Softswitch
Architecture for VoIP

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CHAPTER 2

The Public Switched Telephone Network (PSTN)

An understanding of the workings of the *Public Switched Telephone Network* (PSTN) is best grasped by understanding its three major components: access, switching, and transport (see Figure 2-1). Each element has evolved over the 100-plus year history of the PSTN. Access pertains to how a user accesses the network. Switching refers to how a call is "switched" or routed through the network, and transport describes how a call travels or is "transported" over the network.

Access

Access refers to how the user accesses the telephone network. For most users, access is gained to the network via a telephone handset. Transmission and reception is via diaphragms where the mouthpiece converts the air pressure of voice into an analog electromagnetic wave for transmission to the switch. The earpiece performs this process in reverse. The most sophisticated aspect of the handset is its *Dual-Tone Multifrequency* (DTMF) function, which signals the switch by tones. The handset is usually connected to the central office (where the switch is located) via copper wire known as *twisted pair* because, in most cases, it consists of a twisted pair of copper wire. The stretch of copper wire connects the telephone handset to the central office. Everything that runs between the subscriber and the central office is known as *outside plant*. Telephone equipment at the subscriber end is called *customer premise equipment* (CPE).

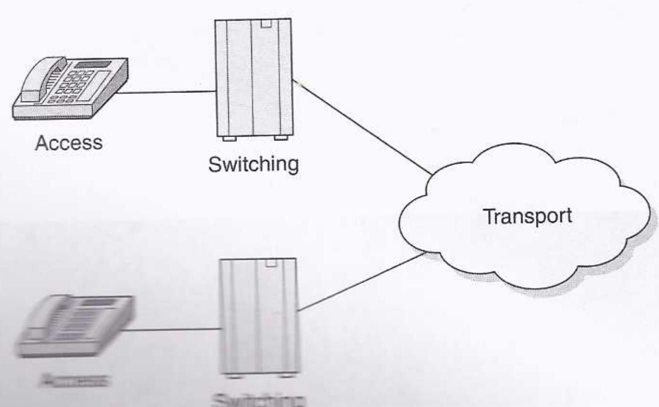


Figure 2-1 The three components of a telephone network: access, switching, and transport

Switching

The PSTN is a star network; that is, every subscriber is connected to another via at least one if not many hubs known as offices. In those offices are switches. Very simply, local offices are used for local service connections and tandem offices for long-distance service. Local offices, better known as central offices, use Class 5 switches, and tandem offices use Class 4 switches. Figure 2-2 details the relationship between Class 4 and 5 switches. A large city might have several central offices. Denver (population 2 million), for example, is estimated to have almost 40 central offices. Central offices in a large city often take up much of a city block and are recognizable as large brick buildings with no windows.

The first telephone switches were human. Taking a telephone handset off hook alerted a telephone operator of the caller's intention to place a call. The caller informed the operator of their intended called party and the operator set up the call by manually connecting the two parties.

Mechanical switching is credited to Almon Stowger, an undertaker in Kansas City, Missouri, who realized he was losing business when families of the deceased picked up their telephone handset and simply asked the operator to connect them with "the undertaker." The sole operator in this

