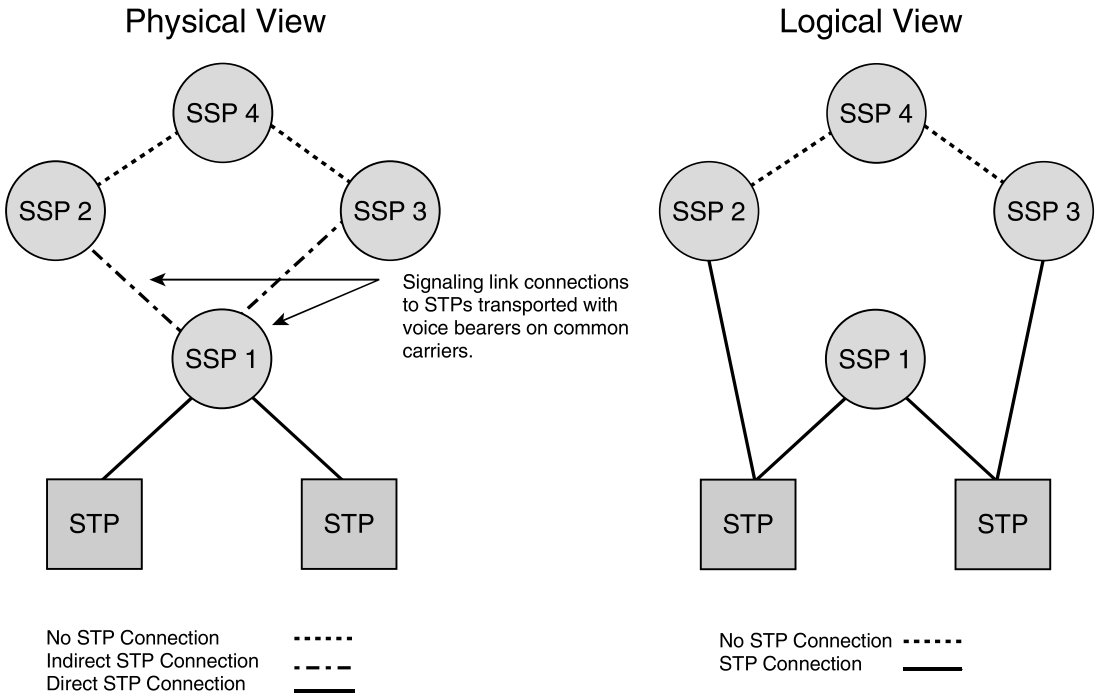
where it also has direct trunks. It has no STP connection at all. SSP 2 and SSP 3 are connected to the STP pair via nailed-up connections at SSP 1.

Figure 4-17 *Example of Direct and Indirect SSP Interconnections to STPs*



Normally within networks that do not use STPs, circuit-related (call-related) signaling takes the same path through the network as user traffic because there is no physical need to take a different route. This mode of operation is called *associated signaling* and is prevalent outside North America. Referring back to Figure 4-14, both the user traffic and the signaling take the same path between SSP B and SSP C.

Because standalone STPs are used to form the SS7 backbone within North America, and standalone STPs do not support user traffic switching, the SSP's signaling mode is usually quasi-associated, as illustrated between SSP A and SSP B in Figure 4-14.

In certain circumstances, the SSP uses associated signaling within North America. A great deal of signaling traffic might exist between two SSPs, so it might make more sense to place a signaling link directly between them rather than to force all signaling through an STP.

SS7 Protocol Overview

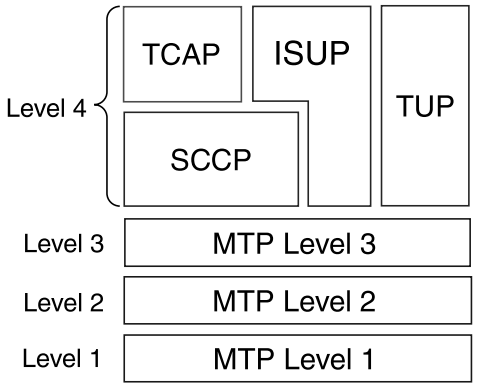
The number of possible protocol stack combinations is growing. It depends on whether SS7 is used for cellular-specific services or intelligent network services, whether transportation is over IP or is controlling broadband ATM networks instead of time-division multiplexing

(TDM) networks, and so forth. This requires coining a new term—traditional SS7—to refer to a stack consisting of the protocols widely deployed from the 1980s to the present:

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

Figure 4-18 shows a common introductory SS7 stack.

Figure 4-18 *Introductory SS7 Protocol Stack*



Such a stack uses TDM for transport. This book focuses on traditional SS7 because that is what is implemented. Newer implementations are beginning to appear that use different transport means such as IP and that have associated new protocols to deal with the revised transport.

The SS7 physical layer is called MTP level 1 (MTP1), the data link layer is called MTP level 2 (MTP2), and the network layer is called MTP level 3 (MTP3). Collectively they are called the *Message Transfer Part (MTP)*. The MTP protocol is SS7's native means of packet transport. In recent years there has been an interest in the facility to transport SS7 signaling over IP instead of using SS7's native MTP. This effort has largely been carried out by the Internet Engineering Task Force (IETF) SigTran (Signaling Transport) working group. The protocols derived by the SigTran working group so far are outside the scope of this introductory chapter on SS7. However, full details of SigTran can be found in Chapter 14, "SS7 in the Converged World."

TUP and ISUP both perform the signaling required to set up and tear down telephone calls. As such, both are circuit-related signaling protocols. TUP was the first call control protocol specified. It could support only plain old telephone service (POTS) calls. Most countries

are replacing TUP with ISUP. Both North America and Japan bypassed TUP and went straight from earlier signaling systems to ISUP. ISUP supports both POTS and ISDN calls. It also has more flexibility and features than TUP.

With reference to the Open System Interconnection (OSI) seven-layer reference model, SS7 uses a four-level protocol stack. OSI Layer 1 through 3 services are provided by the MTP together with the SCCP. The SS7 architecture currently has no protocols that map into OSI Layers 4 through 6. TUP, ISUP, and TCAP are considered as corresponding to OSI Layer 7 [111]. SS7 and the OSI model were created at about the same time. For this reason, they use some differing terminology.

SS7 uses the term *levels* when referring to its architecture. The term *levels* should not be confused with OSI layers, because they do not directly correspond to each other. *Levels* was a term introduced to help in the discussion and presentation of the SS7 protocol stack. Levels 1, 2, and 3 correspond to MTP 1, 2, and 3, respectively. Level 4 refers to an MTP *user*. The term *user* refers to any protocol that directly uses the transport capability provided by the MTP—namely, TUP, ISUP, and SCCP in traditional SS7. The four-level terminology originated back when SS7 had only a call control protocol (TUP) and the MTP, before SCCP and TCAP were added.

The following sections provide a brief outline of protocols found in the introductory SS7 protocol stack, as illustrated in Figure 4-18.

MTP

MTP levels 1 through 3 are collectively referred to as the MTP. The MTP comprises the functions to transport information from one SP to another.

The MTP transfers the signaling message, in the correct sequence, without loss or duplication, between the SPs that make up the SS7 network. The MTP provides reliable transfer and delivery of signaling messages. The MTP was originally designed to transfer circuit-related signaling because no noncircuit-related protocol was defined at the time.

The recommendations refer to MTP1, MTP2, and MTP3 as the physical layer, data link layer, and network layer, respectively. The following sections discuss MTP2 and MTP3. (MTP1 isn't discussed because it refers to the physical network.) For information on the physical aspects of the Public Switched Telephone Network (PSTN), see Chapter 5, "The Public Switched Telephone Network (PSTN)."

MTP2

Signaling links are provided by the combination of MTP1 and MTP2. MTP2 ensures reliable transfer of signaling messages. It encapsulates signaling messages into variable-length SS7 packets. SS7 packets are called signal units (SUs). MTP2 provides delineation of SUs, alignment of SUs, signaling link error monitoring, error correction by retransmission, and flow control. The MTP2 protocol is specific to narrowband links (56 or 64 kbps).

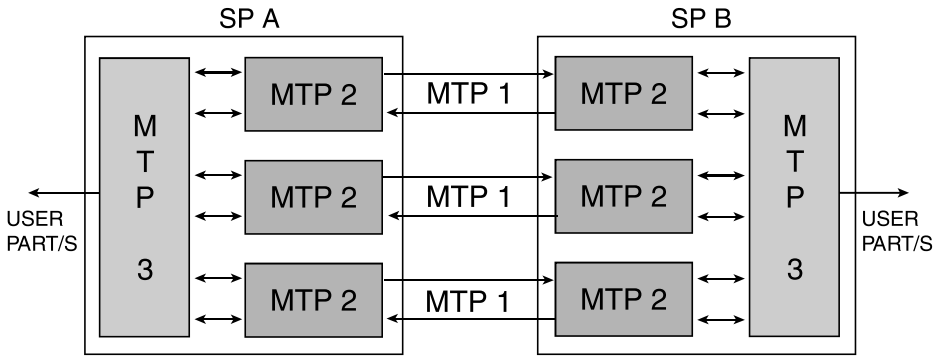
MTP3

MTP3 performs two functions:

- **Signaling Message Handling (SMH)**—Delivers incoming messages to their intended User Part and routes outgoing messages toward their destination. MTP3 uses the PC to identify the correct node for message delivery. Each message has both an Origination Point Code (OPC) and a DPC. The OPC is inserted into messages at the MTP3 level to identify the SP that originated the message. The DPC is inserted to identify the address of the destination SP. Routing tables within an SS7 node are used to route messages.
- **Signaling Network Management (SNM)**—Monitors linksets and routesets, providing status to network nodes so that traffic can be rerouted when necessary. SNM also provides procedures to take corrective action when failures occur, providing a self-healing mechanism for the SS7 network.

Figure 4-19 shows the relationship between levels 1, 2, and 3.

Figure 4-19 *A Single MTP3 Controls Many MTP2s, Each of Which Is Connected to a Single MTP1*



TUP and ISUP

TUP and ISUP sit on top of MTP to provide circuit-related signaling to set up, maintain, and tear down calls. TUP has been replaced in most countries because it supports only POTS calls. Its successor, ISUP, supports both POTS and ISDN calls as well as a host of other features and added flexibility. Both TUP and ISUP are used to perform interswitch call signaling. ISUP also has inherent support for supplementary services, such as automatic callback, calling line identification, and so on.

