Telecommunications Essentials

The Complete Global Source for Communications Fundamentals, Data Networking and the Internet, and Next-Generation Networks

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Chapter 5

The PSTN

This chapter talks about the public switched telephone network (PSTN). It talks about what comprises the PSTN, what sorts of technologies have been used to complete the connections, how the signaling systems operate, and what the basic backbone architectures entail in terms of components and transmission capacities. This chapter also discusses intelligent networks (INs) and what they promise in terms of service logic and feature availability. Finally, this chapter describes some of the trends in the evolution of the PSTN that will support the new generation of applications.

The PSTN Infrastructure

Our views about what a network should be designed to support and what the infrastructure should be comprised of have changed quite a bit over the years, as applications and technology have changed. Before discussing what is needed in a network today, this chapter takes a look at how the PSTN infrastructure evolved and where it is today.

The traditional PSTN infrastructure was specifically designed to support only voice communications. At the time this infrastructure was being designed, we had no notion of data communications. Initially the traffic type the PSTN was designed to support was continuous real-time voice.

Another variable that's important to the design of the PSTN has to do with the length of calls. Most voice calls are quite short, so the circuit switches in the PSTN are engineered for call durations of three minutes or less. The average Internet session, on the other hand, lasts around an hour. This means that increased Internet access through the PSTN has, in some locales, put a strain on the local exchanges. If a circuit switch is blocked because it is carrying a long Internet session, people may not be able to get a dial tone. There are several solutions to this problem. For example, as discussed in Chapter 10, "Next-Generation Networks," we can apply intelligence in front of some exchanges so that calls destined for ISPs can be diverted over a packet-switched network to the ISP rather than being completed on a circuit-switched basis through the local exchange.

Yet another variable that's important to the design of the PSTN has to do with what it was designed to support. The capacities of the channels in the PSTN are of the narrowband generation—they are based on 64Kbps channels. The worldwide infrastructure to accommodate voice communications evolved to include a series of circuit switches. Different switches are used based on the locations to which they're connecting. The switches have a high degree of intelligence built into them, both for establishing the communications channels and for delivering the service logic to activate a growing array of features. In the traditional framework, the monolithic switches in the network had all the smarts. The switch manufacturer and the carrier worked together very closely, and the carrier was not able to introduce new features and services into a particular area until a software release was available for the switch platform through which the neighborhood was being serviced. Thus, carriers were often unable to roll out new services and features because they hadn't yet received the new software releases from the switch manufacturers. Over time, we have separated the functions of switching and connection establishment from the functions involved in the intelligence that enables various services and features to be activated.

The traditional PSTN is associated with highly developed, although not necessarily integrated, operational support systems (such as billing systems, provisioning systems, network management systems, customer contact systems, and security systems). These systems have very well-developed business processes and techniques for managing their environments. But the various systems' databases cannot yet all speak to one another to give one comprehensive view. (But at least those systems exist, unlike in the public Internet, where the operational support systems are only now beginning to emerge to help manage that environment.)

The backbone of the traditional PSTN was largely based on a generation that we call the Plesiochronous Digital Hierarchy (PDH), which includes the T-carrier, E-carrier, and J-carrier standards. The local loop of the PSTN was provisioned as a twisted-copper-pair analog subscriber line.

Service Providers

Many abbreviations and acronyms are used to define the various players and the parts of the network in which they play. Some telcos can and do fulfill more than one of these functions; the extent to which they can or do fulfill more than one of these functions partly depends on the policy, regulatory, and licensing conditions that prevail in different countries. The following terms are largely used in the United States, but they are important to the discussion in this chapter because they illustrate the functions service providers are addressing:

- **PTO**—PTO stands for *public telecommunications operator*, which is the name for an incumbent carrier in places other than the United States.
- **VAN**—VAN stands for *value-added network provider*. This term originated around 1970 and was applied to companies that were competing to provide telecommunications services, specifically with offerings focused on data communications and data networking. VANs provided more than a simple pipe from Point A to Point B. They provided some additional intelligence in the network, to, for example, perform error detection and correction, or to convert protocols or languages that different computers speak so that you could have interoperability across the network.
- **LEC**—In the local environment we use the acronym LEC for *local exchange carrier*. There was originally no competition among LECs, but as soon as competition in the local loop picked up, LECs were segmented into ILECs, CLECs, and DCLECs.
- **ILEC**—The ILEC is the *incumbent local exchange carrier*, the original common carrier that either once had, or in some countries still has, monopoly rights in the local loop. For most residents in the United States, this would be one of the four "baby Bells"—Qwest Communications International, SBC Communications, BellSouth Corporation, and Verizon Communications.
- **CLEC**—The CLEC is the *competitive local exchange carrier*. CLECs came about as a result of the Telecommunications Act of 1996, which opened up competition in the local loop. The CLEC is the competitor to the ILEC. Although the decline of the telecommunications economy in 2000 and 2001 forced several CLECs out of business, there are still some CLECs in the United States, and they currently focus on delivering dial tone to business customers.
- **DCLEC (or DLEC)**—DCLEC stands for *data competitive local exchange carrier*. The DCLEC is a company that is specifically focused on supporting data services (for example, providers that offer DSL services to end users).
- **ELEC**—ELEC stands for *Ethernet local exchange carrier*. The ELEC specializes in providing Ethernet solutions in the local loop and metro area.
- **IXC**—The *interexchange carrier* (IXC) is the carrier for long-distance and international communications. AT&T Corporation, WorldCom, Sprint, Qwest, and Verizon are the primary IXCs in the United States. Unless certain

stringent requirements imposed by the Federal Communications Commission are met, an IXC cannot offer long-distance services in the areas where it is also the ILEC.

- SP—Because so many lines are being blurred today by bundled services and bundled territories of operation, the basic term *service provider* (SP) is commonly used to refer generically to providers of different types of services.

Network Access

Figure 5.1 is a simple diagram of network access. On the left-hand side is the customer environment, which includes residences (single-line instruments being served by an access line) and business premises (with onsite telephone systems such as private branch exchange [PBXs] or key telephone systems—smaller site systems for installations where there are 50 or fewer employees). Those in the customer environment are connected to the PSTN via access lines. The *access network*, or the *local loop* we so often talk about, includes whatever equipment resides at the customer premise (that is, the customer premises equipment [CPE]), the access line leading to the local exchange, the components at the local exchange on which those access lines terminate (that is, the distribution cross-connects), and the logic used to help control the flow of traffic over the access lines. In the United States, competition is allowed in the local loop, and a myriad of players are interested in owning the local loop (for example, Internet service providers [ISPs], wireless operators, cable TV companies, power utilities). However, worldwide, the incumbent local providers continue to dominate the local loop, and, as usual, politics and economics are principal factors in delaying the mass deployment of high-speed residential access.

The local exchange, in the center of Figure 5.1, is the backbone, or the core, of the network. From the local exchange, we can establish connections into the other providers, such as IXCs for long distance, international carriers for overseas calls, cellular providers, and ISPs.

Services Beyond the Local Loop

Traditionally, we have thought of the local loop as leading to the home or to the business and ending there. But the need for additional bandwidth and capability is now shifting: We need these things within the premise, as well as on the local loop. It is therefore a logical extension for the service provider to not only give you access lines and termination, but also to provide you with the home area networking facilities you need in order to have an end-to-end broadband package. Chapter 15, "The Broadband Home and HANs," talks more about this.

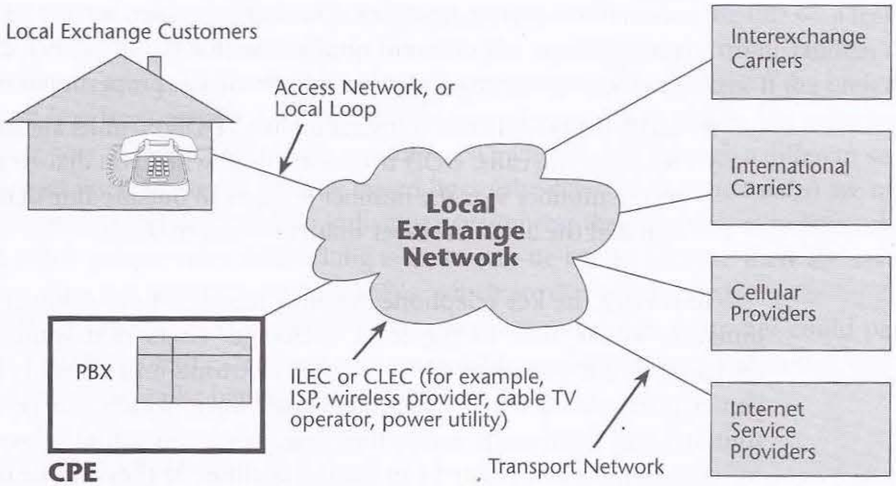


Figure 5.1 Network access

The underlying network access facilities can be either analog or digital loops, and they connect the exchanges to the customer premises. At the customer premises there are the network interfaces, CPE, premises distribution systems where wiring is cross-connected, and network interfaces. The equipment for providing switch access services includes line-termination cards, carrier and multiplexer equipment, and local exchange switching capabilities that support addressing, supervisory alerting, call progress, and other signaling functions.

Access Services

The main categories of access services are trunks, business lines for key telephone systems, centrex service, leased lines, and residential subscriber lines.

Trunks are used to provide connections into the PBX environment. There are three subcategories of trunks:

- **Two-way local exchange trunks**—On these trunks, traffic flows in both the incoming and outgoing directions.
- **DID trunks**—Direct inward dialing (DID) trunks are designed for only incoming calls. A benefit of DID trunks is that they enable the dialed number to ring directly on a user's phone rather than having to go through a centralized attendant. If the population knows whom they want to call directly, and if you want to ease the process of connecting the call, this can be a very useful feature. Another benefit of DID trunks is that they make it

seem like a private line goes directly to the user, but with DID you can support perhaps 100 different numbers with a group of only 25 to 35 trunks (traffic engineering is used to determine the proper number of trunks).

- **DOD trunks**—Direct outward dialing (DOD) trunks are used specifically for outgoing calls. DOD trunks are used when you dial an access code such as the number 9 or the number 8 to get an outside-line dial tone before you can dial the actual number that you want to reach.

To service the key telephone systems, business lines connect the network termination at the user to the local exchange. Users that want to use the local exchange as if it were their PBX rent centrex trunks on a monthly basis. Large companies often access the network via leased lines, which can be a very expensive solution, and home users access the network via residential subscriber lines.