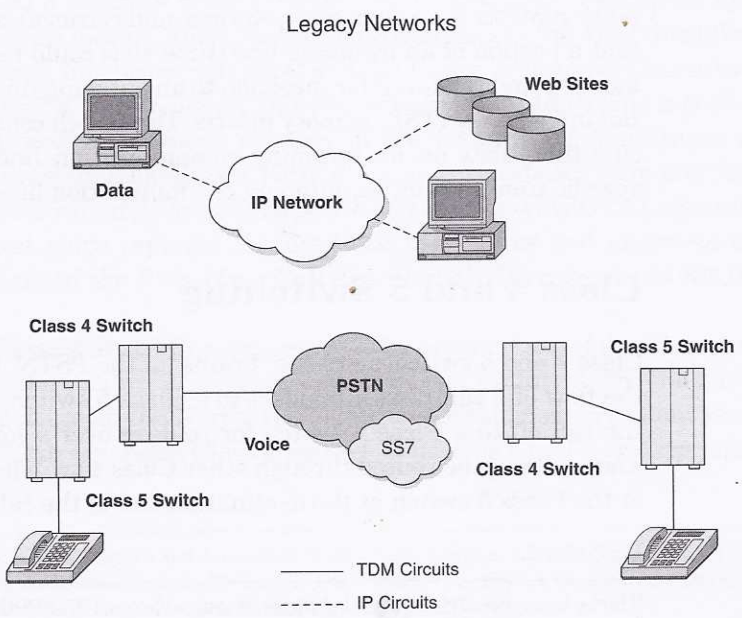


Figure 2-2 The traditional relationship of Class 4, Class 5, and data networks

town was engaged to an undertaker competing with Stowger. This competing undertaker had promised to marry the operator once he had the financial means to do so. The operator, in turn, was more than willing to help him achieve that goal.

Stowger, realizing he was losing business to his competitor due to the intercession of the telephone operator, proceeded to invent an electromechanical telephone handset and switch that enabled the caller, by virtue of dialing the called party's number, to complete the connection without human intervention. Telephone companies realized the enormous savings in manpower (or womanpower as the majority of telephone operators at the time were women) by automating the call setup and takedown process. Stowger switches (also known as crossbar switches) can still be found in the central offices of rural America and lesser developed countries.

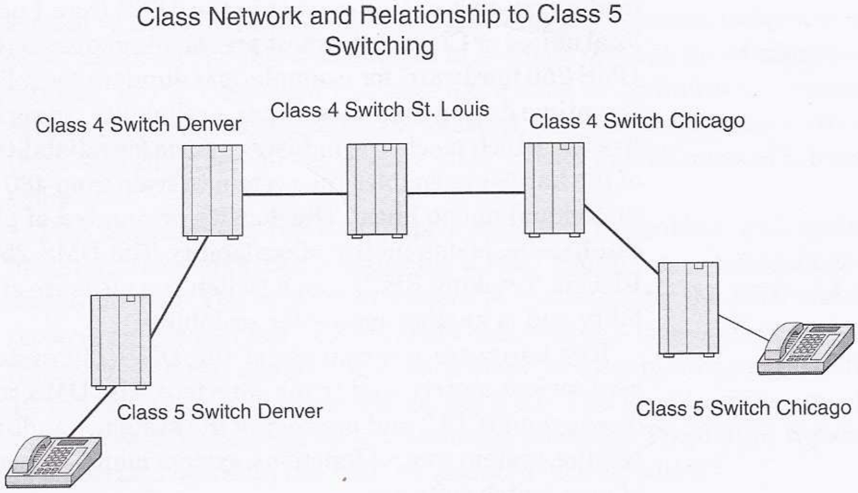
Stowger's design remained the predominant telephone switching technology until the mid-1970s. Beginning in the '70s, switching technology evolved to mainframe computers; that is, no moving parts were used and the computer telephony applications made such features as conferencing and call forwarding possible. In 1976, AT&T installed its first *#4 Electronic Switching System* (4ESS) tandem switch. This was followed shortly thereafter with the 5ESS as a central office switch. ESS central office switches did not require a physical connection between incoming and outgoing circuits. Paths between the circuits consisted of temporary memory locations that enabled the temporary storage of traffic. For an ESS system, a computer controls the assignment, storage, and retrieval of memory locations so that a portion of an incoming line (time slot) could be stored in temporary memory and retrieved for insertion to an outgoing line. This is called a *time slot interchange* (TSI) memory matrix. The switch control system maps specific time slots on an incoming communication line (such as a DS3) to specific time slots on an outgoing communication line.[1]

Class 4 and 5 Switching

Class 4 and 5 switches are the "brains" of the PSTN. Figure 2-3 illustrates the flow of a call from a handset to a Class 5 switch, which in turn hands the call off to a Class 4 switch for routing over a long-distance network. That call may be routed through other Class 4 switches before terminating at the Class 5 switch at the destination end of the call before being passed

[1]Harte, Lawrence. *Telecom Made Simple*. Fuquay-Varina, NC: APDG Publishing, 2002.

Figure 2-3
Relationship of Class 4 and 5 switching



on to the terminating handset. Class 5 switches handle local calling and Class 4 switches handle long-distance calls. The performance metrics for the Class 4 and 5 have been reliability, scalability, *quality of service* (QoS), signaling, and features.