SCCP

The combination of the MTP and the SCCP is called the *Network Service Part (NSP)* in the specifications (but outside the specifications, this term is seldom used).

The addition of the SCCP provides a more flexible means of routing and provides mechanisms to transfer data over the SS7 network. Such additional features are used to support noncircuit-related signaling, which is mostly used to interact with databases (SCPs). It is also used to connect the radio-related components in cellular networks and for inter-SSP communication supporting CLASS services. SCCP also provides application management functions. Applications are mostly SCP database driven and are called subsystems. For example, in cellular networks, SCCP transfers queries and responses between the Visitor Location Register (VLR) and Home Location Register (HLR) databases. Such transfers take place for a number of reasons. The primary reason is to update the subscriber's HLR with the current VLR serving area so that incoming calls can be delivered.

Enhanced routing is called global title (GT) routing. It keeps SPs from having overly large routing tables that would be difficult to provision and maintain. A GT is a directory number that serves as an alias for a physical network address. A physical address consists of a point code and an application reference called a subsystem number (SSN). GT routing allows SPs to use alias addressing to save them from having to maintain overly large physical address tables. Centralized STPs are then used to convert the GT address into a physical address; this process is called Global Title Translation (GTT). This provides the mapping of traditional telephony addresses (phone numbers) to SS7 addresses (PC and/or SSN) for enhanced services. GTT is typically performed at STPs.

NOTE It is important not to confuse the mapping of telephony numbers using GTT with the translation of telephony numbers done during normal call setup. Voice switches internally map telephony addresses to SS7 addresses during normal call processing using number translation tables. This process does not use GTT. GTT is used only for noncircuit-related information, such as network supplementary services (Calling Name Delivery) or database services (toll-free).

In addition to mapping telephony addresses to SS7 addresses, SCCP provides a set of subsystem management functions to monitor and respond to the condition of subsystems. These management functions are discussed further, along with the other aspects of SCCP, in Chapter 9, "Signaling Connection Control Part (SCCP)."

TCAP

TCAP allows applications (called subsystems) to communicate with each other (over the SS7 network) using agreed-upon data elements. These data elements are called *components*. Components can be viewed as instructions sent between applications. For example, when a subscriber changes VLR location in a global system for mobile communication (GSM) cellular network, his or her HLR is updated with the new VLR location by means

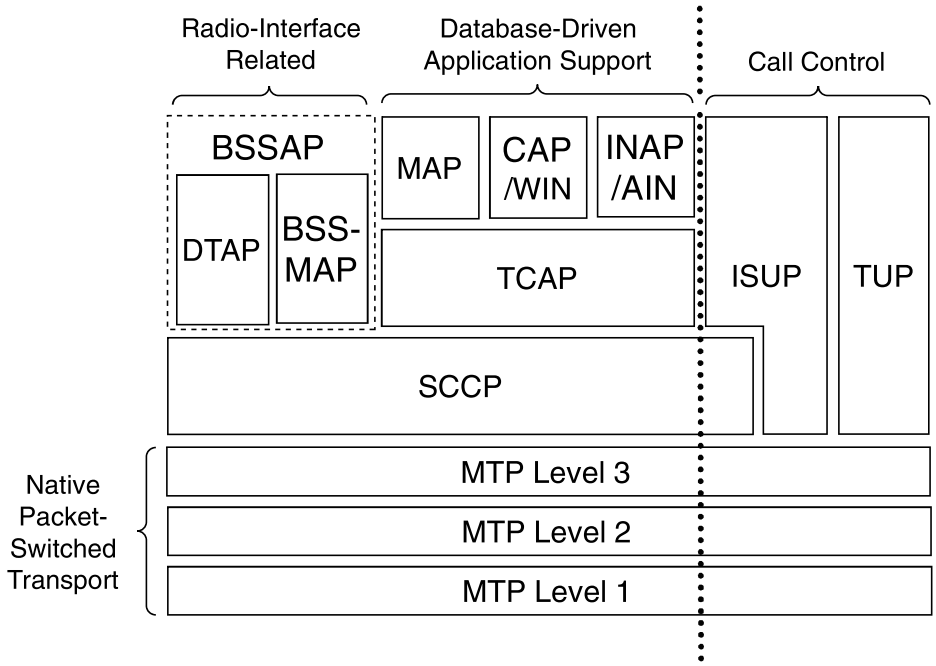
of an UpdateLocation component. TCAP also provides transaction management, allowing multiple messages to be associated with a particular communications exchange, known as a transaction.

There are a number of subsystems; the most common are

- Toll-free (E800)
- Advanced Intelligent Network (AIN)
- Intelligent Network Application Protocol (INAP)
- Customizable Applications for Mobile Enhanced Logic (CAMEL)
- Mobile Application Part (MAP)

Figure 4-20 illustrates these subsystems as well as another protocol that uses SCCP, the Base Station Subsystem Application Part. It is used to control the radio-related component in cellular networks.

Figure 4-20 *Some Protocols That Might Exist on Top of the SCCP, Depending on the Application*



It is highly unlikely that a protocol such as the one shown in Figure 4-20 would exist at any one SP. Instead, protocol stacks vary as required by SP type. For example, because an STP is a routing device, it has only MTP1, MTP2, MTP3, and SCCP. A fixed-line switch without IN support might have only MTP1, MTP2, MTP3, and ISUP, and so forth. A diagram showing how the SS7 protocol stack varies by SP can be found in Chapter 13.

Summary

SS7 is a data communications network that acts as the nervous system to bring the components of telecommunications networks to life. It acts as a platform for various services described throughout this book. SS7 nodes are called signaling points (SPs), of which there are three types:

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

SSPs provide the SS7 functionality of a switch. STPs may be either standalone or integrated STPs (SSP and STP) and are used to transfer signaling messages. SCPs interface the SS7 network to query telecommunication databases, allowing service logic and additional routing information to be obtained to execute services.

SPs are connected to each other using signaling links. Signaling links are logically grouped into a linkset. Links may be referenced as A through F links, depending on where they are in the network.

Signaling is transferred using the packet-switching facilities afforded by SS7. These packets are called signal units (SUs). The Message Transfer Part (MTP) and the Signaling Connection Control Part (SCCP) provide the transfer protocols. MTP is used to reliably transport messages between nodes, and SCCP is used for noncircuit-related signaling (typically, transactions with SCPs). The ISDN User Part (ISUP) is used to set up and tear down both ordinary (analog subscriber) and ISDN calls. The Transaction Capabilities Application Part (TCAP) allows applications to communicate with each other using agreed-upon data components and manages transactions.

CHAPTER 5

The Public Switched Telephone Network (PSTN)

The term Public Switched Telephone Network (PSTN) describes the various equipment and interconnecting facilities that provide phone service to the public. The network continues to evolve with the introduction of new technologies. The PSTN began in the United States in 1878 with a manual mechanical switchboard that connected different parties and allowed them to carry on a conversation. Today, the PSTN is a network of computers and other electronic equipment that converts speech into digital data and provides a multitude of sophisticated phone features, data services, and mobile wireless access.

TIP PSTN voice facilities transport speech or voice-band data (such as fax/modems and digital data), which is data that has been modulated to voice frequencies.

At the core of the PSTN are digital switches. The term "switch" describes the ability to cross-connect a phone line with many other phone lines and switching from one connection to another. The PSTN is well known for providing reliable communications to its subscribers. The phrase "five nines reliability," representing network availability of 99.999 percent for PSTN equipment, has become ubiquitous within the telecommunications industry.

This chapter provides a fundamental view of how the PSTN works, particularly in the areas of signaling and digital switching. SS7 provides control signaling for the PSTN, so you should understand the PSTN infrastructure to fully appreciate how it affects signaling and switching. This chapter is divided into the following sections:

- Network Topology
- PSTN Hierarchy
- Access and Transmission Facilities
- Network Timing

- The Central Office
- Integration of SS7 into the PSTN
- Evolving the PSTN to the Next Generation

We conclude with a summary of the PTSN infrastructure and its continuing evolution.

Network Topology

The topology of a network describes the various network nodes and how they interconnect. Regulatory policies play a major role in exactly how voice network topologies are defined in each country, but general similarities exist. While topologies in competitive markets represent an interconnection of networks owned by different service providers, monopolistic markets are generally an interconnection of switches owned by the same operator.

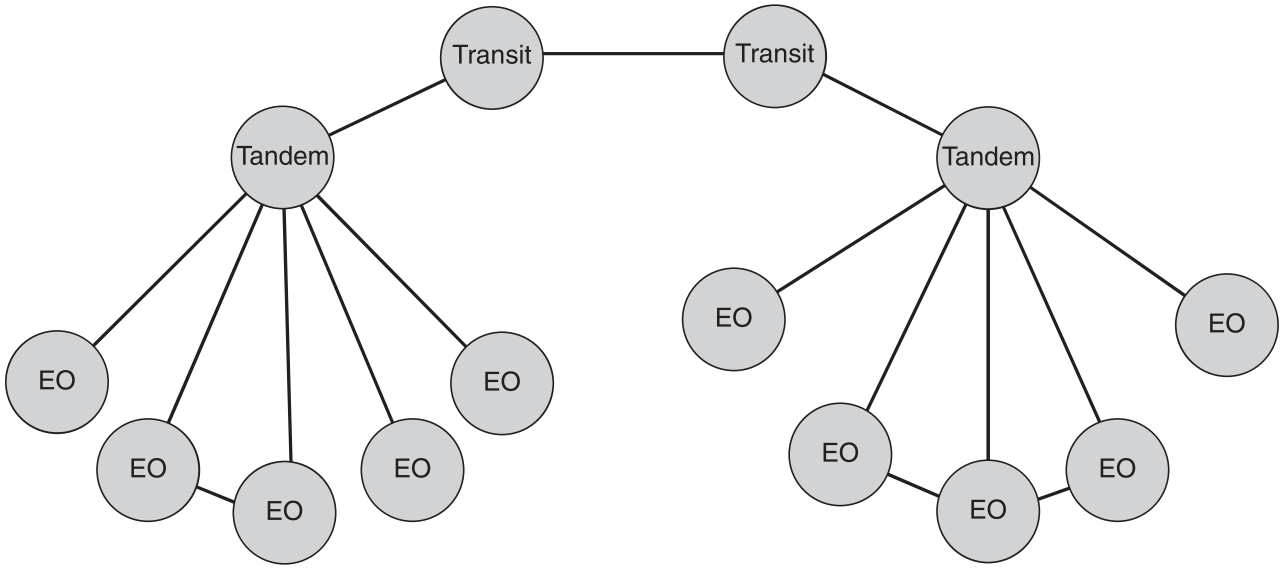
Depending on geographical region, PSTN nodes are sometimes referred to by different names. The three node types we discuss in this chapter include:

- **End Office (EO)**—Also called a Local Exchange. The End Office provides network access for the subscriber. It is located at the bottom of the network hierarchy.
- **Tandem**—Connects EOs together, providing an aggregation point for traffic between them. In some cases, the Tandem node provides the EO access to the next hierarchical level of the network.
- **Transit**—Provides an interface to another hierarchical network level. Transit switches are generally used to aggregate traffic that is carried across long geographical distances.