Access lines can either be in analog facilities or they can be digital carrier services. Analog transmission is often called *plain old telephone service* (POTS for short). Three main types of digital services are offered by using twisted-pair cable. The first type of digital services involves T-1 access (at 1.5Mbps), E-1 access (at 2. 048Mbps), and J-1 access (at 1.544Mbps). The second type of digital services is narrowband ISDN (N-ISDN) services, including Basic Rate Interface (BRI) for residences and small businesses and Primary Rate Interface (PRI) for larger businesses. The third type of digital services is the xDSL subscriber lines and high-speed digital subscriber lines that enable the all-important applications of Internet access and multimedia exploration. (Chapter 3, "Transmission Media: Characteristics and Applications," describes the digital services in more detail.)

Transport Services

Transport services are the network switching, transmission, and related services that support information transfer between the originating and terminating access facilities. The underlying facilities include local exchanges and tandem switches, toll and transit switches, international gateways, and interoffice transmission equipment. Transport services include switched services, nonswitched services, and virtual private networks (VPNs).

Switched Services

There are two main types of switched services: public and private.

Switched public services include local calling, long-distance calling, toll-free calling, international calling, directory assistance, operator assistance, and emergency services.

Switched private services can be switchable either because they are deployed within the CPE or because they are deployed on a carrier basis. With CPE-based services, you can add capabilities to the telephone systems onsite in the PBXs—a feature called *electronic tandem networking*. For example, you can use electronic tandem networking to gain some flexibility in routing around congestion points: If the preferred leased line from Switch A to Switch B is occupied or not available, the switch can decide how to reroute that traffic to still reach Switch B, but through a different series of leased lines. However, because leased lines (also referred to as *tie trunks*) are mileage sensitive and dedicated to individual customers, they are very expensive; thus, not much private voice networking is done over tie trunks because there are several more attractive solutions, such as VPNs, which are discussed shortly.

With carrier-based switched private services, a centrex customer could partition and implement extensions across multiple local exchanges and in this way be able to switch traffic between those locations.

Nonswitched Services

Nonswitched services include leased lines, foreign exchange (FX) lines, and off-premises exchanges (OPXs). With leased lines, two locations or two devices are always on, using the same transmission path.

FX lines allow you to make a toll call appear to be a local call. For example, you might have a dedicated leased line that runs from your customer premise to a local exchange in a distant area where you call large numbers of customers. When anyone behind your PBX dials a number associated with that foreign local exchange, the PBX automatically selects the FX line. The dial tone the caller receives is actually coming from the distant local exchange, and the call proceeds as if it were a local call. The tradeoff with FX lines is that although you are not charged per call for your long-distance calls to the specified exchange, you pay a flat monthly fee for the leased line and you have to apply some traffic engineering to ensure that you're not making people wait for the FX line to become available. So with FX lines, you need to find the right balance point between reducing costs and ensuring a high level of service.

OPXs are used in distributed environments, such as a city government. Say that the city government has public works stations, libraries, fire stations, and parks and recreation facilities that are too far from the PBX to be served by the normal cabling. The city uses an OPX setup: It leases a circuit from the PBX to the off-premise location and ties it in as if it were part of that PBX. City government employees can then call one another, using their normal extension plan, their call accounting information can be accumulated so that cost allocations can be performed, and the employees can have access to the full suite of features that a business PBX offers.

VPNs

Although you might think that VPNs are related to the Internet or to Internet Protocol (IP) and are a somewhat new development, they actually originated in the circuit-

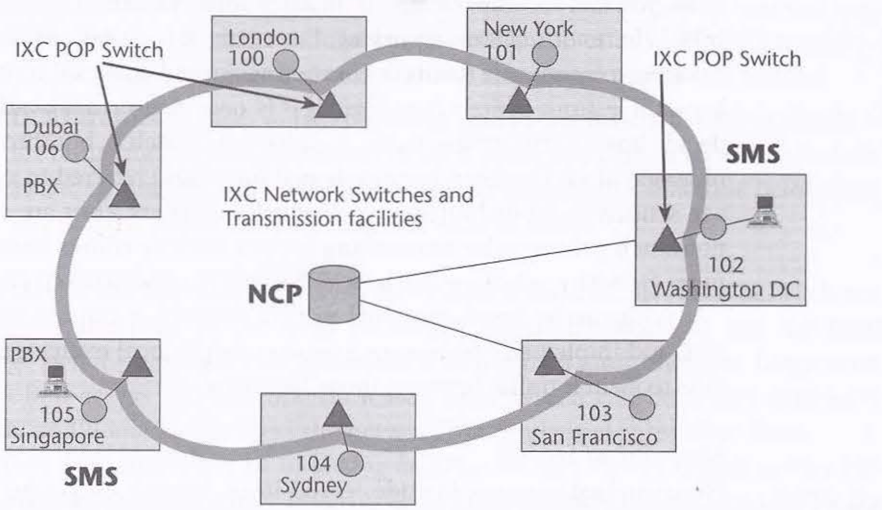


Figure 5.2 An example of a VPN

switched network environment, with AT&T's software-defined network (SDN) in the early 1980s. A VPN is a concept, not a technology platform or a set of networking techniques. A VPN defines a network in which customer traffic is isolated over shared-service provider facilities, so as more customers share the same facilities, their costs go down. The purpose of a VPN, then, is to reduce the high cost of leased lines, while still providing high quality of service and guaranteeing that private traffic has capacity between locations. Figure 5.2 shows an example of a VPN.

The underlying facilities of a VPN include the carrier public network, augmented by network control points and service management systems. Under computer control, the traffic is then routed through the public network in a manner that makes the VPN service seem like a facilities-based private network. Access to the VPN can occur via dedicated access, leased lines, or carrier-switched access, using either an analog or a digital carrier.

The network control point represents a centralized database that stores a subscriber's unique VPN information. The network control point screens every call and then applies call processing in accordance with the customer-defined requirements. A common-channel signaling network connects the various network elements so that they can exchange information with each other in real-time. (Common-channel signaling is discussed later in this chapter, in the section "Signaling Systems.")

A service management system is used to build and maintain the VPN database. It allows customers to program specific functions to accommodate their particular business applications. It transmits information to the network control points, with important instructions on a customer-by-customer basis. Thus, VPNs introduce to the realm of the PSTN a lower-cost alternative to building a private voice network.

PSTN Architecture

The PSTN includes a number of transmission links and nodes. There are basically four types of nodes: CPE nodes, switching nodes, transmission nodes, and service nodes.

CPE Nodes

CPE nodes generally refer to the equipment that's located at the customer site. The main function of CPE nodes is to transmit and receive user information. The other key function is to exchange control information with the network. In the traditional realm, this equipment includes PBXs, key telephone systems, and single-line telephones.

Switching Nodes

Switching nodes interconnect transmission facilities at various locations and route traffic through a network. They set up the circuit connections for a signal path, based on the number dialed. To facilitate this type of switching, the ITU standardized a worldwide numbering plan (based on ITU E.164) that essentially acts as the routing instructions for how to complete a call through the PSTN. The switching nodes include the local exchanges, tandem exchanges (for routing calls between local exchanges within a city), toll offices (for routing calls to or from other cities), and international gateways (for routing calls to or from other countries). Primary network intelligence is contained in the Class 4 switches (that is, toll offices switches) and Class 5 switches (that is, local exchange switches). The Class 4 toll switches provide long-distance switching and network features, and the Class 5 switches provide the local switching and telephony features that subscribers subscribe to. Figure 5.3 shows where the types of telephone exchanges are located.

The Local Exchange The local exchange (also called the Class 5 office or central office) is where communications common carriers terminate customer lines and locate the switching equipment that interconnects those lines. This class office represents the local network. Every subscriber line location in a local exchange is assigned a number, generally seven or eight digits. The first three (or four) digits represent the exchange and identify the local exchange switch that serves a particular telephone. The last four digits identify the individual line number, which is a circuit that is physically connected from the local exchange to the subscriber. The traditional local exchange switch can handle one or more exchanges, with each

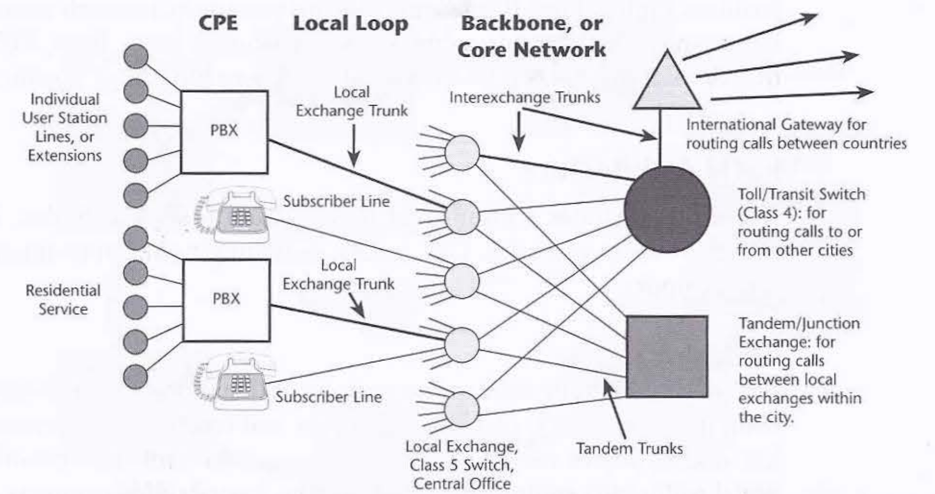


Figure 5.3 Types of telephone exchanges

exchange capable of handling up to 10,000 subscriber lines, numbered 0000 to 9999. In large metropolitan areas, it is common to find one local exchange building housing more than one local exchange switch and for each switch to handle five or more exchanges. These offices are sometimes referred to as *multi-entity buildings*.

The Tandem Office The tandem office, or junction network, is an exchange that is used primarily as a switching point for traffic between local exchanges in a metropolitan area. It is an office that is used to interconnect the local end offices over tandem trunks in a densely settled exchange area where it is not economical for a telephone company to provide direct interconnection between all end offices. The tandem office completes all calls between the end offices but is not directly connected to subscribers.

The Toll Office The toll office (also called the trunk exchange or transit switch) is a telephone company switching center where channels and toll message circuits terminate—in other words, where national long-distance connections are made. This is usually one particular exchange in a city, but larger cities may have several exchanges where toll message circuits terminate.

The International Gateway An international gateway is the point to and from which international services are available in each country. Protocol conversion may take place in the gateway; in ITU terminology, this is called a *centre de transit* (CT). C1 and C2 international exchanges connect only international circuits. CT2 exchanges switch traffic between regional groups of countries, and CT1 exchanges switch traffic between continents. CT3 exchanges connect switch traffic between the national PSTN and the international gateway.

