation, SGs must still follow the 256-octet boundary to prevent the fragmentation of messages being delivered to the PSTN.

Another role of M3UA is the mapping of point codes to IP addresses. As discussed earlier, the routing is based on four parameters: DPC, OPC, *service indicator octet* (SIO), and the CIC range (ISUP messages). For TCAP, the routing is based on DPC, OPC, SIO, and SSN.

MTP2 User Adaptation Layer (M2UA)

M2UA provides the same services as MTP2, including link management. M2UA maintains link states and has the capability to perform changeovers and link management functions related to processor outages.

One difference between MTP2 and M2UA is the absence of forward and backward sequence numbers. Instead, SCTP provides *transmit sequence numbers* (TSNs) in their place. Figure 11.12 shows the protocol stack including M3UA.

When M3UA is added, there are several layers of protocol, yet the use of IP is still far more efficient than MTP over TDM circuits.

Appendix A

ITU-TSS SS7 Publications

The following listing includes all *Signaling System 7* (SS7) standards published by the *International Telecommunication Union—Telecommunication Standardization Sector* (ITU-TSS) as of 1994. These standards define the protocol and the network at the international plane. The standards in the U.S. and in all other countries are based on these ITU standards.

Message Transfer Part (MTP)

Q.701—Functional Description of the Message Transfer Part

Q.702—Signaling Data Link

Q.703—Signaling Link

Q.704—Signaling Network Functions and Messages

Q.705—Signaling Network Structure

Q.706—MTP Signaling Performance

Q.707—Testing and Maintenance

Q.708—Numbering of International Signaling Point Codes

Q.709—Hypothetical Signaling Reference Connection

Q.710—Simplified MTP Version for Small Systems

Signaling Connection Control Part (SCCP)

Q.711—Functional Description of SCCP

Q.712—Definition and Functions of SCCP Messages

Q.713—SCCP Formats and Codes

Q.714—SCCP Procedures

Q.716—SCCP Performance

Telephone User Part (TUP)

Q.721—Functional Description of TUP

Q.722—General Function of Telephone Messages and Signals

Q.723—Formats and Codes

Q.724—Signaling Procedures

Q.725—Signaling Performance in the Telephone Applications

ISDN Supplementary Services

Q.730—ISDN Supplementary Services

Q.731—Stage 3 Description for Number ID Supplementary Services Using SS7

Q.733—Stage 3 Description for Call Completion Supplementary Services

Data User Part (DUP)

Q.741—SS7 Data User Part

ISDN User Part (ISUP)

Q.761—Functional Description of ISUP

Q.762—General Function of Messages and Signals

Q.763—Formats and Codes

Q.764—Signaling Procedures

Q.766—Performance Objectives in the ISDN Application

Q.767—Application of the ISUP for International ISDN Interconnections

Transaction Capabilities Application Part (TCAP)

Q.771—Functional Description of TCAP

Q.772—Transaction Capabilities Information Element Definitions

Q.773—TCAP Formats and Encoding

Q.774—TCAP Procedures

Q.775—SS7 Test Specification General Description

Test Specification

Q.780—SS7 Test Specification—General Description

Q.781—MTP Level 2 Test Specification

Q.782—MTP Level 3 Test Specification

Q.783—TUP Test Specification

Q.784—ISUP Basic Call Test Specification

Q.785—ISUP Protocol Test Specification for Supplementary Services

Monitoring and Measurements

Q.791—Monitoring and Measurements for SS7 Networks

Operations, Administration, and Maintenance (OAM)

Q.795—Operations, Maintenance, and Administration Part

Appendix

B

American National Standards Institute (ANSI)

The following listing is of *Signaling System 7* (SS7) standards published by the ANSI. These standards define the protocol and the network at the national plane.