There are two primary methods of connecting switching nodes. The first approach is a mesh topology, in which all nodes are interconnected. This approach does not scale well when you must connect a large number of nodes. You must connect each new node to every existing node. This approach does have its merits, however; it simplifies routing traffic between nodes and avoids bottlenecks by involving only those switches that are in direct communication with each other. The second approach is a hierarchical tree in which nodes are aggregated as the hierarchy traverses from the subscriber access points to the top of the tree. PSTN networks use a combination of these two methods, which are largely driven by cost and the traffic patterns between exchanges.

Figure 5-1 shows a generic PSTN hierarchy, in which End Offices are connected locally and through tandem switches. Transit switches provide further aggregation points for connecting multiple tandems between different networks. While actual network topologies vary, most follow some variation of this basic pattern.

Figure 5-1 *Generic PSTN Hierarchies*



PSTN Hierarchy

The PSTN hierarchy is implemented differently in the United States and the United Kingdom. The following sections provide an overview of the PSTN hierarchy and its related terminology in each of these countries.

PSTN Hierarchy in the United States

In the United States, the PSTN is generally divided into three categories:

- Local Exchange Networks
- InterExchange Networks
- International Networks

Local Exchange Carriers (LECs) operate Local Exchange networks, while InterExchange Carriers (IXCs) operate InterExchange and International networks.

The PSTN hierarchy in the United States is also influenced by market deregulation, which has allowed service providers to compete for business and by the divestiture of Bell.

Local Exchange Network

The Local Exchange network consists of the digital switching nodes (EOs) that provide network access to the subscriber. The Local Exchange terminates both lines and trunks, providing the subscriber access to the PSTN.

A Tandem Office often connects End Offices within a local area, but they can also be connected directly. In the United States, Tandem Offices are usually designated as either Local Tandem (LT) or Access Tandem (AT). The primary purpose of a Local Tandem is to provide interconnection between End Offices in a localized geographic region. An Access Tandem provides interconnection between local End Offices and serves as a primary point of access for IXCs. Trunks are the facilities that connect all of the offices, thereby transporting internodal traffic.

InterExchange Network

The InterExchange network is comprised of digital switching nodes that provide the connection between Local Exchange networks. Because they are points of high traffic aggregation and they cover larger geographical distances, high-speed transports are typically used between transit switches. In the deregulated U.S. market, transit switches are usually referred to as *carrier switches*. In the U.S., IXCs access the Local Exchange network at designated points, referred to as a Point of Presence (POP). POPs can be connections at the Access Tandem, or direct connections to the End Office.

International Network

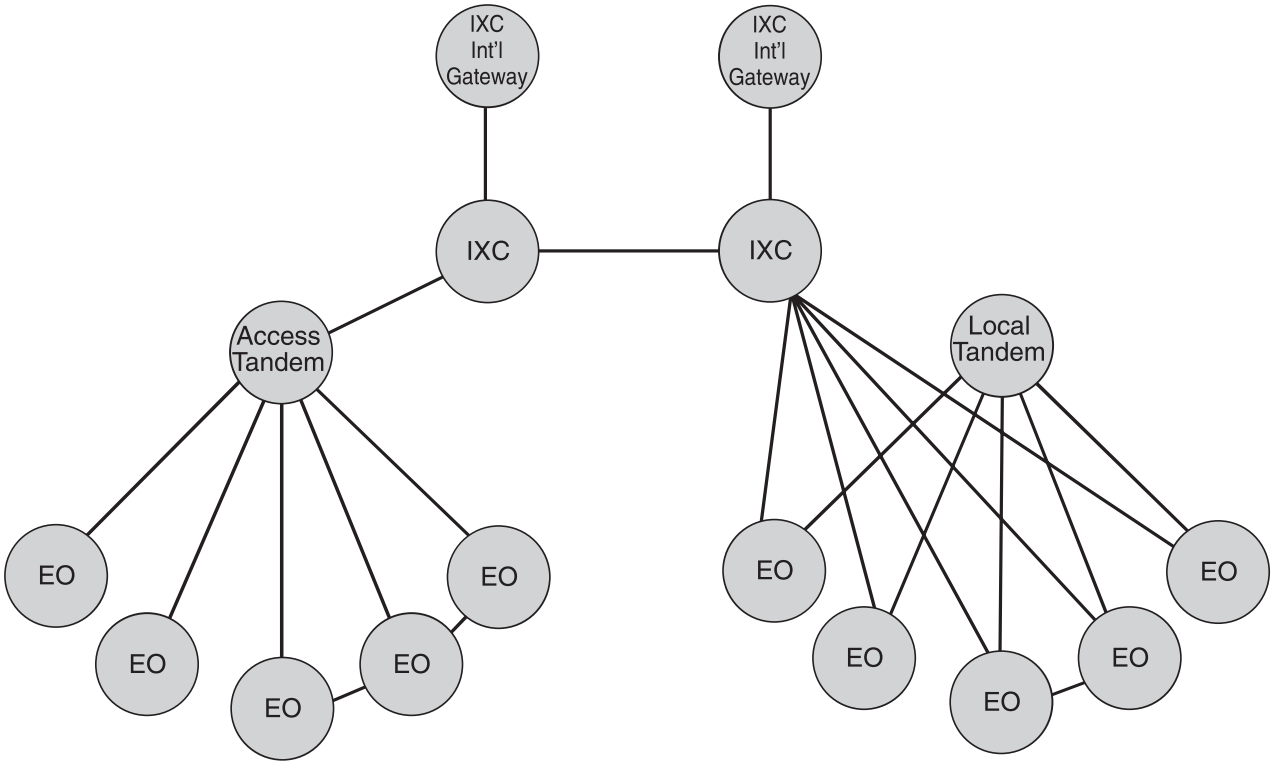
The International network consists of digital switching nodes, which are located in each country and act as international gateways to destinations outside of their respective countries. These gateways adhere to the ITU international standards to ensure interoperability between national networks. The international switch also performs the protocol conversions between national and international signaling. The gateway also performs PCM conversions between A-law and μ-law to produce compatible speech encoding between networks, when necessary.

Service Providers

Deregulation policies in the United States have allowed network operators to compete for business, first in the long-distance market (InterExchange and International) beginning in the mid 1980s, and later in the local market in the mid 1990s. As previously mentioned, LECs operate Local Exchange networks, while IXCs operate the long-distance networks. Figure 5-2 shows a typical arrangement of LEC-owned EOs and tandems interconnected to

IXC-owned transit switches. The IXC switches provide long-haul transport between Local Exchange networks, and international connections through International gateway switches.

Figure 5-2 *Generic U.S. Hierarchies*



Over the last several years, the terms ILEC and CLEC have emerged within the Local Exchange market to differentiate between the Incumbent LECs (ILECS) and the Competitive LECs (CLECS). ILECs are the incumbents, who own existing access lines to residences and corporate facilities; CLECs are new entrants into the Local Exchange market. Most of the ILECs in the United States came about with the divestiture of AT&T into the seven Regional Bell Operating Companies (RBOC). The remainder belonged to Independent Operating Companies (IOCs). Most of these post-divestiture companies have been significantly transformed today by mergers and acquisitions in the competitive market. New companies have experienced difficulty entering into the Local Exchange market, which is dominated by ILECs. The ILECs own the wire to the subscriber's home, often called the "last mile" wiring. Last mile wiring is expensive to install and gives the ILECs tremendous market leverage. The long-distance market has been easier for new entrants because it does not require an investment in last mile wiring.

Pre-Divestiture Bell System Hierarchy

Vestiges of terminology relating to network topology remain in use today from the North American Bell System's hierarchy, as it existed prior to divestiture in 1984. Telephone switching offices are often still referred to by *class*. For example, an EO is commonly called a class 5 office, and an AT is called a class 4 office. Before divestiture, each layer of the network hierarchy was assigned a class number.

Prior to divestiture, offices were categorized by class number, with class 1 being the highest office category and class 5 being the lowest (nearest to subscriber access). Aggregation of transit phone traffic moved from the class 5 office up through the class 1 office. Each class of traffic aggregation points contained a smaller number of offices. Table 5-1 lists the class categories and office types used in the Bell System Hierarchy.

Table 5-1 *Pre-Divestiture Class Categories and Office Types*

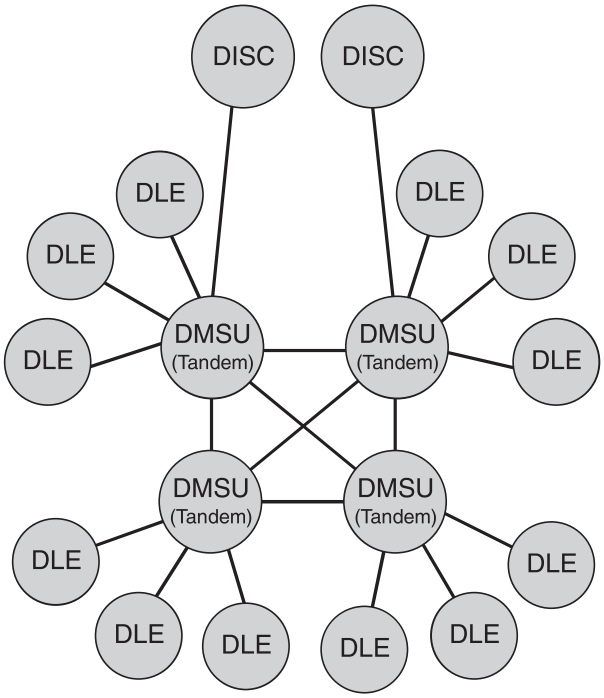
Class	Office Type
1	Regional Center
2	Sectional Center
3	Primary Center
4	Toll Center
5	End Office

Local calls remained within class 5 offices, while a cross-country call traversed the hierarchy up to a regional switching center. This system no longer exists, but we included it to give relevance to the class terminology, which the industry still uses often.