Transmission Nodes

Transmission nodes are part of the transport infrastructure, and they provide communication paths that carry user traffic and network control information between the nodes in a network. The transmission nodes include the transmission media discussed in Chapter 3, as well as transport equipment, including amplifiers and/or repeaters, multiplexers, digital cross-connects, and digital loop carriers.

Service Nodes

Service nodes handle *signaling*, which is the transmission of information to control the setup, holding, charging, and releasing of connections, as well as the transmission of information to control network operations and billing. A very important area related to service nodes is the ITU standard specification Signaling System 7 (SS7), which is covered later in this chapter.

The Transport Network Infrastructure

The transport network includes two main infrastructures. The first is the PDH, also known as T-carrier, E-carrier, and J-carrier wideband transmission standards. This infrastructure was first introduced in the early 1960s. The second infrastructure of the transport network is the Synchronous Digital Hierarchy (SDH; ITU terminology), also known as Synchronous Optical Network (SONET; ANSI terminology), which was first formalized and standardized in 1988. SDH/SONET is the second generation of digital hierarchy, and it is based on a physical infrastructure of optical fibers.

PDH and SDH/SONET are voice-centric circuit-switched network models that switch millions of 64Kbps circuits between various switching points. Each circuit is multiplexed numerous times for aggregation onto transmission facilities. Aggregation occurs at many points in the network: in the access network, within the local exchange, and throughout the interexchanges. Hence, a significant portion of the cost of a network goes to the equipment that performs this aggregation—the multiplexers and cross-connects in both the PDH and SDH/SONET environments.

The PDH Infrastructure

The term *Plesiochronous* makes PDH sound like a dinosaur, and in a way, it is—it's an outdated architecture from the standpoint of the data rates it offers. But the word *Plesiochronous* means "minute variations in timing," and that refers to the fact that the PDH is an *asynchronous infrastructure.* Each network element—that is, each exchange, multiplexer, cross-connect, repeater, and so on—gets its clocking pulse from a different clocking source, and even though those clocking sources are synchronized, there are minute fluctuations in timing. To differentiate the beginning and the end of a conversation, we have to channelize conversations.

PDH was the first system designed to use digitized voice transmission. It was born of the telcos' desire to better use their cable facilities and to enhance the quality of calls. PDH was first used by telcos as a means of aggregating multiple voice channels into a single high-speed digital backbone. Standards that are used today for all-digital switching and transmission come from the original PDH specifications.

PDH defines several things: First, it's an integrated digital network, so it can carry a range of traffic, as long as that traffic is being presented in a digital manner. Therefore, PDH represented the first opportunity for users and carriers to combine voice and data traffic over the same pipes. Second, it specifies the different transmission levels or data rates, some of which are available for customers to subscribe to and others of which are used by operators internally within the backbones. Third, it defines within each of the transmission levels how many channels can be made available.

The T-, E-, and J-Carrier Standards

T-carrier, E-carrier, and J-carrier are PDH standards that are followed in different regions of the world: J-carrier is followed throughout Japan; T-carrier is followed throughout North America; and E-carrier is followed throughout Europe and the majority of other locations throughout the world, including large parts of Asia, Latin America, and Africa. Figure 5.4 compares these three standards. They all share one increment as a common denominator: 64Kbps. But each of the three standards multiplexes together a different number of these 64Kbps channels to derive higher transmission rates.

Having three separate standards—T-, E-, and J-carrier—means that we have to cross between systems that use different standards, and in doing so, we incur additional overhead.

Elements of the PDH Infrastructure

As shown in Figure 5.5, the following are the key elements of the PDH infrastructure:

- Transmission media
- Repeaters
- Channel service units (CSUs)
- Multiplexers
- Digital loop carriers (DLCs)
- Digital cross-connect systems (DCSs)

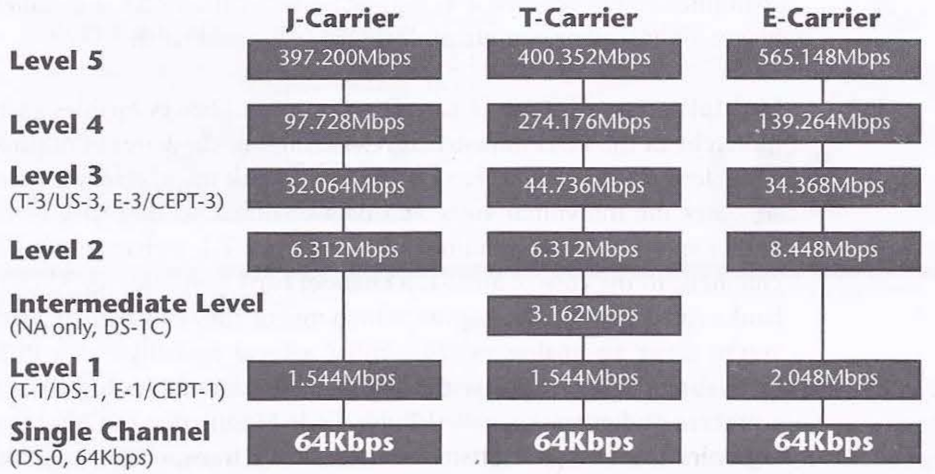


Figure 5.4 T-carrier, E-carrier, and J-carrier standards

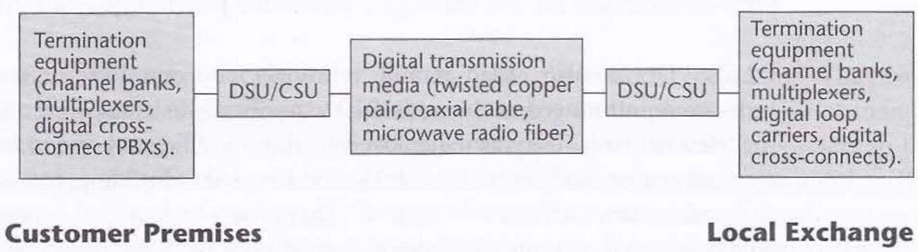


Figure 5.5 PDH components

Transmission media PDH can include a wide variety of transmission media, and the type you use is contingent on the bandwidth you want to be able to support. You could use copper pairs to provision T-1, E-1, or J-1 services, but if you wanted to get into the higher-bandwidth capacities afforded under T-3, E-3, or J-3, you would deploy a higher-bandwidth medium, such as coax, microwave, or fiber. PDH operates on four-wire circuits, which means it operates in full-duplex and you can communicate in both directions simultaneously.

CSUs A CSU terminates each end of a T-, E-, or J-carrier facility. It equalizes the received signal, filters the transmitted and received wave forms, and interacts with customers' and carriers' test facilities. You use a CSU to perform diagnostic tests on span lines and to set up a T-1, E-1, or J-1 line with a PBX, a channel bank, a multiplexer, or any other compliant data terminal equipment.

Multiplexers A series of time-division multiplexers enables us to move up the hierarchy of the PDH infrastructure. The first in the series of multiplexers is generally referred to as *channel banks*. A channel bank has several purposes. First, it consolidates the individual voice and data channels so that they can travel over the higher-speed transmission line. In the case of a T-1, a channel bank consolidates 24 channels; in the case of an E-1, a channel bank consolidates 32 channels. Channel banks can accept analog inputs, which means they can digitize analog voice. So, if you're using an analog switch—either a local exchange or a PBX—the channel bank should be equipped with the codecs that run an analog voice stream through a process of digitization called Pulse Code Modulation (PCM) to convert the analog voice into a digital bitstream that can be transported over the digital carrier. (Codecs are discussed in Chapter 2, "Telecommunications Technology Fundamentals," and PCM is discussed later in this chapter.)

Beyond the channel bank, the multiplexing hierarchy steps through the individual transmission levels. In the case of T-carrier, the levels are T-1 through T-4; for E-carrier, they are E-1 through E-5; and for J-carrier, they are J-1 through J-5.

DLCs DLCs—also called remote terminals, concentrators, or remote concentrators—were introduced in the mid-1970s, specifically as a way to economically expand the telco network. They were deployed to improve efficiency and to lower costs. DLCs reduced analog facilities by up to 80%, and they led to building, real estate liquidation, and maintenance efficiencies as well. They also eliminated the need for load coils, which are used to improve transmission on wire pairs for distances greater than 3.4 miles (5.5 kilometers). DLCs also reduced the number of pairs of copper wires required between the local exchange and the subscriber; they did this by sharing pairs or transmission facilities among many multiplexed conversations. Essentially, the DLC architecture, shown in Figure 5.6, reduces the loop lengths and makes more effective use of high-capacity trunks from a neighborhood into the local exchange.

DLCs continue to evolve, and as they do so, they become smaller systems. The original DLCs were built so that an individual system could service around 600 subscribers, but these boxes achieved only about a 50% fill ratio, which meant that half of the capacity was not being used. Now, given the distribution and density of neighborhoods and populations, smaller DLCs are being created. These systems service up to about 96 subscribers, and utilization is at around a 90% level. These smaller DLCs allow for faster service rollout and a shorter payback period for the deployment. They also facilitate quick response to growth in services and competition.

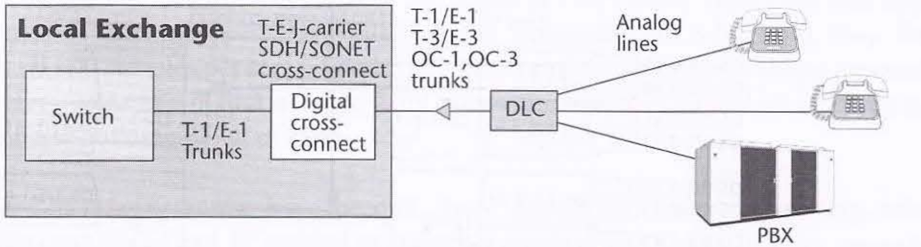


Figure 5.6 DLC architecture

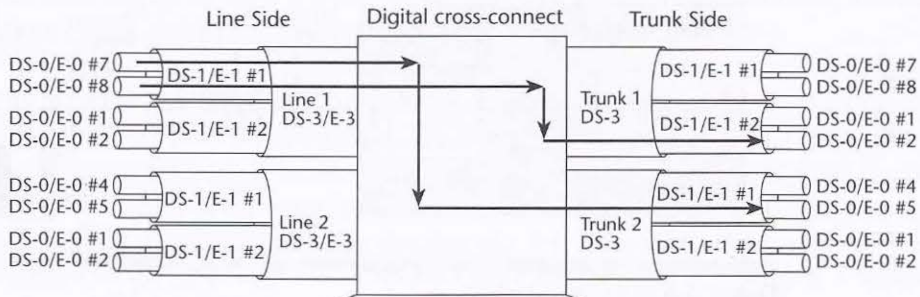
With ever-increasing interest in high-speed broadband access, DLCs could be a tool for shortening loop length, thereby bringing more bandwidth to the customer. Consequently, some of the additional changes that have occurred with the newer generations of DLCs also provide interfaces for SDH/SONET or optical fibers. However, bear in mind that the vast majority of DLCs deployed are incompatible with the xDSL services. It is imperative that the proper generation of DLC be deployed in order to meet the customer's demand for broadband residential access via twisted-pair.