ANSI T1.110 Telecommunications Signaling System No. 7 (SS7)—General Information

ANSI T1.111 Telecommunications Signaling System No. 7 (SS7)—Functional Description of the Signaling System Message Transfer Part (MTP)

ANSI T1.112 Telecommunications Signaling System No. 7 (SS7)—Signaling Connection Control Part (SCCP)

ANSI T1.113 Telecommunications Signaling System No. 7 (SS7)—Integrated Services Digital Network (ISDN) User Part (ISUP)

ANSI T1.113 Telecommunications Signaling System No. 7 (SS7)—Integrated Services Digital Network (ISDN) User Part (NxDS0 Multirate Connection) (supplement to ANSI T1.113-1992)

ANSI 1.114 Telecommunications Signaling System No. 7 (SS7)—Transaction Capabilities Application Part (TCAP)

ANSI T1.115 Telecommunications—Monitoring and Measurements for Signaling System No. 7 (SS7) Networks

ANSI T1.116 Telecommunications Signaling System No. 7 (SS7)—Operations, Maintenance, and Administration Part (OMAP)

ANSI T1.118 Telecommunications Signaling System No. 7 (SS7)—Intermediate Signaling Network Identification (ISNI)

ANSI T1.226 Telecommunications—Operations, Administration, Maintenance, and Provisioning (OAMP)—Management of Functions for Signaling System No. 7 (SS7) Network Interconnections

ANSI T1.609 Telecommunications—Interworking Between the ISDN User-to-Network Interwork Interface Protocol and the Signaling System No. 7 ISDN User Part (ISUP)

ANSI T1.611 Telecommunications—Signaling System No. 7 (SS7)—Supplementary Services for Non-ISDN Subscribers

ANSI T1.631 Telecommunications—Signaling System No. 7 (SS7)—High Probability of Completion (HPC) Network Capability

Bell Communications Research (Bellcore)

The following publications are Bellcore publications, which are directly related to the SS7 network and its protocol. Many other related publications that define the requirements of various network components are not listed here.

TR-TSY-000024 Service Switching Points (SSPs) Generic Requirements

TR-NWT-000082 Signaling Transfer Point (STP) Generic Requirements

TR-NWT-000246 Bell Communications Research Specification of Signaling System No. 7 (SS7)

TA-TSY-000372 CCS Switching System Maintenance Requirements

TR-NWT-000394 Switching System Generic Requirements for Interexchange Carrier Interconnection Using the Integrated Services Digital Network User Part (ISDNUP)

TR-TSV-000905 Common Channel Signaling (CCS) Network Interface Specification

Appendix C

Cause Codes

The following are the cause codes used in *Release* (REL) messages. These cause codes are typically dependent on implementation, but fairly self-explanatory. They are included in this appendix as a quick reference.

Cause codes are divided into various classes or groups. The class designation is found in bits 5 through 7. The decimal value for each cause code is shown in parenthesis.

ITU-T Standard (Coding Standard 00)

Class 000 and 001, Normal Event

0	0	0	0	0	0	1	(1)	Unallocated (unassigned) number
0	0	0	0	0	1	0	(2)	No route to specified transit network
0	0	0	0	0	1	1	(3)	No route to destination
0	0	0	0	1	0	0	(4)	Send special information tone
0	0	0	0	1	0	1	(5)	Misdialed trunk prefix (no procedure specified for U.S. networks)
0	0	0	1	0	0	0	(8)	Preemption
0	0	0	1	0	0	1	(9)	Preemption—circuit reserved for reuse
0	0	1	0	0	0	0	(16)	Normal clearing
0	0	1	0	0	0	1	(17)	User busy
0	0	1	0	0	1	0	(18)	No user responding
0	0	1	0	0	1	1	(19)	No answer from user (user alerted)
0	0	1	0	1	0	0	(20)	Subscriber absent
0	0	1	0	1	0	1	(21)	Call rejected
0	0	1	0	1	1	0	(22)	Number changed
0	0	1	0	1	1	1	(23)	Redirect to new destination
0	0	1	1	0	1	1	(27)	Destination out of service
0	0	1	1	1	0	0	(28)	Invalid number format (address incomplete)
0	0	1	1	1	0	1	(29)	Facility rejected
0	0	1	1	1	1	1	(31)	Normal—unspecified (default)

Class 010, Resource Unavailable

0	1	0	0	0	1	0	(34)	No circuit/channel available
0	1	0	0	1	1	0	(38)	Network out of order
0	1	0	1	0	0	1	(41)	Temporary failure
0	1	0	1	0	1	0	(42)	Switching equipment congestion
0	1	0	1	0	1	1	(43)	Access information discarded
0	1	0	1	1	0	0	(44)	Requested circuit/channel not available
0	1	0	1	1	1	0	(46)	Precedence call blocked
0	1	0	1	1	1	1	(47)	Resource unavailable—unspecified (default)