PSTN Hierarchy in the United Kingdom

Figure 5-3 shows the PSTN topology used in the United Kingdom. End Offices are referred to as Digital Local Exchanges (DLE). A fully meshed tandem network of Digital Main Switching Units (DMSU) connects the DLEs. Digital International Switching Centers (DISC) connect the DMSU tandem switches for international call connections.

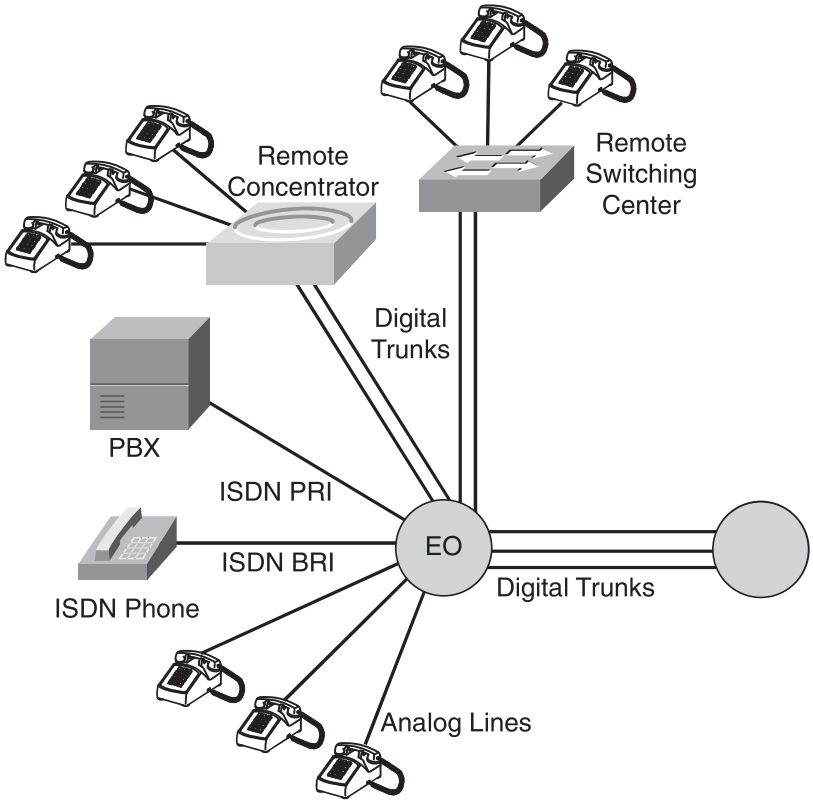
Figure 5-3 *U.K. PSTN Hierarchy*



Access and Transmission Facilities

Connections to PSTN switches can be divided into two basic categories: lines and trunks. Individual telephone lines connect subscribers to the Central Office (CO) by wire pairs, while trunks are used to interconnect PSTN switches. Trunks also provide access to corporate phone environments, which often use a Private Branch eXchange (PBX)—or in the case of some very large businesses, their own digital switch. Figure 5-4 illustrates a number of common interfaces to the Central Office.

Figure 5-4 *End Office Facility Interfaces*



Lines

Lines are used to connect the subscriber to the CO, providing the subscriber access into the PSTN. The following sections describe the facilities used for lines, and the access signaling between the subscriber and the CO.

- The Local Loop
- Analog Line Signaling
- Dialing
- Ringing and Answer
- Voice Encoding
- ISDN BRI

The Local Loop

The local loop consists of a pair of copper wires extending from the CO to a residence or business that connects to the phone, fax, modem, or other telephony device. The wire pair

consists of a tip wire and a ring wire. The terms *tip* and *ring* are vestiges of the manual switchboards that were used a number of years ago; they refer to the tip and ring of the actual switchboard plug operators used to connect calls. The local loop allows a subscriber to access the PSTN through its connection to the CO. The local loop terminates on the Main Distribution Frame (MDF) at the CO, or on a remote line concentrator.

Remote line concentrators, also referred to as Subscriber Line Multiplexers or Subscriber Line Concentrators, extend the line interface from the CO toward the subscribers, thereby reducing the amount of wire pairs back to the CO and converting the signal from analog to digital closer to the subscriber access point. In some cases, remote switching centers are used instead of remote concentrators.

Remote switching centers provide local switching between subtending lines without using the resources of the CO. *Remotes,* as they are often generically referred to, are typically used for subscribers who are located far away from the CO. While terminating the physical loop, remotes transport the digitized voice stream back to the CO over a trunk circuit, in digital form.

Analog Line Signaling

Currently, most phone lines are analog phone lines. They are referred to as analog lines because they use an analog signal over the local loop, between the phone and the CO. The analog signal carries two components that comprise the communication between the phone and the CO: the voice component, and the signaling component.

The signaling that takes place between the analog phone and the CO is called in-band signaling. In-band signaling is primitive when compared to the out-of-band signaling used in access methods such as ISDN; see the "ISDN BRI" section in this chapter for more information. DC current from the CO powers the local loop between the phone and the CO. The voltage levels vary between different countries, but an on-hook voltage of –48 to –54 volts is common in North America and a number of other geographic regions, including the United Kingdom.

TIP The actual line loop voltage varies, based on the distance and the charge level of the batteries connected to the loop at the CO. When the phone receiver is on-hook, the CO sees practically no current over the loop to the phone set. When the phone is off-hook, the resistance level changes, changing the current seen at the CO. The actual amount of loop current that triggers an on/off-hook signal also varies among different countries. In North America, a current flow of greater than 20 milliamps indicates an off-hook condition. When the CO has detected the off-hook condition, it provides a dial tone by connecting a tone generation circuit to the line.

Dialing

When a subscriber dials a number, the number is signaled to the CO as either a series of pulses based on the number dialed, or by Dual Tone Multi-Frequency (DTMF) signals. The DTMF signal is a combination of two tones that are generated at different frequencies. A total of seven frequencies are combined to provide unique DTMF signals for the 12 keys (three columns by four rows) on the standard phone keypad. Usually, the dialing plan of the CO determines when all digits have been collected.

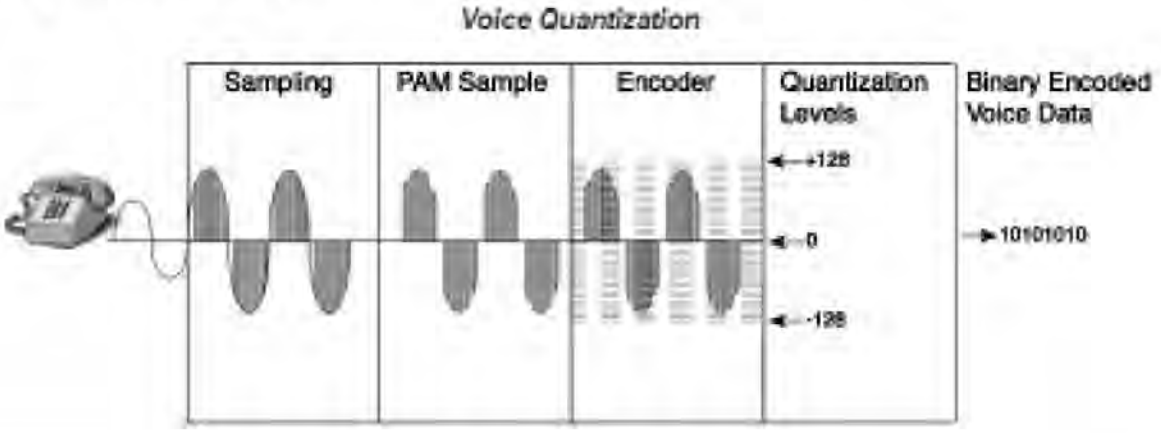
Ringing and Answer

To notify the called party of an incoming call, the CO sends AC ringing voltage over the local loop to the terminating line. The incoming voltage activates the ringing circuit within the phone to generate an audible ring signal. The CO also sends an audible ring-back tone over the originating local loop to indicate that the call is proceeding and the destination phone is ringing. When the destination phone is taken off-hook, the CO detects the change in loop current and stops generating the ringing voltage. This procedure is commonly referred to as *ring trip*. The off-hook signals the CO that the call has been answered; the conversation path is then completed between the two parties and other actions, such as billing, can be initiated, if necessary.

Voice Encoding

An analog voice signal must be encoded into digital information for transmission over the digital switching network. The conversion is completed using a codec (coder/decoder), which converts between analog and digital data. The ITU G.711 standard specifies the Pulse Coded Modulation (PCM) method used throughout most of the PSTN. An analog-to-digital converter samples the analog voice 8000 times per second and then assigns a quantization value based on 256 decision levels. The quantization value is then encoded into a binary number to represent the individual data point of the sample. Figure 5-5 illustrates the process of sampling and encoding the analog voice data.

Figure 5-5 *Voice Encoding Process*



Two variations of encoding schemes are used for the actual quantization values: A-law and μ-Law encoding. North America uses μ-Law encoding, and European countries use A-law encoding. When voice is transmitted from the digital switch over the analog loop, the digital voice data is decoded and converted back into an analog signal before transmitting over the loop.

The emergence of voice over IP (VoIP) has prompted the use of other voice-encoding standards, such as ITU G.723, G.726, and ITU G.729. These encoding methods use algorithms that produce more efficient and compressed data, making them more suitable for use in packet networks. Each encoding method involves trade-offs between bandwidth, processing power required for the encoding/decoding function, and voice quality. For example, G.711 encoding/decoding requires little processing and produces high quality speech, but consumes more bandwidth. In contrast, G.723.1 consumes little bandwidth, but requires more processing power and results in lower quality speech.