DCSs DCSs were developed in 1981 and officially introduced in 1985. They basically automated the process of circuit provisioning and replaced the use of manual patch panels. The key feature of DCSs is called drop and insert. This refers to the capability of the cross-connect to exchange channels from one facility to another. It is used to implement appropriate routing of traffic, to reroute around congestion or failure, and to allow customers to dynamically reconfigure their own networks. Generally it keeps communications paths in place for continuous use over a period of months, or sometimes even years, but it does allow change as demand warrants.

Essentially, a DCS is a computer system with a variety of software databases that describe first-choice routes and alternate routes (see Figure 5.7). If Channel 7 normally goes out over Line 1 and then goes out over Trunk 1, but Trunk 1 fails, the digital cross-connect can consult its alternate routing table, which might say to reroute that particular line over Trunk 2. A reconfiguration can take place in a matter of minutes.

Digital cross-connects provide for four levels of switching. You can switch between DS-3s and DS-3s or between E-3s and E-3s. You can switch between DS-1s and DS-1s or between E-1s and E-1s. You can switch between DS-0s and E-0s, and you can also potentially switch below that level by using submultiplexed data streams within the DS-0 channel. Some of the individual intelligent multiplexers, such as T-1/E-1 muxes and T-3/E-3 muxes, also offer this capability.

The fact that reconfigurations can be implemented in a matter of minutes—and that customers can implement this capability on their own private networks—is the most



Port In	DS-1	DS-0	Port Out	DS-1	DS-0
Line 1	1	1	Trunk 2	1	5
Line 1	1	2	Trunk 1	2	2
Line 1	2	1	Trunk 1	2	1
Line 1	2	2	Trunk 2	2	4
Line 2	1	6	Trunk 2	2	1
Line 2	1	7	Trunk 2	2	2
Line 2	2	3	Trunk 1	1	8
Line 2	2	4	Trunk 1	1	7

Figure 5.7 The DCS

important and favorable characteristic of the DCS. The main applications for cross-connects are to provide disaster recovery, to bypass a system during maintenance without affecting traffic flows, to reconfigure the network to address peak traffic demands, and to implement a temporary application that can be reconfigured as needed.

Voice Compression Standards

Let me take a moment to talk about how voice was digitalized to carry over the PSTN under the original sampling theorem.

PCM When PCM was developed, we lived in an environment that was largely analog. Therefore, in designing a digitization scheme, it was important to consider that voice would have to undergo many conversions between analog and digital as it was transmitted through the numerous switching nodes and components of a network. And if it had to go through a number of conversions, it could withstand only so many conversions before it began to lose toll quality. Therefore, the sampling theorem that was developed suggested that in order to reproduce voice in toll-quality manner, you have to sample that voice at twice the rate of the highest frequency carried. The highest frequency being carried in the telephone channel was 4,000Hz, so we needed a sampling rate of 8,000 samples per second. Every time we take a sample, we're measuring the amplitude, or voltage, of the signal at that point. Say that the amplitude of the signal is +1.25 volts. We would take that amplitude value and convert it into a binary expression, an 8-bit word. Now we have 8,000 samples per second and 8 bits per sample, resulting in 64Kbps required to carry voice in a digital manner. This is how the 64Kbps channel was derived as the basic building block of PDH.

ADPCM As networks have become more digitalized, fewer conversions take place, and voice can be carried at a higher quality over fewer bits per second. Another standard that is used in the PSTN is Adaptive Differential PCM (ADPCM), which essentially carries digital voice at 32Kbps. ADPCM does something wonderful for an end user. Say you have a traditional T-1 line with PCM channel banks. Over that one T-1, you can extract 24 channels, each of which carries 64Kbps. But your traffic increases, and you need more channels to carry voice traffic. You have two options: You can add another T-1 line, which means a substantial monthly investment, or you can put ADPCM channel banks on the T-1 that you already have, which gives you 48 channels of 64Kbps each. In essence, you can double the capacity of the network without having to add more subscriber lines.

Needless to say, voice compression continues to be applied, and not just in the PSTN. For instance, in wireless networks such as cellular networks, where spectrum is at a premium, we compress voice down to 8Kbps so that we can support more callers within each of the cells.

T-Carrier and E-Carrier Signal Hierarchy

Because competition has entered the marketplace, different operators in an area have often bought equipment from different manufacturers, which means there are a number of standards to deal with. Even if a country once followed ITU standards, new companies may have entered the country with North American–standard

DS-*x* Versus TX-*x* and CEPT-*x* Versus E-*x*

Technically, the DS-*x* and CEPT-*x* terminology (DS-1, DS-3, CEPT-1, CEPT-3, and so on) indicates a specific signal level (and thus usable bandwidth), as well as the electrical interface specification. T-*x* and E-*x* terminology (T-1, T-3, E-1, E-3, and so on) indicates the type of carrier—a specific physical implementation of a DS-*x*/CEPT-*x*. Today, however, the terms DS-*x* and T-*x* are often used interchangeably. For example, someone might use the term DS-1 and another person might use the term T-1 to refer to the same thing—a digital transport that can carry 1.544Mbps over a total of 24 channels. The same applies to the use of the European designation: E-1 is the same as CEPT-1, and so on.

equipment and interfaced with the existing ITU-based equipment. Thus, you really need to be familiar with all the standards. This section covers the signal hierarchy for both T-carrier and E-carrier standards.

The T-Carrier Digital Signal Hierarchy Table 5.1 lists the levels in the T-carrier digital signal hierarchy, the basic building block of which, DS-0, is the 64Kbps channel.

Table 5.1 The T-Carrier Digital Signal Hierarchy

Digital Signal Level	Bit Rate	DS-0 Channel	Number of T-1 Lines
DS-0 (T-0)	64Kbps	1	—
DS-1 (T-1)	1.544Mbps	24	1
DS-2 (T-2)	6.312Mbps	96	4
DS-3 (T-3)	44.736Mbps	672	28
DS-4 (T-4)	274.176Mbps	4,032	168

The first subscriber level, Digital Signal Level 1 (DS-1), provides 1.544Mbps and a total of 24 channels. The DS-2 level is not a subscriber level, nor is it used very frequently in the PSTN. You might see it installed on some campus networks, or perhaps to bundle some DS-1s out of a tandem exchange to a toll exchange.

DS-3 is a high-bandwidth alternative for subscribers, and it is used for interexchange trunks. Both users and carriers get 44.736Mbps with DS-3, which is a total of 672 channels that can carry combined voice, data, fax, and image traffic.

T-Carrier and J-Carrier: Stealing Bits for Signaling

In the T-carrier and J-carrier system, the signaling for each conversation flows in the same channel as the conversation. Thus, for voice purposes, we have the full 64Kbps channel, and every now and then we steal one of the bits of digital voice and replace it with the proper signaling bit. This does not affect the understandability or voice quality of the message. However, if a data stream were traveling through that channel and we went about stealing bits, we would obviously be changing the meaning of the content. Therefore, we accommodate 56Kbps of data within each channel, leaving a little bit of room for the signaling bits to be inserted as needed.

The DS-4 level is used only within the telco, again on interexchange trunks. DS-4 offers roughly 274Mbps and 4,032 channels.

With each of these levels, you must go through a separate multiplexing level. Remember that each of the muxes is driven by a different clocking source, so they each bundle their channels in a slightly different framework. In building the 64Kbps channels up to a T-1 and then building those T-1s up to T-2s and those T-2s up to T-3s, everything is fine unless somewhere along the way one customer decides to extract some capacity to drop off that allocation midway. Say, for example, that you're in Washington, DC, and you need to connect to San Francisco. In Washington, DC, you'd have a T-1 coming into the local exchange, along with multiple other customers in the neighborhood. The local exchange might bundle those T-1s onto a T-2 to pass off to the tandem, and the tandem would bundle them up into T-3s to send to the toll center. The toll center would bundle them up for T-4s to pass across the long haul to San Francisco. This works great, but then you need to add an office in Kansas City. So you need to add a T-4 mux to break it down to all the respective T-3s. Then you need to break down the T-3s into their T-2s, and then break down the T-2s into all their T-1s, and then find your T-1 so that you can extract the channels you want to drop off. Then you need to bundle them all back up onto the T-1 and go back up the scale again. This strict hierarchy requires you to go through all the changes—you can't jump steps as you bundle and unbundle traffic. Therefore, the PDH hierarchy is characterized by a lot of back-to-back multiplexing and demultiplexing in order to drop and add payload. That is one of the highest-cost components of this generation of the PSTN.

The E-Carrier Digital Signal Hierarchy As shown in Table 5.2, E-carrier signals are often called CEPT levels (for Common European Postal and Telegraphy); 64Kbps is the basic increment in E-carrier. CEPT-1 (or E-1) operates at 2.048Mbps and is delivered over 32 channels.

CEPT-2, like T-2, is not used much. CEPT-3, the high-bandwidth alternative, offers 34Mbps and 512 channels. CEPT-4 and CEPT-5 are largely used within telco

E-Carrier: Separate Signaling Channels

The E-carrier system is different from the T-carrier and J-carrier systems in an important way: In the E-carrier system, the signaling information travels in separate channels from the voice and data traffic. Two of the 32 channels are devoted to carrying signaling and control information, and the other 30 channels are available to carry customer payload at 64Kbps. In T-carrier and J-carrier, because the signaling information flows in the conversation channel, voice channels are 64Kbps, but data channels are 56Kbps, with the remaining capacity reserved for signaling.

Table 5.2 The E-Carrier Digital Signal Hierarchy

CEPT Signal Level	Bit Rate	E-0 Channel	Number of E-1 Lines
CEPT-0 (E-0)	64Kbps	—	—
CEPT-1 (E-1)	2.048Mbps	32	1
CEPT-2 (E-2)	8.488Mbps	128	4
CEPT-3 (E-3)	34.368Mbps	512	16
CEPT-4 (E-4)	139.246Mbps	2,048	64
CEPT-5 (E-5)	565.148Mbps	8,192	256

networks, again for their interexchange trunks. Like T-carrier, E-carrier has a strict hierarchy of multiplexers.