Class 011, Service or Option Unavailable

0	1	1	0	0	1	0	(50)	Requested facility not subscribed
0	1	1	0	1	0	1	(53)	Outgoing calls barred within CUG (No procedure specified for U.S. networks)
0	1	1	0	1	1	1	(55)	Incoming calls barred within CUG (No procedure specified for U.S. networks)
0	1	1	1	0	0	1	(57)	Bearer capability not authorized
0	1	1	1	0	1	0	(58)	Bearer capability not presently available
0	1	1	1	1	1	0	(62)	Inconsistency in designated outgoing access information and subscriber class
0	1	1	1	1	1	1	(63)	Service or option not available—unspecified (default)

Class 100, Service or Option Not Implemented

1	0	0	0	0	0	1	(65)	Bearer capability not implemented
1	0	0	0	1	0	1	(69)	Requested facility not implemented
1	0	0	0	1	1	0	(70)	Only restricted digital information bearer capability is available
1	0	0	1	1	1	1	(79)	Service or option not implemented—unspecified (default)

Class 101, Invalid Message (Parameter Out of Range)

1	0	1	0	1	1	1	(87)	User not member of CUG (no procedure specified for U.S. networks)
1	0	1	1	0	0	0	(88)	Incompatible destination
1	0	1	1	0	1	0	(90)	Nonexistent CUG (no procedure specified for U.S. networks)
1	0	1	1	0	1	1	(91)	Invalid transit network selection
1	0	1	1	1	1	1	(95)	Invalid message, unspecified (default)

Class 110, Protocol Error

1	1	0	0	0	0	1	(97)	Message type nonexistent or not implemented
1	1	0	0	0	1	1	(99)	Information element/parameter nonexistent or not implemented—discarded
1	1	0	0	1	1	0	(102)	Recovery on timer expiry
1	1	0	0	1	1	1	(103)	Parameter nonexistent or not implemented—passed on
1	1	0	1	1	1	0	(110)	Message with unrecognized parameter, discarded
1	1	0	1	1	1	1	(111)	Protocol error, unspecified (default)

Class 111, Interworking Class

1	1	1	1	1	1	1	(127)	Interworking, unspecified (default)

ANSI Standard (Coding Standard Value 10)

Class 000 and 001, Normal Event

0	0	1	0	1	1	1	(23)	Unallocated destination number
0	0	1	1	0	0	0	(24)	Unknown business group
0	0	1	1	0	0	1	(25)	Exchange routing error
0	0	1	1	0	1	0	(26)	Misrouted call to a ported number
0	0	1	1	0	1	1	(27)	Number portability (NP) query on release (QoR)—number not found (no procedures for U.S. networks)

Class 010, Resource Unavailable

0	1	0	1	1	0	1	(45)	Preemption (used by T1.113-1988 and T1.619-1992)
0	1	0	1	1	1	0	(46)	Precedence call blocked (used by T1.113-1988 and T1.619-1992)

Class 011, Service or Option Not Available

0	1	1	0	0	1	1	(51)	Call type incompatible with service request
0	1	1	0	1	1	0	(54)	Call blocked due to group restrictions

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Travis Russell is a Sales Manager with the Network Systems Division at Tekelec in North Carolina. He has been a field engineer in the telecommunications business for over 20 years and has lectured on subjects ranging from switching technologies to data communications at colleges and universities, as well as industry seminars here in the United States and around the world. In addition to this book, he is also the author of *Telecommunications Protocols* and *The Telecommunications Pocket Reference*, and is coauthor of *CDPD* (all published by McGraw-Hill). He resides in Clayton, North Carolina, and can be reached via e-mail at travisruss@aol.com.

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Originally designed for analog telephone networks, SS7 has continually undergone changes to accommodate the ever-evolving world of telecom. Today, SS7 is used for data, voice, video, audio, and voice-over IP networks – and no other resource even comes close to providing such a complete understanding of the signaling network, its architecture, and protocols used to communicate through it like Travis Russell's *Signaling System #7*.

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