ISDN BRI

Although Integrated Services Digital Network (ISDN) deployment began in the 1980s, it has been a relatively slow-moving technology in terms of number of installations. ISDN moves the point of digital encoding to the customer premises. Combining ISDN on the access portion of the network with digital trunks on the core network provides total end-to-end digital connectivity. ISDN also provides out-of-band signaling over the local loop. ISDN access signaling coupled with SS7 signaling in the core network achieves end-to-end out-of-band signaling. ISDN access signaling is designed to complement SS7 signaling in the core network.

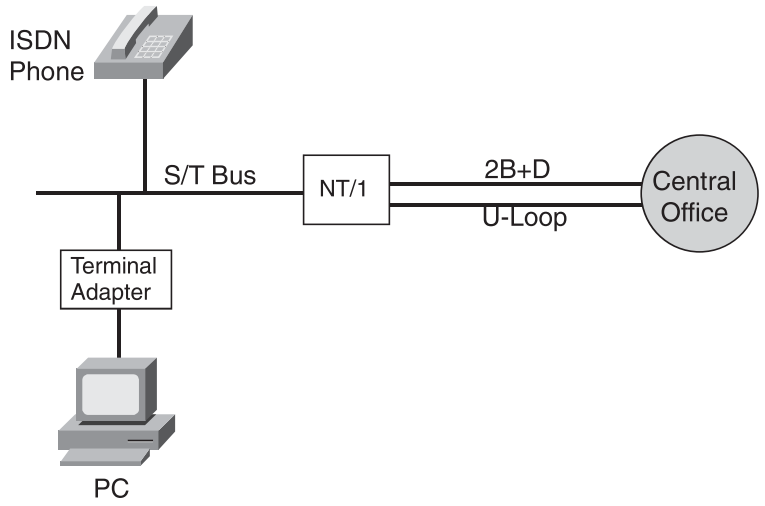
There are two ISDN interface types: Basic Rate Interface (BRI) for lines, and Primary Rate Interface (PRI) for trunks. BRI multiplexes two bearer (2B) channels and one signaling (D) channel over the local loop between the subscriber and the CO; this is commonly referred to as 2B+D. The two B channels each operate at 64 kb/s and can be used for voice or data communication. The D channel operates at 16 kb/s and is used for call control signaling for the two B channels. The D channel can also be used for very low speed data transmission. Within the context of ISDN reference points, the local loop is referred to as the U-loop. It uses different electrical characteristics than those of an analog loop.

Voice quantization is performed within the ISDN phone (or a Terminal Adapter, if an analog phone is used) and sent to a local bus: the S/T bus. The S/T bus is a four-wire bus that connects local ISDN devices at the customer premises to a Network Termination 1 (NT1) device. The NT1 provides the interface between the Customer Premises Equipment (CPE) and the U-loop.

TIP CPE refers to any of the ISDN-capable devices that are attached to the S/T bus.

The NT1 provides the proper termination for the local S/T bus to individual devices and multiplexes the digital information from the devices into the 2B+D format for transmission over the U-loop. Figure 5-6 illustrates the BRI interface to the CO. Only ISDN devices connect directly to the S/T bus. The PC uses an ISDN Terminal Adapter (TA) card to provide the proper interface to the bus.

Figure 5-6 *ISDN Basic Rate Interface*



The ISDN U-Loop terminates at the CO on a line card that is specifically designed to handle the 2B+D transmission format. The call control signaling messages from the D channel are designed to map to SS7 messages easily for outbound calls over SS7 signaled trunks.

TIP For U.S. networks, the Telcordia TR-444 (Generic Switching Systems Requirements Supporting ISDN access using the ISDN User Part) standard specifies the inter-working of ISDN and SS7.

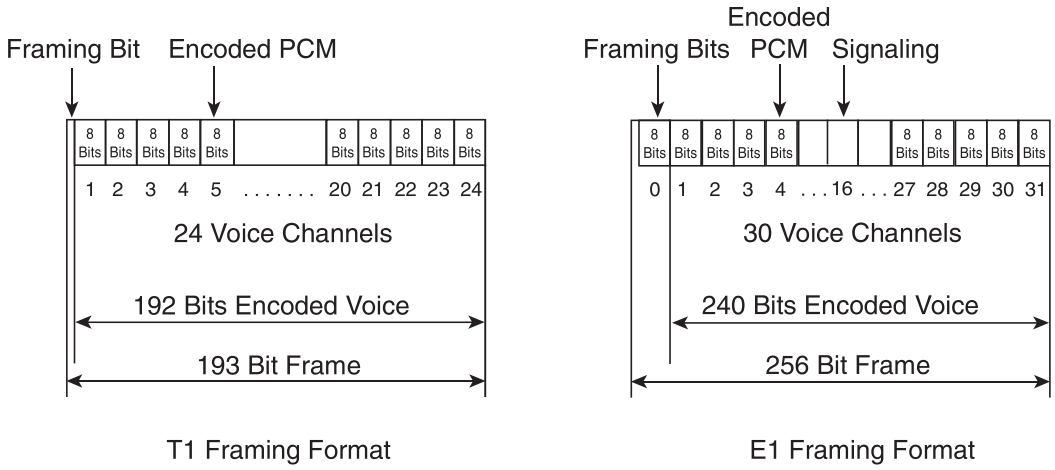
Trunks

Trunks carry traffic between telephony switching nodes. While analog trunks still exist, most trunks in use today are digital trunks, which are the focus of this section. Digital trunks may be either four-wire (twisted pairs) or fiber optic medium for higher capacity. T1 and E1 are the most common trunk types for connecting to End Offices. North American networks use T1, and European networks use E1.

On the T1/E1 facility, voice channels are multiplexed into digital bit streams using Time Division Multiplexing (TDM). TDM allocates one timeslot from each digital data stream's frame to transmit a voice sample from a conversation. Each frame carries a total

of 24 multiplexed voice channels for T1 and 30 channels for E1. The T1 frame uses a single bit for framing, while E1 uses a byte. Figure 5-7 shows the formats for T1 and E1 framing.

Figure 5-7 *T1/E1 Framing Formats*



The E1 format also contains a channel dedicated to signaling when using in-band signaling. The T1 format uses "robbed bit" signaling when using in-band signaling. The term "robbed bit" comes from the fact that bits are taken from the PCM data to convey trunk supervisory signals, such as on/off-hook status and winks. This is also referred to as A/B bit signaling. In every sixth frame, the least significant bits from each PCM sample are used as signaling bits. In the case of Extended Superframe trunks (ESF), A/B/C/D bits are used to indicate trunk supervision signals. A/B bit signaling has been widely replaced by SS7 signaling, but it still exists in some areas.

Trunks are multiplexed onto higher capacity transport facilities as traffic is aggregated toward tandems and transit switches. The higher up in the switching hierarchy, the more likely optical fiber will be used for trunk facilities for its increased bandwidth capacity. In North America, Synchronous Optical Network (SONET) is the standard specification for transmission over optical fiber. SONET defines the physical interface, frame format, optical line rates, and an OAM&P protocol. In countries outside of North America, Synchronous Digital Hierarchy (SDH) is the equivalent optical standard. Fiber can accommodate a much higher bandwidth than copper transmission facilities, making it the medium of choice for high-density trunking.

Standard designations describe trunk bandwidth in terms of its capacity in bits/second. The basic unit of transmission is Digital Signal 0 (DS0), representing a single 64 kb/s channel that occupies one timeslot of a Time Division Multiplex (TDM) trunk. Transmission rates are calculated in multiples of DS0 rates. For example, a T1 uses 24 voice channels at 64 kb/s per channel to produce a DS1 transmission rate of 1.544 mb/s, calculated as follows:

24 x 64 kb/s = 1.536 kb/s + 8000 b/s framing bits = 1.544 mb/s

The optical transmission rates in the SONET transport hierarchy are designated in Optical Carrier (OC) units. OC-1 is equivalent to T3. Higher OC units are multiples of OC-1; for example, OC-3 is simply three times the rate of OC-1. In North America, the electrical equivalent signals are designated as Synchronous Transport Signal (STS) levels. The ITU SDH standard uses the STM to designate the hierarchical level of transmission. Table 5-2 summarizes the electrical transmission rates, and Table 5-3 summarizes the SONET/SDH transmission rates.

Table 5-2 *Electrical Transmission Rates*

Designation	Voice Channels	Transmission Rate mb/s
T1 (North America)	24	1.544
E1 (Europe)	30	2.048
E3 (Europe)	480	34.368
T3 (North America)	672	44.736

Table 5-3 *SONET/SDH Transmission Rates*

SONET Optical Level	SONET Electrical Level	SDH Level	Voice Channels	Transmission Rate mb/s
OC-1	STS-1	—	672	51.840
OC-3	STS-3	STM-1	2016	155.520
OC-12	STS-12	STM-4	8064	622.080
OC-48	STS-48	STM-16	32,256	2488.320
OC-96	STS-96	STM-32	64,512	4976.64
OC-192	STS-192	STM-64	129,024	9953.280
OC-768	STS-768	STM-256	516,096	39,813.120

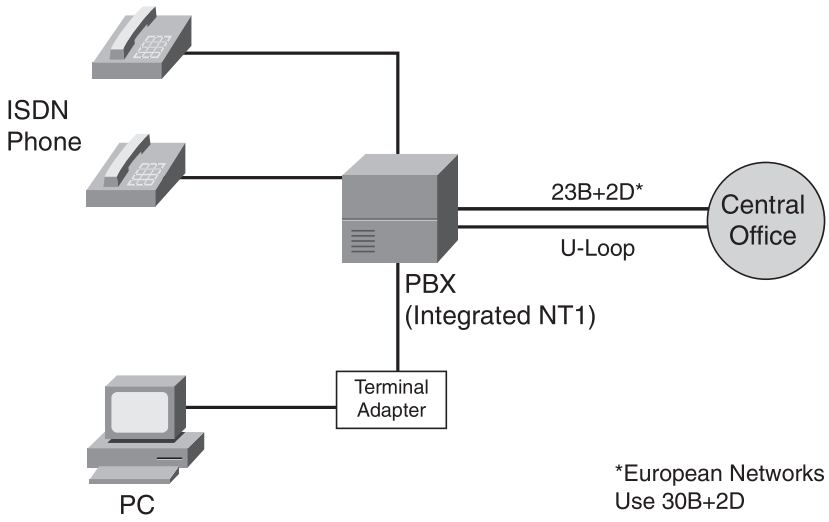
In addition to copper and fiber transmission mediums, microwave stations and satellites are also used to communicate using radio signals between offices. This is particularly useful where it is geographically difficult to install copper and fiber into the ground or across rivers.