The SDH/SONET Infrastructure

SDH/SONET, created in the mid-1980s, is the second generation of digital hierarchy. Whereas PDH involves a lot of overhead because it includes three standards throughout the worldwide, SDH/SONET uses one common standard that applies to networks worldwide. SDH is the ITU standard and followed throughout the parts of the world where ITU standards dominate; SONET is the ANSI standard, which is part of SDH, and it is used in North America and Japan. SDH/SONET was created to be an industry standard for high-speed transmission over optical fiber. It was actually part of a much bigger standard in the works at that time—Broadband ISDN. Broadband ISDN was envisioned for use with advanced applications (for example, tele-education, telesurveillance, telegambling, the ability to collaborate, HDTV). Two technologies were required in order to support such applications—a transport infrastructure that had the significant bandwidth needed to support them (SDH/SONET) and a switching technology (ATM) that could ensure that latencies could be controlled and kept very low. Consequently, SDH/SONET and ATM, as modern broadband technologies, were both born out of the Broadband ISDN standard and a desire to be able to deliver advanced applications.

SDH/SONET is a family of transmission standards designed to achieve compatibility between different fiber-optic transport products, as well as to provide compatibility with the existing digital hierarchy, PDH. A lot of fiber-optic systems have been deployed in the past 16 or more years, but they're not all compatible with one another. They use different forms of cables with different diameters, and they use different types of light sources. And where there are physical incompatibilities, you can't achieve *midspan meet*—where two carrier services have to come together. A railroad analogy can be used here. Think back to when people had just begun to build railroads. Oftentimes, one provider was building tracks going east to west that were a certain width and another provider was building tracks going west to east that were a different width. When the providers met in the middle, their cars couldn't cross each other's tracks. The same thing happens when there's a lack of physical compatibility in fiber-optic transport, and that means you lose the ability to carry network management information on an end-to-end basis. You may not be able to do end-to-end monitoring and control of a network that is multivendor or multicarrier in nature if the vendors or carriers use incompatible physical fiber-optic equipment.

It's always important to develop and have available very strong network management tools. The goal of network management is not to *eliminate* downtime—because we know that would be impractical; rather, it is to *minimize* the resolution time. So the ability to do end-to-end testing remotely is very critical to quick recoverability. SDH/SONET provides the physical layer (that is, Layer 1) framework for broadband applications. It provides a standardized list of optical parameters that define the types of cables and light sources that are allowed. It defines a new table of data rates that are much higher than older transmission rates. It redefines how the multiplexing process occurs as you move within the different transmission levels. It also affords very robust operations capabilities, such as service restoration.

The SDH/SONET specifications define a *frame format*—that is, how the bits are packaged together to be transported over the fiber. As mentioned earlier, they define the nature of the physical interfaces (for example, couplers, light sources). They define the optical carrier line rates, or transmission levels, and they define the sorts of messages that are exchanged in order to support operations, administration, maintenance, and provisioning.

An important aspect of SDH/SONET is that it introduced the notion of a ring operation to address network survivability by handling rapid restoration. With SDH/SONET, we use a dual-counter-rotating ring. Imagine that you have four network nodes. As shown in Figure 5.8, with a dual-counter-rotating ring, you link each of these four network nodes together by using one pair of fibers; that pair of fibers becomes the primary fiber, and information will flow over it in a clockwise manner. You run another pair of fibers, which may actually be housed in the same cable as the first pair of fibers, to join the four nodes together. The second pair of fibers become the protect fiber, which is designed to carry information in a counter-clockwise manner. In theory, if a cable is cut between Node A and Node B, you can still move a message from A to B by reversing the information flow and going from A to D to C to B. This enables you to recover immediately—within 50 milliseconds—from outages that occur, for example, because a construction crew

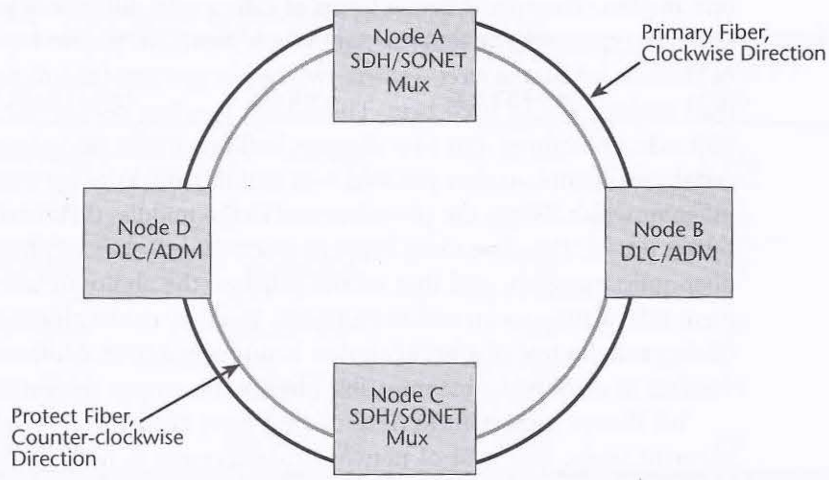


Figure 5.8 SDH/SONET ring architecture

has cut a cable. Obviously, if a major earthquake hit and all the streets were broken up, a counter-rotating ring wouldn't necessarily ensure survivability, but for smaller-scale problems, it can very adequately handle a backup. This is one of the greatest strengths of SDH/SONET and will likely keep it operational in networks for another 10 to 20 years. But these types of capabilities are also being introduced in the new generations of standards, such as WDM, and when that occurs, we will start to move away from SDH/SONET because SDH/SONET is a TDM system that does not take advantage of the fact that light can be spatially multiplexed, allowing multiple wavelengths to be carried over one fiber pair.

SDH/SONET is also important because it grooms and routes traffic. *Grooming* means that SDH/SONET selectively removes channels from a digital facility for routing to a designated remote location via another digital facility; basically, it enables you to drop and add payload flexibly. SDH/SONET also provides for performance monitoring so that you can understand the performance of the network, its components, and the congestion levels.

The SDH/SONET Signal Hierarchy

The SDH/SONET signal hierarchy deals with *optical carrier levels*, which refer to the optical aspect of the transmission—the optical pulse as it travels through the fibers. These optical pulses go through electronic muxes, and when the signal is going through these network elements, the bits are packaged in a frame for transport across the fiber. In the case of SONET, this frame is called the Synchronous Transport Signal (STS), and in SDH, the frame is called Synchronous Transport

Module (STM). Two types of rates are important in the realm of SDH/SONET: The *payload rate* refers to the capacity available to carry customer content, and the *data rate* refers to the total capacity available for customer content as well as network management information.

Table 5.3 shows the SDH/SONET signal hierarchy. You don't have to memorize all these levels, but you'll consistently encounter four or five of them in your readings that you should commit to memory.

Table 5.3 The SDH/SONET Signal Hierarchy

OC Level	SONET	SDH	Data Rate (Mbps)	Payload Rate (Mbps)
OC-1	STS-1	STM-0	51.48	50.840
OC-3	STS-3	STM-1	155.52	150.336
OC-9	STS-9	STM-3	466.56	451.008
OC-12	STS-12	STM-4	622.08	601.344
OC-18	STS-18	STM-6	933.12	902.016
OC-24	STS-24	STM-8	1,244.16	1,202.688
OC-36	STS-36	STM-12	1,866.00	1,804.032
OC-48	STS-48	STM-16	2,488.32	2,405.376
OC-96	STS-96	STM-32	4,876.64	4,810.752
OC-192	STS-192	STM-64	9,953.28	9,621.504

The levels of the SDH/SONET signal hierarchy that are most important to be familiar with are OC-1, OC-3, OC-12, OC-48, and OC-192:

- OC-1—OC-1 offers about 51Mbps and is generally used as customer access lines. Early adopter types of customers—such as universities, airports, financial institutions, large government agencies, and ISPs—would use OC-1.
- OC-3—OC-3 provides about 155Mbps. End users such as companies in the aerospace industry and high-tier ISPs would need this extensive level.
- OC-12—OC-12 provides about 622Mbps. It is another capacity toward which high-tier ISPs are moving. It was originally deployed for the metropolitan area

fiber rings built out across cities worldwide, although those rings are now moving to OC-48.

- OC-48—OC-48 offers about 2.5Gbps. This capacity has been deployed for backbone, or core, networks. Today the metropolitan area rings are moving from OC-12 to OC-48, and the backbone links are moving from OC-48 to OC-192.
- OC-192—OC-192 supports about 10Gbps and is being used for backbone networks.

There are more levels in the SDH/SONET signal hierarchy, but the ones discussed here are the ones for which equipment is currently being manufactured. We are in early stages of building new muxes that can also operate at OC-768 and that will support 40Gbps. Some people feel that electronic muxes really are not suitable for the higher data rates and that we should concentrate on moving to all-optical muxes and switches.

How do the high optical-carrier levels relate to all the lower-level signals out there—such as those from a 1.5Mbps T-1 or a 2Mbps E-1? There are mechanisms that enable us to map signal levels below DS-3 (that is, below 45Mbps) into what SDH calls *virtual containers* or what SONET calls *virtual tributaries*. A virtual container or tributary basically defines the data structure for the transport and switching of sub-51Mbps network services such as DS-1, E-1, DS-2, and E-3. Table 5.4 shows the various line rates that are supported and what existing standard each refers to. For most people, this type of detail won't make or break success in the industry, but it's important to know that a virtual tributary or virtual container can provide a highway for lower-rate data signals to coexist in high-speed optical pipes.

Table 5.4 Virtual Container/Virtual Tributary Line Rates and Standards

Virtual Container/ Virtual Tributary Level	Line Rate	Standard
VC-11/VT-1.5	1.728Mbps	DS-1/E-1
VC-2/VT-2	2.304Mbps	E-1
VT-3	3.456Mbps	DS-1C
VC-2/VT-6	6.912Mbps	DS-2
VT-6-N	$n \times 6.9$Mbps	(future)
async DS-3/VC-3	44.736/34.368Mbps	DS-3/E-3
VC-4	139.264Mbps	DS-4/E-4

In contrast to PDH, SDH/SONET is a *synchronous* infrastructure. This means that each of the network elements draws its clocking pulse from one clocking source—so everybody is marching to the beat of the same drummer. Instead of using special framing bits to delineate channels, SDH/SONET uses a special pointer bit in front of each conversation that essentially says "start of a new conversation." When it's time to drop that channel off at a customer premise, we can identify it by its pointer bit and extract it without having to disturb any of the other traffic. This reduces the overhead associated with multiplexers by a factor of 10.