Class 4 and 5 Architecture One reason for the reputation of Class 4 and 5 switches being reliable is that they have been tested by time in the legacy market. Incremental improvements to the 4ESS included new interfaces, hardware, software, and databases to improve *Operations, Administration, Maintenance, and Provisioning* (OAM&P). The inclusion of the 1A processor improved memory in the 4 and 5ESS mainframe, allowing for translation databases. Ultimately, those databases were interfaced with the *Centralized Automatic Reporting on Trunks* (CAROT). Later, integrated circuit chips replaced the magnetic core stores and improved memory and boosted the *Busy Hour Call Attempt* (BHCA) capacity to 700,000 BHCAs.[2]

Class 4 and 5 Components The architecture of the Class 4 and 5 switch is the product of 25-plus years of design evolution. For the purposes of this discussion, the Nortel DMS-250, one of the most prevalent products in the North American Class 4 market, is used as a real-world example. The other

[2]Chapuis, Robert, and Amos Joel. "In the United States, AT&T's Digital Switch Entry No. 4 ESS, First Generation Time Division Digital Switch." *Electronics, Computers, and Telephone Systems.* New York: North Holland Publishing, 1990, p. 337–338.

leading product in this market is the 4ESS from Lucent Technologies. For local offices or Class 5, the most prevalent product is the 5ESS from Lucent. DMS-250 hardware, for example, is redundant for reliability and decreased downtime during upgrades. It has a reliability rating of 99.999 percent (the five 9s), which meets the industry metric for reliability. The modular design of the hardware enables the system to scale from 480 to over 100,000 DS0s (individual phone lines). The density, or number of phone lines the switch can handle, is one metric of scalability. The DMS-250 is rated at 800,000 BHCAs. Tracking BHCAs on a switch is a measure of call-processing capability and is another metric for scalability.

Key hardware components of the DMS-250 system include the DMS core, switch matrix, and trunk interface. The DMS core is the *central processing unit* (CPU) and memory of the system, handling high-level call processing, system control functions, system maintenance, and the installation of new switch software.

The DMS-250 switching matrix switches calls to their destinations. Its nonblocking architecture enables the switch to communicate with peripherals through fiber optic connections. The trunk interfaces are peripheral modules that form a bridge between the DMS-250 switching matrix and the trunks it serves. They handle voice and data traffic to and from customers and other switching systems. DMS-250 trunk interfaces terminate DS-1, *Integrated Services Digital Network* (ISDN) *Primary Rate Interface* (PRI), X.75/X.75 packet networking, and analog trunks. They also accommodate test and service circuits used in office and facility maintenance. It is important to note that the Class 4 switching matrix is a part of the centralized architecture of the Class 4. Unlike the media gateways in a softswitch solution, it must be collocated with the other components of the Class 4.

DMS-250 billing requires the maintenance of real-time, transaction-based billing records for many thousands of customers and scores of variants in service pricing. The DMS-250 system automatically provides detailed data, formats the data into call detail records, and constructs bills.[3]

Private Branch Exchange (PBX)

As the name would imply, a *private branch exchange* (PBX) is a switch owned and maintained by a business with many (20 or more) users. A key

[3]Nortel Networks. "Product Service Information-DMS300/250 System Advantage." www.nortel.com, 2001.

system is used by smaller offices. PBXs and key systems today are computer based and enable soft changes to be made through an administration terminal or PC. Unless the business has a need for technical telecommunications personnel on staff for other reasons, the business will normally contract with their vendor for routine adds, moves, and changes of telephone equipment.

PBX systems are often equipped with key assemblies and systems, including voice mail, call accounting, a local maintenance terminal, and a dial-in modem. The voice mail system is controlled by the PBX and only receives calls when the PBX software determines a message can be left or retrieved. The call accounting system receives system message details on all call activities that occur within the PBX. The local terminal provides onsite access to the PBX for maintenance activities. The dial-in capability also provides access to the PBX for maintenance activities.[4]

Centrex

After PBXs caught on in the industry, local exchange carriers began to lose some of their more lucrative business margins. The response to the PBX was Centrex. Centrex is a service offered by a local telephone service provider (primarily to businesses) that enables the customer to have features that are typically associated with a PBX. These features include three- or four-digit dialing, intercom features, distinctive line ringing for inside and outside lines, voice mail, call-waiting indication, and others. Centrex services flourished and still have a place for many large, dispersed entities such as large universities and major medical centers.