ISDN PRI

Primary Rate Interface (PRI) provides ISDN access signaling over trunks and is primarily used to connect PBXs to the CO. As with BRI, PRI converts all data at the customer premises into digital format before transmitting it over the PRI interface. In the United States, PRI uses 23 bearer channels for voice/data and one signaling channel for call control. The single signaling channel handles the signaling for calls on the other 23 channels. This scheme is

commonly referred to as 23B+D. Each channel operates at a rate of 64 kb/s. Figure 5-8 illustrates a PBX connected to the CO through a PRI trunk.

Figure 5-8 *ISDN Primary Rate Interface*



Other variations of this scheme use a single D channel to control more than 23 bearer channels. You can also designate a channel as a backup D channel to provide redundancy in case of a primary D channel failure. In the United States, U-Loop for PRI is a four-wire interface that operates at 1.544 mb/s. The U-Loop terminates to an NT1, which is typically integrated into the PBX at the customer premises.

In Europe, PRI is based on 32 channels at a transmission rate of 2.048 mb/s. There are 30 bearer channels and two signaling channels, which are referred to as 30B+2D.

Network Timing

Digital trunks between two connecting nodes require clock synchronization in order to ensure proper framing of the voice channels. The sending switch clocks the bits in each frame onto the transmission facility. They are clocked into the receiving switch at the other end of the facility. Digital facility interfaces use buffering techniques to store the incoming frame and accommodate slight variation in the timing of the data sent between the two ends. A problem arises if the other digital switch that is connected to the facility has a clock signal that is out of phase with the first switch. The variation in clock signals eventually causes errors in identifying the beginning of a frame. This condition is known as *slip*, and it results in buffer overrun or buffer underrun. Buffer overrun occurs if the frequency of the sending clock is greater than the frequency of the receiving clock, discarding an entire frame of data. Buffer underrun occurs if the frequency of the sending clock is less than the frequency of the receiving clock, repeating a frame of data. Occasional slips do not present a real problem for voice calls, although excessive slips result in degraded speech quality. However, they

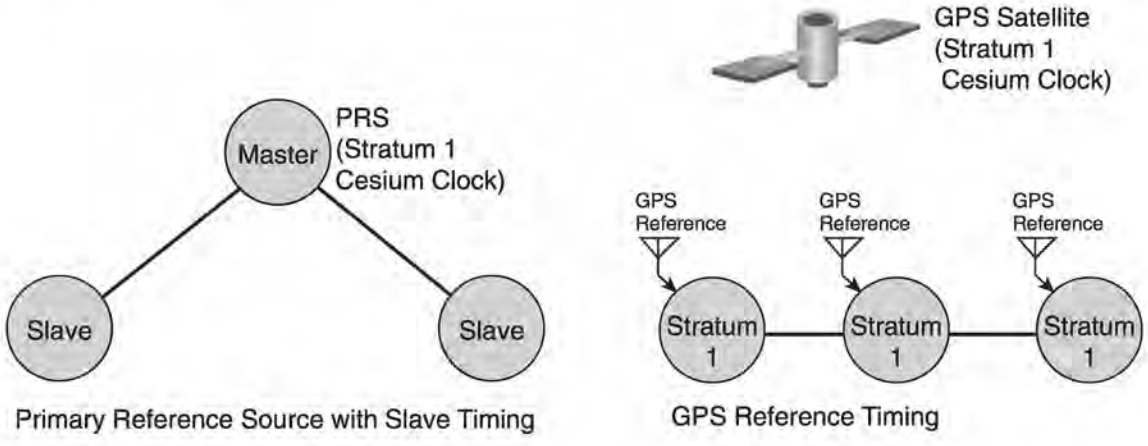
are more detrimental to the data transfer, in which each bit is important. Therefore, synchronization of time sources between the digital switches is important. Because digital transmission facilities connect switches throughout the network, this requirement escalates to a network level, where the synchronization of many switches is required.

There are various methods of synchronizing nodes. One method involves a single master clock source, from which other nodes derive timing in a master/slave arrangement. Another method uses a plesiochronous arrangement, where each node contains an independent clock whose accuracy is so great that it remains independently synchronized with other nodes. You can also use a combination of the two methods by using highly accurate clocks as a Primary Reference Source (PRS) in a number of nodes, providing timing to subtending nodes in the network.

The clocks' accuracy is rated in terms of stratum levels. Stratums 1 through 4 denote timing sources in order of descending accuracy. A stratum 1 clock provides the most accurate clock source with a free-running accuracy of $\pm 1 \times 10^{-11}$, meaning only one error can occur in 10^{11} parts. A stratum 4 clock provides an accuracy of $\pm 32 \times 10^{-6}$.

Since the deployment of Global Positioning System (GPS) satellites, each with a number of atomic clocks on-board, GPS clocks have become the preferred method of establishing a clock reference signal. Having a GPS clock receiver at each node that receives a stratum 1-quality timing signal from the GPS satellite flattens the distributed timing hierarchy. If the GPS receiver loses the satellite signal, the receiver typically runs free at stratum 2 or less. By using a flattened hierarchy based on GPS receivers, you remove the need to distribute the clock signal and provide a highly accurate reference source for each node. Figure 5-9 shows an example that uses a stratum 1 clock at a digital switching office to distribute timing to subtending nodes, and also shows an example that uses a GPS satellite clock receiver at each office.

Figure 5-9 *Network Timing for Digital Transmission*



SS7 links are subject to the same timing constraints as the trunk facilities that carry voice/data information because they use digital trunk transmission facilities for transport. If they produce unrecoverable errors, slips on the transmission facilities might affect SS7 messages. Therefore, you must always consider network timing when establishing SS7 links between nodes in the PSTN.

The Central Office

The Central Office (CO) houses the digital switching equipment that terminates subscribers' lines and trunks and *switch* calls. The term *switch* is a vestige of the switchboard era, when call connections were manually created using cords to connect lines on a plugboard. Electro-mechanical switches replaced manual switchboards, and those eventually evolved into the computer-driven digital switches of today's network. Now switching between calls is done electronically, under software control.

The following section focuses on these areas of the CO:

- The Main Distribution Frame
- The Digital Switch
- The Switching Matrix
- Call Processing

Main Distribution Frame

Incoming lines and trunks are terminated on the Main Distribution Frame (MDF). The MDF provides a junction point where the external facilities connect to the equipment within the CO. Jumpers make the connections between the external facilities and the CO equipment, thereby allowing connections to be changed easily. Line connections from the MDF to the digital switching equipment terminate on line cards that are designed to interface with the particular type of line being connected—such as POTS, ISDN BRI, and Electronic Key Telephone Set (EKTS) phone lines. For analog lines, this is normally the point at which voice encoding takes place. Trunk connections from the MDF are terminated on trunk interface cards, providing the necessary functions for message framing, transmission, and reception.

The Digital Switch

The digital switch provides a software-controlled matrix of interconnections between phone subscribers. A handful of telecommunications vendors produce the digital switches that comprise the majority of the modern PSTN; Nortel, Lucent, Siemens, Alcatel, and Ericsson hold the leading market share. While the digital switch's basic functionality is

common across vendors, the actual implementation is vendor dependent. This section provides a general perspective on the functions of the digital switch that are common across different implementations.

All digital switches are designed with some degree of distributed processing. A typical architecture includes a central processing unit that controls peripheral processors interfacing with the voice channels. Redundancy is always employed in the design to provide the high reliability that is expected in the telephony network. For example, the failure of one central processing unit results in the activation of an alternate processing unit.

The line and trunk interface cards, mentioned previously, represent the point of entry into the digital switch. These cards typically reside in peripheral equipment that is ultimately controlled by the central processor. Within the digital switch, all voice streams are digitized data. Some voice streams, such as those from ISDN facilities and digital trunks, enter the switch as digital data. Other voice streams, such as the analog phone, enter as analog data but undergo digital conversion at their point of entry. Analog lines interface with line cards that contain codecs, which perform the PCM processing to provide digital data to the switch and analog data to the line. Using the distributed processing architecture, many functions related to the individual voice channels are delegated to the peripheral interface equipment. This relieves the central processor of CPU intensive, low-level processing functions, such as scanning for on/off hooks on each individual line to determine when a subscriber wants to place a call.

The central processing unit monitors information from peripheral processors on call events—such as origination, digit collection, answer, and termination—and orchestrates the actual call setup and release. Information from these events is also used to perform call accounting, billing, and statistical information such as Operational Measurements (OM).

Although the main purpose of the digital switch is to perform call processing, much of its functionality is dedicated to maintenance, diagnostics, and fault recovery to ensure reliability.

TIP An OM is a counter that records an event of particular interest, such as the number of call attempts or the number of a particular type of message received, to service providers. OMs can also be used to record usage in terms of how long a resource is used. Modern digital switches usually record hundreds, or even thousands of different types of OMs for various events taking place in the switch.

Switching Matrix

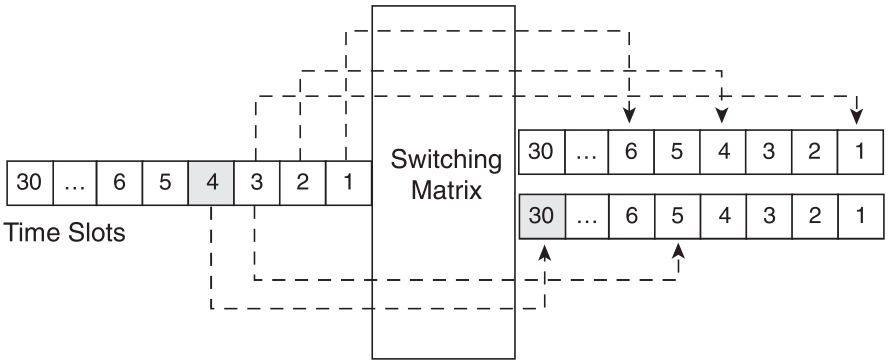
A modern digital switch can process many voice channels. The actual number of channels it processes varies with the switch vendor and particular model of switch, but they often

process tens of thousands of voice channels in a single switch. A number of switches have capacities of over 100,000 connections.