SDH/SONET Muxes and Cross-Connects

SDH/SONET was built for and largely relies on fiber-optic transmission media. It also includes a variety of multiplexers and cross-connects, as well as equipment that could be placed at the customer premise. There are two main categories of SDH/SONET multiplexers (see Figure 5.9):

- **Terminal muxes**—Terminal muxes enable signals to move through the hierarchy of optical carrier levels. They act as access nodes and support current services by accepting electrical interfaces and lower-level signals, including DS-1/E-1, DS-2, and DS-3/E-3. They concentrate one or more optical carrier signals and represent one of the optical carrier levels.
- **Add/drop muxes (ADMs)**—ADMs facilitate easy dropping and adding of payload and are therefore the building blocks of the SDH/SONET network.

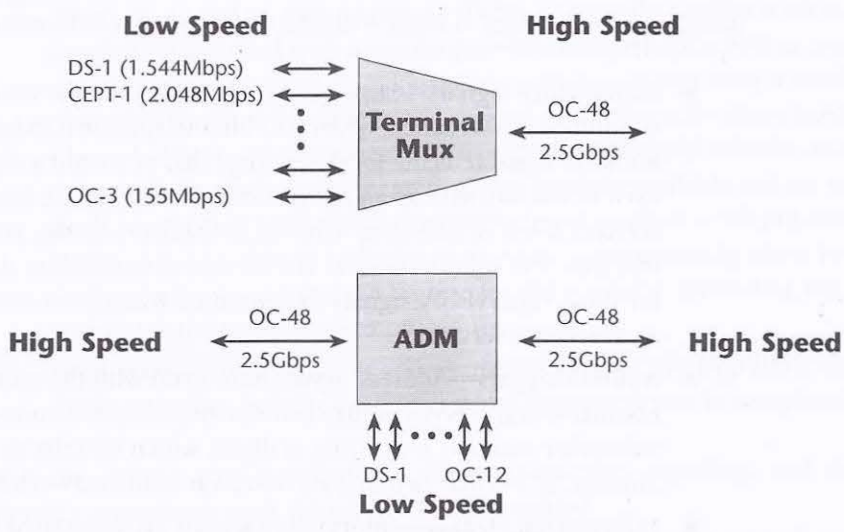


Figure 5.9 Terminal muxes versus ADMs

An add/drop mux converts one or more lower-level signals, such as T-1 or E-1 signals, into and from one of the optical carrier levels. It can drop lower-rate signals to be transported on different facilities, or it can add lower-rate signals into the higher-rate optical carrier levels, and basically it allows telcos to add and drop traffic easily and conveniently all along the network.

There are also two categories of SDH/SONET cross-connects:

- **Wideband digital cross-connects**—These terminate SDH/SONET and DS-3/E-3 signals. Switching occurs at the DS-0, DS-1/E-1, and VT/VC levels.
- **Broadband digital cross-connects**—Broadband digital cross-connects interface at the various SDH/SONET signal levels as well as the legacy DS-3/E-3 levels, but they then switch at the optical carrier levels. They can make cross-connections at DS-3/E-3, OC-1, and concatenated levels (that is, where you combine several frames of an OC-1 together). Generally, a broadband digital cross-connect is used as an SDH/SONET hub that grooms the optical carrier levels for broadband restoration purposes or for routing traffic.

Signaling Systems

This section discusses the nervous system of the network: the signaling system. A great deal of information needs to be passed back and forth between the network elements in the completion of a call and also in the servicing of specialized features. Four main types of signals handle this passing of information:

- **Supervisory signals**—Supervisory signals handle the on-hook/off-hook condition. For instance, when you lift a telephone handset (that is, go off-hook), a signal tells the local exchange that you want a dial tone, and if you exist in the database as an authenticated user, you are then delivered that service; when you hang up (that is, go back on-hook), you send a notice that says you want to remove the service. A network is always monitoring for these supervisory signals to determine when someone needs to activate or deactivate service.
- **Address signals**—Address signals have to do with the number dialed, which essentially consists of country codes, city codes, area codes, prefixes, and the subscriber number. This string of digits, which we refer to as the telephone number, is, in effect, a routing instruction to the network hierarchy.
- **Information signals**—Information signals are associated with activating and delivering various enhanced features. For instance, a call-waiting tone

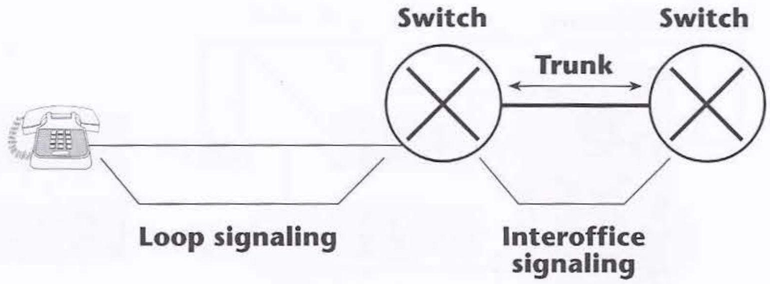


Figure 5.10 Customer loop and interoffice signaling

is an information signal, and pressing *72 on your phone might send an information signal that tells your local exchange to forward your calls.

- **Alerting signals**—Alerting signals are the ringing tones, the busy tones, and any specific busy alerts that are used to indicate network congestion or unavailability.

Signaling takes place in two key parts of the network: in the access network, where it's called *loop signaling*, and in the core, where it's called *interoffice signaling* (see Figure 5.10).

With analog loop signaling, two types of starts exist:

- **Ground start**—*Ground start* means that when you seize that line, it's immediately grounded so that no other call can potentially conflict with it. Ground start is used with a contentious system, perhaps a PBX at a corporate enterprise, to avoid collisions. For example, say you seize a trunk and place a call, and now you're in the ringing state. There are short periods of silence between ringing tones. The local exchange could mistake one of these periods of silence to mean that that trunk is available and try to send a call in over that same trunk that you're trying to place a call out over; this would cause a collision (referred to as *glare*). Consequently, when you're dealing with systems and contention for the resource, grounding the trunk up front is the most efficient procedure.
- **Loop start**—Pay telephones and residential phones use *loop start,* which means that the circuit is grounded when the connection is completed.

There are various start standards for digital subscriber signaling, and they are defined in accordance with the service being provided.

Interoffice signaling has been through several generations of signaling approaches. In the first generation, called per-trunk signaling, the complete path—

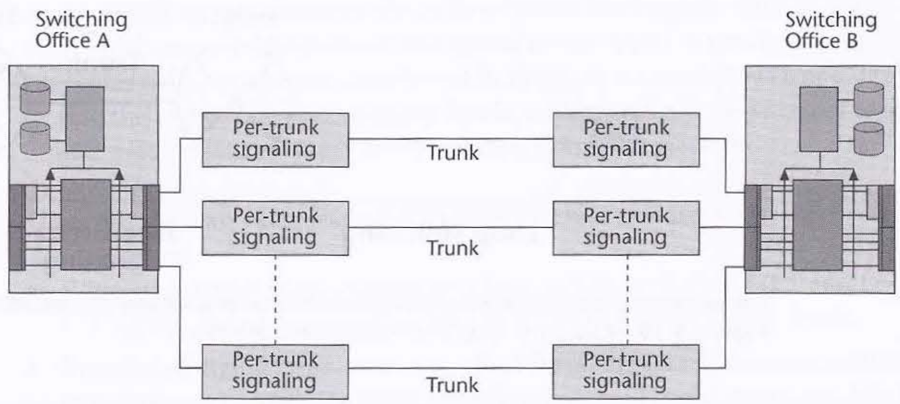


Figure 5.11 Per-trunk signaling

all the way to the destination point—is set up in order to just carry the signaling information in the first place (see Figure 5.11). This method uses trunks very inefficiently; trunks may be put into place to carry 20 or 30 ringing tones, but if nobody is on the other end to take that call, the network trunk is being used but not generating any revenue. Also, when a call is initiated and begins to progress, you can no longer send any other signaling information over that trunk; being able to pass a call-waiting tone, for instance, would not be feasible.

We have moved away from the per-trunk signaling environment to what we use today—common-channel signaling (see Figure 5.12). You can think of common-channel signaling as being a separate subnetwork over which the signaling message flows between intelligent networking components that assist in the call completion and assist in the delivery of the service logic needed to deliver the requested feature. Today, we predominantly use the ITU-T standard for common-channel signaling: SS7.

SS7 Architecture

SS7 is critical to the functioning and operation of the modern network. With SS7, a packet data network overlays and controls the operation of the underlying voice networks; signaling information is carried on an entirely different path than voice and data traffic. Signaling doesn't take a great deal of time, so we can multiplex many signaling messages over one channel, and that's why the signaling system is a packet network. The signaling system takes advantage of the efficiencies of statistical multiplexing for what is essentially bursty data. The SS7 signaling data link is a full-duplex digital transmission channel that operates at

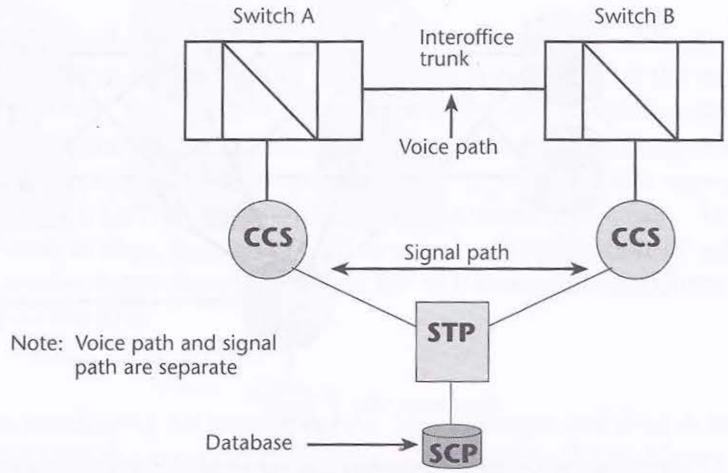


Figure 5.12 Common-channel signaling

either 56Kbps or 64Kbps, depending on the standards under which the network is operating (for example, T-carrier and J-carrier operate at 56Kbps, E-carrier operates at 64Kbps).

SS7 is an entire architecture that performs out-of-band signaling (that is, signaling in which the conversation and the signaling take place over different paths) in support of the information-exchange functions that are necessary in the PSTN, such as call establishment, billing, and routing. Database access messages convey information between toll centers and centralized databases to permit real-time access to billing-related information and other services. The SS7 architecture defines the procedures for the setup, ongoing management, and clearing of a call, and it allows you to pass along customer-related information (for example, the identity of the caller, the primary carrier chosen) that helps in routing calls. The efficiency of the network also results in faster call setup times and provides for more efficient use of the circuits when carrying the voice or data traffic. In addition, SS7 supports services that require signaling during a call as it is occurring—not in the same band as the conversation.

SS7 permits the telephone company to offer one database to several switches, thereby freeing up switch capacity for other functions, and this is what makes SS7 the foundation for INs and advanced intelligent networks (AINs). It is also the foundation for network interconnection and enhanced services. Without SS7, we would not be able to enjoy the level of interoperability we have today. SS7 is also a key to the development of new generations of services on the Internet, particularly those that support traditional telephony services. To be able to accommodate

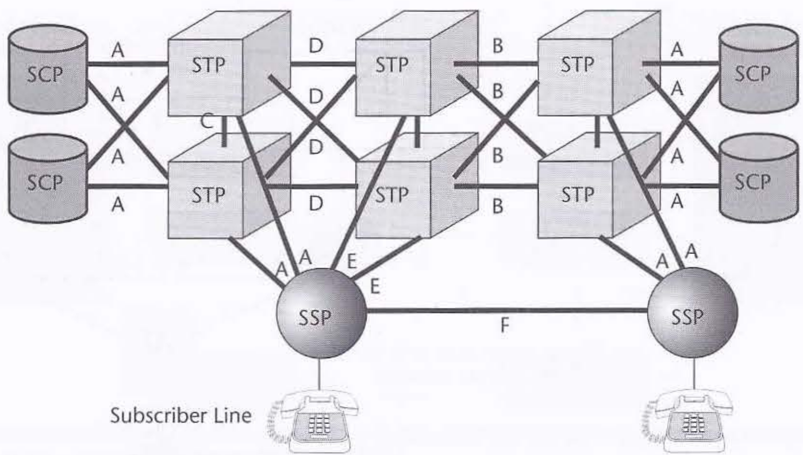


Figure 5.13 An SS7 network

features such as call forwarding, call waiting, and conference calling, you must be able to tap into the service logic that delivers those features. Until quite recently, the Internet has not been able to do this, but the year 2000 saw the introduction of SS7 gateways, which allow an interface between circuit-switched networks (with their powerful SS7 infrastructure) and the emerging packet-switched networks that need to be capable of handling the more traditional type of voice communications on a more cost-effective basis.

As Figure 5.13 shows, there are the three prerequisite components in the SS7 network: service switching points (SSPs), service control points (SCPs), and signal transfer points (STPs).

SSPs

SSPs are the switches that originate and terminate calls. They receive signals from the CPE and perform call processing on behalf of a user. The user, by dialing particular digits, triggers the network to request certain services. For instance, if you preface a number with a toll-free prefix, that toll-free arrangement triggers the local exchange, or SSP, to initiate a database lookup to determine the physical address of that toll-free number (that is, where it resides in the network). The SSP reaches into the network to find the database that can translate the toll-free number into a physical address in order to then complete the toll-free call. The SSP does this by interacting with a device called the SCP, which is discussed shortly.

SSPs are typically implemented at local exchanges, access tandem offices, or toll centers that contain the network-signaling protocols. The SSP serves as the source and destination point for the SS7 messages.

SCPs

The second key component of SS7 is SCP. This is the network element that interfaces with the SSP as well as the STP. Most importantly, the SCP is the network element that contains the network configuration and call-completion database; in other words, it contains the service logic to act on the types of calls and features the users are requesting. SCPs are centralized nodes that contain service logic—basically software and databases—for the management of the call. They provide functions such as digit translation, call routing, and verification of credit cards. The SCPs receive traffic from the SSP via the STP and return responses, based on that query, via the STP.

STPs

The STP is responsible for translating the SS7 messages and then routing those messages between the appropriate network nodes and databases. Notice in Figure 5.13 that the SCPs and the STPs are both redundant, and that the links running between them are also redundant.

SS7 and the Internet

If a network loses its signaling system, it loses the capability to complete calls, as well as to do any form of billing or passing along of management information. This makes SS7 critical. The SS7 signaling data link, as mentioned earlier in the chapter, is a full-duplex digital transmission channel that operates at either 56Kbps or 64Kbps. A variety of other SS7 links are defined as well, and each has specific uses within the signaling network:

- **A (access) links**—An A link interconnects an STP with either an SSP or an SCP. The SSP and SCP, collectively, are referred to as the *signaling endpoints*. A message sent to and from the SSPs or SCPs first goes to its home STP, which, in turn, processes or routes the message.
- **B (bridge) links, D (diagonal) links, and B/D links**—A B link connects an STP to another STP. Typically, a quad of B links interconnect peer (or primary) STPs (for example, the STPs from one network to the STPs of another network). The distinction between a B link and a D link is rather arbitrary, and such links may be referred to as *B/D links*.
- **C (cross) links**—C links interconnect mated STPs.
- **E (extended) links**—E links provide enhanced reliability by providing a set of links from the SSP to a second STP pair.
- **F (fully associated) links**—F links are links that directly connect to signaling endpoints.

Intelligent Networks

The ITU's standardization of SS7, in 1980, began the evolution toward the concept of intelligent networking. An IN includes a set of nodes that rely on widespread distribution of call-handling functions and capabilities (see Figure 5.14). Before the advent of INs, customers could have only the services and features available from their local exchanges. Their ability to demand and achieve new services from the operator was very much tied to the generation of software in the local exchange and whether it had yet incorporated the feature of interest. With INs, you can centrally place this type of service and feature logic on a node (such as an SCP), and then any switch can reach it and make use of that feature. The objective of the IN initially was to ease the introduction of new services into the network. It also provided a foundation for complex services that would be required and desirable on a networkwide basis, such as the automation of the operator-assistance function. Because of the IN and specialized peripherals—again, computing systems loaded with specific software—we no longer have to use operators to place a credit card call or a collect call.

Intelligent networking gives carriers the capability to directly develop network functionality on outboard processors connected to the switches, instead of having to be tied to their switch manufacturer and having to rely on the internal software. A main feature developed for the IN during the early and mid-1980s was *digit translation*, which was applied to toll-free number translation and VPNs. Customers could develop a unique calling plan that identified their location. They could invent their own numbering plan so that they could dial numbers that were easy

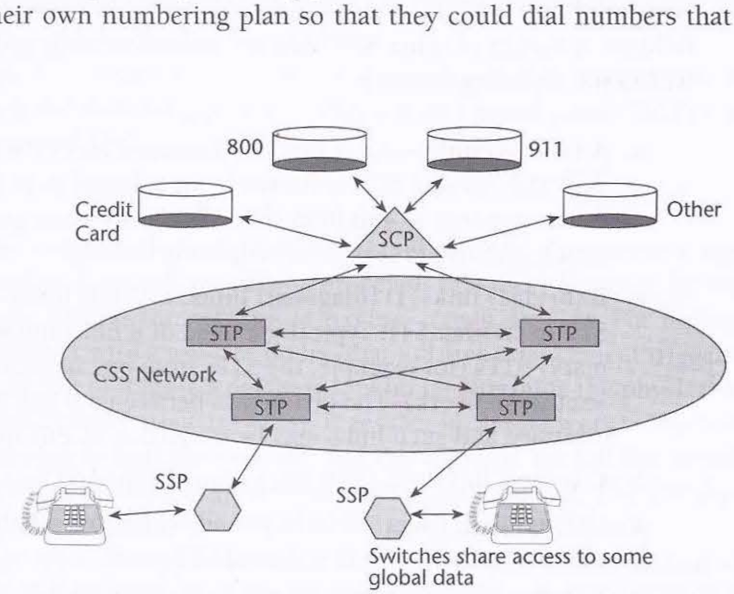


Figure 5.14 An IN

for them to remember, and in the network, the IN infrastructure would translate these private numbers into network physical addresses (for example, country code, city code, area code).

The IN also enables operator-assistance features such as eliminating credit card calling and collect calling as manual fulfillment processes. The IN also enables the identification of primary carriers (where competition exists), so that customers can select their primary carriers. *Local number portability*—which allows you to keep your own telephone number when you move to a new location—is a rather new concept that can be delivered thanks to the sophistication of this IN infrastructure. With local number portability, although your physical address will be different at your new location, you may want to keep your old phone number so your friends and colleagues can easily recall it. But for calls made with your old number to reach your new physical address, there must be translation tables in the network that can identify your correct physical address and properly route incoming calls to you.

AINs

Around the mid-1980s, Bellcore (which is now Telcordia) pioneered the second generation of INs, which we call AINs (see Figure 5.15). AINs move the service logic outside the switch and onto an independent SCP. An AIN is a service-independent network architecture that allows carriers to create and uniformly support telecom services and features via a common architectural platform, with the objective of allowing for rapid creation of customizable telecommunication services.

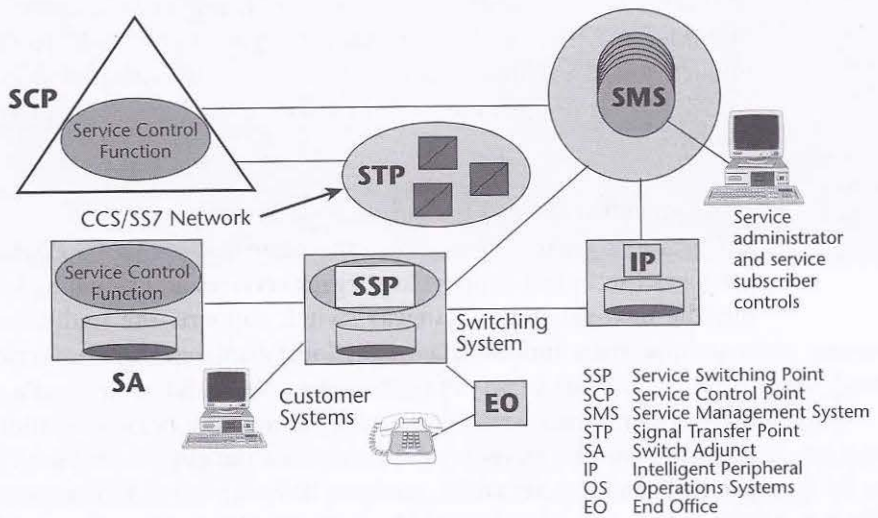


Figure 5.15 AIN architecture

An AIN is composed of intelligent nodes that are linked via SS7 to support a variety of services and advanced call-handling features across multiple vendor domains. With the introduction of the AIN architecture, a few additional components were needed. First, the service management system is a service-creation environment that facilitates the technical and customer service representatives' interface with the provisioning and network management systems. Second, intelligent peripherals are computing platforms that serve a very specific purpose but have a very widespread demand across the network (for example, voice recognition and voice synthesis capabilities to process third-party-assisted calls).