The switch is responsible for many tasks, but one of its primary functions is connecting voice channels to create a bi-directional conversation path between two phone subscribers. All digital switches incorporate some form of switching matrix to allow the connection of voice channels to other voice channels. Once a circuit is set up between the two subscribers, the connection remains for the duration of the call. This method of setting up call connections is commonly known as circuit switching.

Figure 5-10 illustrates how a switching matrix demultiplexes individual timeslots from a multiplexed stream of voice channels and inserts them into the appropriate time slot for a connection on another facility, to connect voice channels. For example, in the figure, time slot 4 from the digital stream on the left connects to timeslot 30 of the digital stream on the right. The figure shows thirty channels, but the number of channels depends on the individual implementation of the switching matrix.

Figure 5-10 *TDM Switching Matrix*



Each timeslot represents a voice connection path. The matrix connects the two paths to provide a conversation path between two parties. For long-distance calls that traverse a number of switches, an individual call goes through multiple switching matrices and is mapped to a new timeslot at each switching point. When the call is set up, it occupies the voice channel that was set up through the network for the duration of the call.

Call Processing

Call processing is associated with the setup, maintenance, and release of calls within the digital switch. The process is driven by software, in response to stimulus from the facilities coming into the switch. Signaling indications, such as on/off-hook, dialing digits, and answer, are all part of the stimuli that drive the processing of calls.

Each call process can be represented as an originating call half and a terminating call half. When combined, the two halves are completely representative of the call. The originating half is created when the switch determines that the originator is attempting a call. The terminating call half is created when the destination has been identified, typically at the translations or routing phase. The Intelligent Network standards have established a standardized call model, which incorporates the half-call concept. A complete discussion of the call model is presented in Chapter 11, "Intelligent Networks (IN)."

Call processing can be broken down in various ways; the following list provides a succinct view of the major stages of establishing and disconnecting a call.

- Origination
- Digit Collection
- Translation (Digit Analysis)
- Routing
- Connection
- Disconnection

Additional functions, such as billing and service interactions, also take place, but are excluded in our simple view of processing.

Origination

For a line, this initial phase of call processing occurs when a subscriber goes off-hook to initiate a call. The actual event provided to the digital switch to indicate a line origination can be a change in loop current for analog lines, or a setup message from an ISDN BRI facility. In-band A/B bit off-hook signaling, an ISDN PRI setup message, or an Initial Address Message from an SS7 signaled trunk can signal a digital trunk's origination. All of these events indicate the origination of a new call. The origination event creates the originating half of the call.

Digit Collection

For analog line originations, the switch collects digits as the caller dials them. Inter-digit timing monitors the amount of time the caller takes to dial each digit so that the line cannot be left in the dialing state for an infinite amount of time. If the caller does not supply the required number of digits for calling within a specified time, the caller is usually connected to a digital announcement to indicate that there is a problem with dialing, a Receiver Off-Hook (ROH) tone, or both. The dialing plan used for the incoming facility usually specifies the number of digits that are required for calling.

For ISDN lines, the dialed digits are sent in an ISDN Setup message.

Translation

Translation, commonly referred to as digit analysis, is the process of analyzing the collected digits and mapping them to a result. The translation process directs calls to their network destination. The dial plan associated with the incoming line, or trunk, is consulted to determine how the digits should be translated. Different dial plans can be associated with different incoming facilities to allow flexibility and customization in the translation of incoming calls. The dial plan specifies such information as the minimum and maximum number of digits to be collected, acceptable number ranges, call type, special billing indicators, and so forth. The translation process can be somewhat complex for calls that involve advanced services like Centrex, which is often associated with business phones.

TIP Centrex is a set of services provided by the local exchange switch to business subscribers, including features like ring again, call parking, and conferencing. Centrex allows businesses to have many of the services provided by a PBX without the overhead of PBX cost, administration, and maintenance.

The process of digit translation can produce several different results. The most common result is a *route* selection for the call to proceed. Other results include connection to a recorded announcement or tone generator, or the sending of an Intelligent Network Query message for calls involving Intelligent Network services. Network administrators provision dial plan, routing information, and other translation-related information on the switch. However, information returned from IN queries can be used to modify or override statically provisioned information, such as routing.

Routing

The call proceeds to the routing stage after translation processing. Routing is the process of selecting a voice channel (on a facility) over which to send the outbound call toward its intended destination, which the dialed digits identify during translation. Routing typically uses route lists, which contain multiple routes from which to choose. For calls that are destined outside of the switching node, a *trunk group* is selected for the outbound call. A trunk group is a collection of trunk members that are connected to a single destination. After a trunk group is selected, an individual *trunk member* is selected from the group. A trunk member occupies an individual time slot on a digital trunk.

Routing algorithms are generally used for selecting the individual trunk circuit. For example, members of an outgoing trunk group are commonly selected using algorithms such as least idle, most idle, and ascending or descending order (based on the numerical trunk member number).

Connection

Call connection must take place on both the transmit and receive paths for a bi-directional conversation to take place. Each involved switch creates a connection between the originating half of the call and the terminating half of the call. This connection must be made through the switching matrix, and the speech path must be *cut through* between the incoming and outgoing voice channels. Supervision messages or signals sent from the central processor to the peripheral interfaces typically cut through the connection for the speech path. The central processor uses supervision signals to indicate how the peripheral processors should handle lower-level functions. It is typical to cut through the backward speech path (from terminator to originator) before cutting through the forward speech path. This approach allows the terminating switch to send the audible ringback over the voice channel, to the originating switch. When the originating switch receives an answer indication, the call path should be connected in both directions.

Disconnection

A call may be disconnected when it is active, meaning that it has been set up and is in the talking state. Disconnection can be indicated in a several ways. For analog lines, the originating or terminating side of the call can go on-hook, causing a disconnection.

TIP Actually, the call is not disconnected when the terminating line goes on-hook, in some cases. These cases are examined further in Chapter 8, "ISUP."

ISDN sets send a Disconnect message to disconnect the call. For trunks using in-band signaling, on-hook is signaled using the signaling bits within the voice channel. For SS7 trunks, a Release message is the signal to disconnect a call.

Call Setup

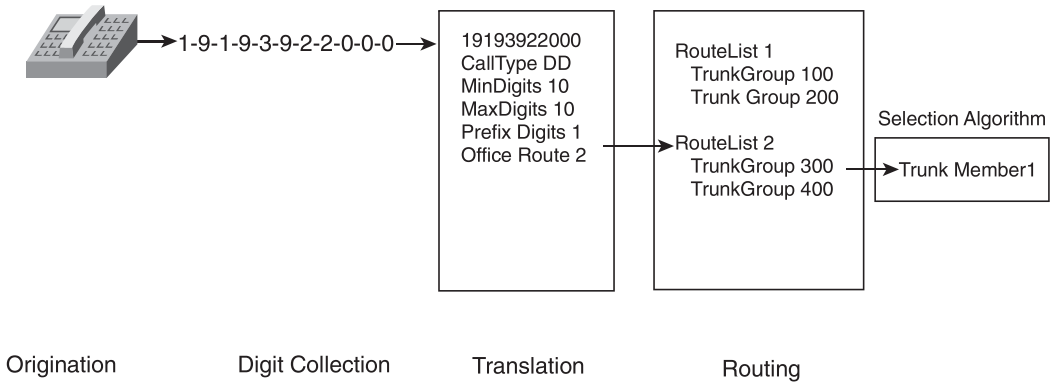
Figure 5-11 shows a typical call setup sequence for a line-to-trunk call. For these calls, the originator dials a number and the digits are collected and processed according to the originating line's dial plan. The dial plan yields a result and points to a list of routes to another switching node. The route list contains a list of trunk groups, from which one group will be selected, usually based on primary and alternate route choices. After the group is selected, an actual trunk member (digital timeslot) is chosen for the outgoing path. The selection of the individual trunk member is typically based on standardized trunk selection algorithms, such as:

- **Most Idle**—The trunk member that has been used the least
- **Least Idle**—The trunk member that has been used the most

- **Ascending**—The next non-busy trunk member, in ascending numerical order
- **Descending**—The next non-busy trunk member, in descending numerical order

Both a call origination endpoint and a call termination endpoint have been established in respect to the digital switch processing the call. The connection can now be made through the switching matrix between the two endpoints. The timing of the actual speech path cut-through between the external interfaces varies based on many factors, but the switch now has the information it needs to complete the full connection path at the appropriate time, as determined by software.

Figure 5-11 *Basic Origination Call Processing*



Integration of SS7 into the PSTN

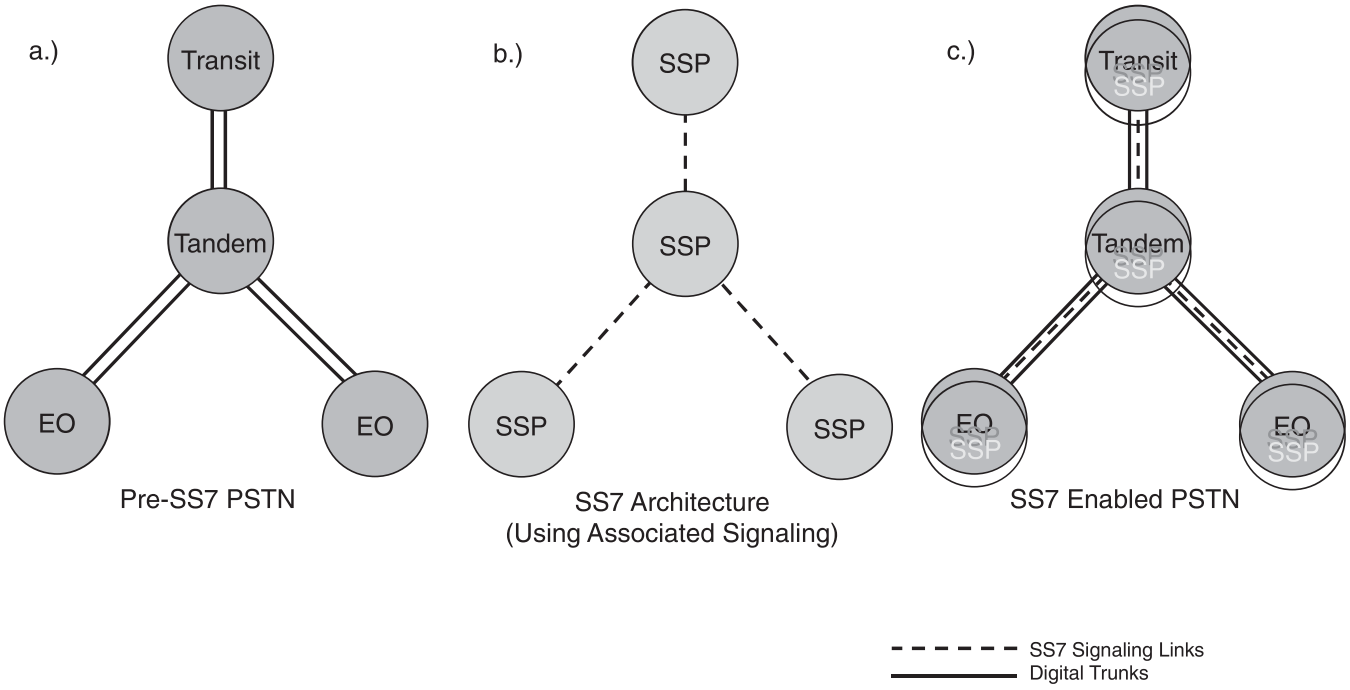
This section provides a brief overview of how the SS7 architecture is applied to the PSTN. Since SS7 has not been presented in great detail, the examples and information are brief and discussed only in the context of the network nodes presented in this section.