AINs can be used for a number of applications, including intelligent call routing, visitor location registration, virtual number service, voice-activated dialing, voice response, speech recognition, and text-to-speech conversion. The AIN infrastructure is critical in mobile communications. The reason you can roam across cellular networks is that IN databases are able to log whether you are present in your home network or in a visitor network, and they can identify whether you're authenticated to use the network. If you are authenticated to use the network, IN databases can identify which services should be made available to you. Virtual number services are also an important application of AINs; for example, a 700 number can identify and locate you within a footprint, rather than requiring you to be at a specific telephone to receive a call.

Next-Generation Networks

The SS7 network acts as the backbone for the AIN—it provides access to the AIN features, allows for efficient call setup and teardown, and interconnects thousands of service providers under one common signaling network. The capability to communicate with SS7 networks is essential for all service providers because SS7 networks give next-generation local exchange carriers access to an existing base of service features.

Next-Generation Network Equipment

SS7 uses new genres of equipment to ensure that packet-based telephony switching gateways can in fact support key legacy services and signaling features. For example, the next-generation gateway switch supports the traditional Class 4, or toll switch, services and the Class 5, or local exchange switch, services. It is designed to support a wide variety of traffic—data, voice, fax, multimedia, and other emerging sensory forms—over a data backbone. The next-generation gateway switch provides seamless interoperability between the circuits that network the PSTN and packet-switching networks, such as IP backbones, ATM networks, Frame Relay networks, and emerging MPLS networks. We can use these gateway switches to connect with the SS7 network and to handle the IP services that are so popular

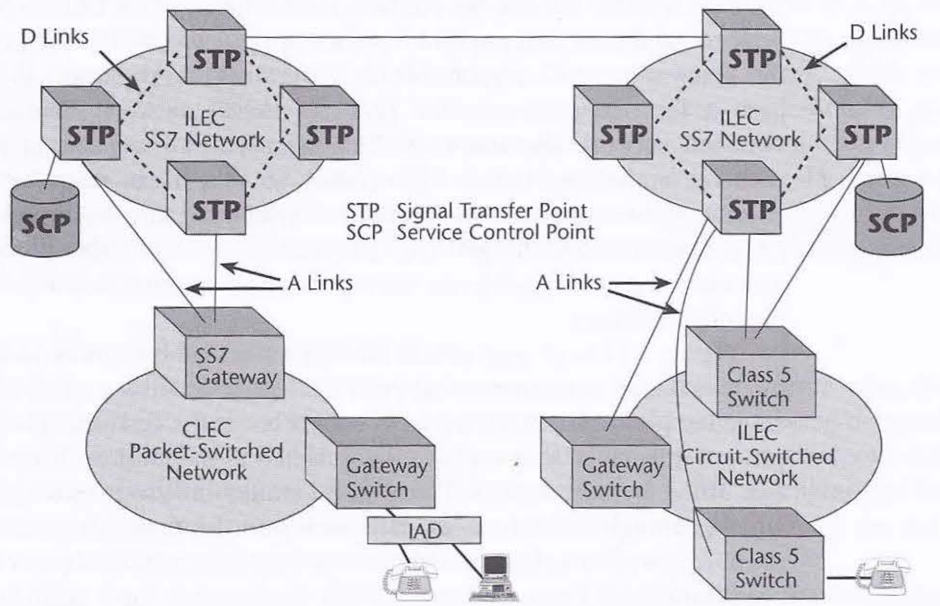


Figure 5.16 Next-generation gateway switches

Telephony Signaling Protocols and SS7

A number of telephony signaling protocols are currently used, and we don't know yet which will become the standard. Today, H.323 is found most frequently and offers the greatest range of vendor interoperability. Session Initiation Protocol (SIP), which is an Internet Engineering Task Force (IETF) standard, has a lot of support from the Internet community, and it is being included on more devices all the time. Over the next few years, we will be better able to determine which will be the dominant protocols, although IETF's SIP protocol is gaining popularity and supplanting H.323. (Signaling protocols are discussed in detail in Chapter 11, "Next-Generation Network Services.")

today. These gateway switches support a variety of telephony signaling protocols (for example, H.323, Session Initiation Protocol [SIP], Media Gateway Control Protocol [MGCP]) for communicating with the underlying SS7 architecture.

As shown in Figure 5.16, there are different forms of these next-generation gateway switches for different purposes. In the bottom right of Figure 5.16 is an ILEC and its resident circuit-switched network, which is a series of Class 5 offices at the perimeter. These Class 5 offices are connected to the SS7 network or the

STPs via A links. These Class 5 offices then connect into a CLEC's packet-switched network, and their first point of interface is a gateway switch. Among other things, this gateway switch is responsible for digitalizing and packetizing the voice to prepare it for transport over the packet-switched network. The CLEC's packet-switched network also has an SS7 gateway, which is capable of communicating with the underlying ILEC's SS7 network so that it can map the appropriate IP addresses associated with the destination telephone number, which is served by a given destination media gateway. The next-generation gateway switch, therefore, provides a means to seamlessly interoperate between two very important and existing infrastructures.

Figure 5.17 is an end-to-end view of a next-generation network. It shows an interconnected environment between the legacy circuit-switched network and the emerging packet-based networks. A subscriber at the customer premise (for example, a residence, a business site) is connected to the local exchange, known as the end office, by access lines. From there, trunks link to a media gateway switch, which, through SS7 interfaces, can reach into the underlying intelligence within the SS7 network and further add the necessary information to process the call as it's been requested. The call then goes out on a packet basis throughout a series of switches or routers (depending on what the provider is using as the backbone) and reaches a destination media gateway switch that unpackages the voice, undigitalizes it, and delivers it to the destination phone.

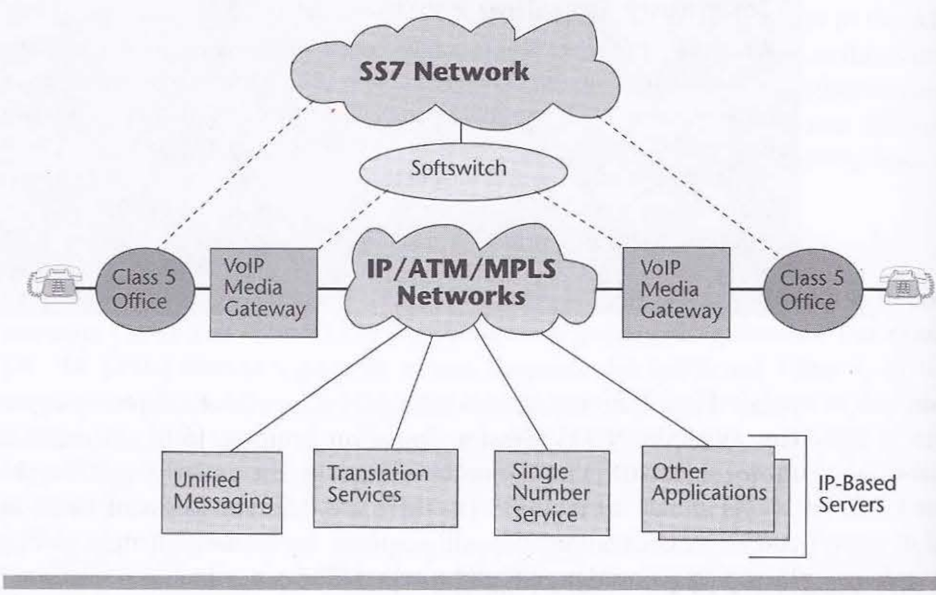


Figure 5.17 Next-generation networks

Although tremendous amounts of time and money have been spent in developing the intelligence that provides the telephony features we know today, there are still many new applications to be developed. These new applications, which are increasingly being developed for IP, include unified messaging, single-number service, and a network-type Rolodex that houses all kinds of contact and other information about people. We will be able to use databases to service calls when we have an integration or an interconnection between two networks. We will be able to provide traditional voice telephony features and introduce new generations of IP-based services.

Next-Generation Networks Versus the PSTN

The characteristics of next-generation networks are very different from what the traditional PSTN was aiming at. Next-generation networks are not being designed for just voice, data, or video. They're being designed for multimedia, and this requires capacities that are broadband in nature, networks that are engineered for extremely low and controllable latencies, and infrastructures that provide the ability to administer quality of service on a very granular level.

This book has talked about the explosion of bandwidth that's occurring because of developments in optics. As you have more and more bandwidth, it becomes cheaper and cheaper. When bandwidth becomes very inexpensive or free, a carrier needs to find other ways to generate revenue, such as by offering a large variety of value-added services (for example, reliability, priority, customer service, and encryption or security). But to administer all these services and to provide differentiated pricing, which can result in handsome revenue streams, there must be mechanisms for controlling, monitoring, and billing.

The following are important features of the next-generation network infrastructure that are covered in detail in later chapters:

- It has very fast packet switching, with capacities that we're beginning to need to measure in terabits per second (Tbps) and soon in petabits per second (1Pbps =1,000Tbps), and on its heels, in exabits per second (1Ebps = 1 billion Gbps). (See Chapter 10.)
- It places great emphasis on optical networking elements, to take advantage of the abundant bandwidth that's inherent in the visible light spectrum. (See Chapter 12, "Optical Networking.")
- Multiservice access is being created, so we will not have separate devices for voice and data as we do today, but we'll have nodes that can accommodate any traffic type. We are also creating intelligent edges; we're displacing the smarts for processing service requests, delivering features, and accommodating advanced applications by deploying them at the edge. This allows for more rapid introduction, as well as more customization of the feature sets. The core

also has to be multiservice, but it also needs to be able to differentiate between the requirements of the different traffic streams. (See Chapter 10.)

- Next-generation telephony is very important for new entrants, particularly because of the costs of deploying normal local exchanges. A regular local exchange costs in the neighborhood of US$3 million to US$5 million, whereas a media gateway will be on the order of US$100,000. For those seeking to become competitors in the local loop environment, next-generation telephony offers a very cost-effective means of gaining entry. (See Chapter 11.)
- Intelligent networking is being applied to the public data infrastructure as well as the Internet. (See Chapter 10.)
- Network operators are introducing video and multimedia elements, video servers, media farms, video compression, and decompression devices, all of which become part of what constitutes the entire communications network. (See Chapter 10.)
- Access is moving toward the broadband realm on both a wired and a wireless basis. (See Chapter 13, "Broadband Access Solutions.")

For more learning resources, quizzes, and discussion forums on concepts related to this chapter, see www.telecomessentials.com/learningcenter.