The PSTN existed long before SS7. The network's general structure was already in place, and it represented a substantial investment. The performance requirements mandated by the 800 portability act of 1993 was one of the primary drivers for the initial deployment of SS7 by ILECs in the United States. IXCs embraced SS7 early to cut down on post-dial delay which translated into significant savings on access/egress charges. Federal regulation, cost savings, and the opportunity to provide new revenue generating services created a need to deploy SS7 into the existing PSTN.

SS7 was designed to integrate easily into the existing PSTN, to preserve the investment and provide minimal disruption to the network. During SS7's initial deployment, additional hardware was added and digital switches received software upgrades to add SS7 capability to existing PSTN nodes. In the SS7 network, a digital switch with SS7 capabilities is referred to as a Service Switching Point (SSP). When looking at the SS7 network topologies in later chapters, it is important to realize that the SSP is not a new node in the network.

Instead, it describes an existing switching node, to which SS7 capabilities have been added. Similarly, SS7 did not introduce new facilities for signaling links, but used timeslots on existing trunk facilities. PSTN diagrams containing End Offices and tandems connected by trunks represent the same physical facilities as those of SS7 diagrams that show SSP nodes with interconnecting links. The introduction of SS7 added new nodes, such as the STP and SCP; however, all of the switching nodes and facilities that existed before SS7 was introduced are still in place. Figure 5-12 shows a simple view of the PSTN, overlaid with SS7-associated signaling capabilities.

Figure 5-12 *SS7 Overlaid onto the PSTN*



View *a* in the previous figure shows that trunk facilities provide the path for voice and in-band signaling. View *b* shows the SS7 topology using simple associated signaling for all nodes. View *c* shows the actual SS7-enabled PSTN topology. The existing switching nodes and facilities are enhanced to provide basic SS7 call processing functionality. Although this associated signaling architecture is still quite common in Europe, the United States primarily uses a quasi-associated signaling architecture.

SS7 Link Interface

The most common method for deploying SS7 links is for each link to occupy a timeslot, such as a T1 or E1, on a digital trunk. As shown in Figure 5-12, the signaling links actually travel on the digital trunk transmission medium throughout the network. At each node, the

SS7 interface equipment must extract the link timeslot from the digital trunk for processing. This process is typically performed using a channel bank, or a Digital Access and Cross-Connect (DAC), which demultiplexes the TDM timeslot from the digital trunk. The channel bank, or DAC, can extract each of the timeslots from the digital stream, allowing them to be processed individually. The individual SS7 link provides the SS7 messages to the digital switch for processing. While implementations vary, dedicated peripheral processors usually process the lower levels of the SS7 protocol (Level 1, Level 2, and possibly a portion of Level 3); call- and service-related information is passed on to the central processor, or to other peripheral processors that are designed for handling call processing–related messages. Of course, this process varies based on the actual equipment vendor.

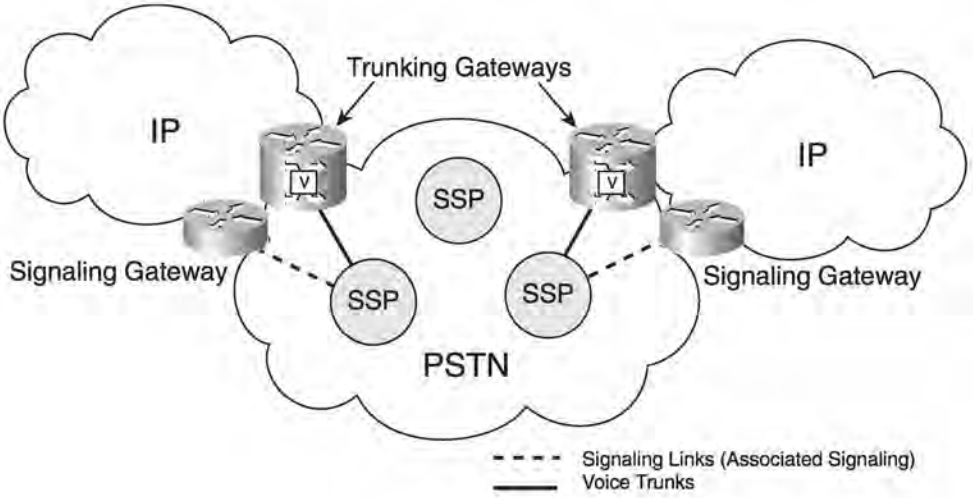
Evolving the PSTN to the Next Generation

The expansion of the Internet continues to drive multiple changes in the PSTN environment. First, more network capacity is used to transport data over the PSTN. Dial-up Internet services use data connections that are set up over the PSTN to carry voice-band data over circuit-switched connections. This is a much different situation than sending data over a data network. Data networks use packet switching, in which many data transactions share the same facilities. Circuit-switched connections are dedicated connections, which occupy a circuit for the duration of a call. The phone networks were originally engineered for the three-minute call, which was the average length used for calculations when engineering the voice network. Of course, Internet connections tend to be much more lengthy, meaning that more network capacity is needed. The changes driven by the Internet, however, reach much further than simply an increase in network traffic. Phone traffic is being moved to both private packet-based networks and the public Internet, thereby providing an alternative to sending calls over the PSTN. Several different architectures and protocols are competing in the VoIP market to establish alternatives to the traditional circuit-switched network presented in this chapter. The technologies are not necessarily exclusive; some solutions combine the various technologies. Among the current leading VoIP technologies are:

- Soft switches
- H.323
- Session Initiation Protocol (SIP)

Each of these VoIP architectures use VoIP-PSTN gateways to provide some means of communication between the traditional PSTN networks and VoIP networks. These gateways provide access points for interconnecting the two networks, thereby creating a migration path from PSTN-based phone service to VoIP phone service. The core network interface connections for VoIP into the PSTN are the trunk facilities that carry the voice channels and the signaling links that carry SS7 signaling. PRI is also commonly used for business to network access. Figure 5-13 shows the interconnection of VoIP architectures to the PSTN using signaling gateways and trunking gateways. Chapter 14, “SS7 in the Converged World,” discusses these VoIP technologies in more detail.

Figure 5-13 *VoIP Gateways to the PSTN*



Summary

This chapter provides an overview of the PSTN, as it existed before VoIP technologies emerged. The majority of the PSTN still appears as this chapter presents it. Many of the diagrams in telecommunications literature illustrating *next generation* technologies—such as soft switches, H.323, and Session Initial Protocol (SIP)—show interfaces to the PSTN. The diagrams refer to the PSTN discussed here, dominated by large, digital switches. The technologies introduced often replace some portion of the existing PSTN; however, they must also remain connected to the existing PSTN to communicate with the rest of the world. The VoIP-PSTN gateways provide this transition point, thus enabling a migration path from the traditional PSTN to the next generation architecture.

While the PSTN varies in its implementation from country to country, a number of common denominators exist. The PSTN is a collection of digital switching nodes that are interconnected by trunks. The network topology is usually a hierarchical structure, but it often incorporates some degree of mesh topology. The topology provides network access to residential and business subscribers for voice and data services. VoIP began another evolution of the PSTN architecture. The PSTN is a large infrastructure that will likely take some time to completely migrate to the next generation of technologies; but this migration process is underway.

CHAPTER 8

ISDN User Part (ISUP)

The *ISDN User Part (ISUP)* is responsible for setting up and releasing trunks used for inter-exchange calls. As its name implies, ISUP was created to provide core network signaling that is compatible with ISDN access signaling. The combination of ISDN access signaling and ISUP network signaling provides an end-to-end transport mechanism for signaling data between subscribers. Today, the use of ISUP in the network has far exceeded the use of ISDN on the access side. ISUP provides signaling for both non-ISDN and ISDN traffic; in fact, the majority of ISUP-signaled traffic currently originates from analog access signaling, like that used by basic telephone service phones.

The primary benefits of ISUP are its speed, increased signaling bandwidth, and standardization of message exchange. Providing faster call setup times than Channel Associated Signaling (CAS), it ultimately uses trunk resources more effectively. The difference in post-dial delay for calls using ISUP trunks is quite noticeable to the subscriber who makes a call that traverses several switches.

NOTE Post-dial delay is the time between when the originator dials the last digit and the originating end receives an indication (or audible ringback).

In addition to its speed efficiencies, ISUP enables more call-related information to be exchanged because it uses Common Channel Signaling (CCS). CAS signaling severely limits the amount of information that can be exchanged over trunks because it shares a small amount of space with a call's voice stream. ISUP defines many messages and parameters, therefore, allowing information about a call to be exchanged both within the network and between end-users. Although messages and parameters do vary between different countries, a given variant provides a standard means of exchanging information between vendor equipment within the national network, and to a large degree, at the international level.

For the reader who is unfamiliar with the PSTN and how switching exchanges work, Chapter 5, "The Public Switched Telephone Network (PSTN)," explains the PSTN, describes the basic concepts of call processing at an exchange, and introduces the concepts of trunks, trunkgroups, and routing.