One of the major selling points for Centrex is the lack of capital expenditure up front. That, coupled with the reliability associated with Centrex due to its location in the telephone company central office, has kept Centrex as the primary telephone system in many of the businesses referenced previously. PBXs, however, have cut into what was once a lucrative market for the telephone companies and are now the rule rather than the exception for business telephone service. This has come about because of inventive ways of funding the initial capital outlay and the significantly lower operating cost of a PBX versus a comparable Centrex offering.

[4] Harte, Lawrence. *Telecom Made Simple*. Fuquay-Varina, NC: APDG Publishing, 2002.

Multiplexing

The earliest approach to getting multiple conversations over one circuit was *frequency division multiplexing* (FDM). FDM was made possible by the vacuum tube where the range of frequencies was divided into parcels that were distributed among subscribers. In the first FDM architectures, the overall system bandwidth was 96 kHz. This 96 kHz could be divided among a number of subscribers into, for example, 5 kHz per subscriber, meaning almost 20 subscribers could use this circuit.

FDM is an analog technology and suffers from a number of shortcomings. It is susceptible to picking up noise along the transmission path. This FDM signal loses its power over the length of the transmission path. FDM requires amplifiers to strengthen the signal over that path. However, the amplifiers cannot separate the noise from the signal and the end result is an amplified noisy signal.

The improvement over FDM was *time division multiplexing* (TDM). TDM was made possible by the transistor that arrived in the market in the 1950s and 1960s. As the name would imply, TDM divides the *time* rather than the frequency of a signal over a given circuit. Although FDM was typified by "some of the frequency all of the time," TDM is "all of the frequency some of the time." TDM is a digital transmission scheme that uses a small number of discrete signal states. Digital carrier systems have only three valid signal values: one positive, one negative, and zero. Everything else is registered as noise. A repeater, known as a regenerator, can receive a weak and noisy digital signal, remove the noise, reconstruct the original signal, and amplify it before transmitting the signal onto the next segment of the transmission facility. Digitization brings with it the advantages of better maintenance and troubleshooting capability, resulting in better reliability. Also, a digital system enables improved configuration flexibility.

TDM has made the multiplexer, also known as the channel bank, possible. In the United States, the multiplexer or "mux" enables 24 channels per single four-wire facility. This is called a T-1, DS1, or T-Carrier. Outside North America and Japan, it is 32 channels per facility and known as E1. These systems came on the market in the early 1960s as a means to transport multiple channels of voice over expensive transmission facilities.

Voice Digitization via Pulse Code Modulation

One of the first processes in the transmission of a telephone call is the conversion of an analog signal into a digital one. This process is called *pulse* *code modulation* (PCM). This is a four-step process consisting of *pulse amplitude modulation* (PAM) sampling, companding, quantization, and encoding.

Pulse Amplitude Modulation (PAM) The first stage in PCM is known as PAM. In order for an analog signal to be represented as a digitally encoded bitstream, the analog signal must be sampled at a rate that is equal to twice the bandwidth of the channel over which the signal is to be transmitted. As each analog voice channel is allocated 4 kHz of bandwidth, each voice signal is sampled at twice that rate, or 8,000 samples per second. In a T-Carrier, the standard in North America and Japan, each channel is sampled every one eight-thousandth of a second in rotation, resulting in the generation of 8,000 pulse amplitude samples from each channel every second. If the sampling rate is too high, too much information is transmitted and bandwidth is wasted. If the sampling rate is too low, aliasing may result. Aliasing is the interpretation of the sample points as a false waveform due to the lack of samples.

Companding The second process of PCM is companding. Companding is the process of compressing the values of the PAM samples to fit the nonlinear quantizing scale that results in bandwidth savings of more than 30 percent. It is called companding as the sample is compressed for transmission and expanded for reception.[5]

Quantization The third stage in PCM is quantization. In quantization, values are assigned to each sample within a constrained range. In using a limited number of bits to represent each sample, the signal is quantized. The difference between the actual level of the input analog signal and the digitized representation is known as quantization noise. Noise is a detraction to voice quality and it is necessary to minimize noise. The way to do this is to use more bits, thus providing better granularity. In this case, an inevitable trade-off takes place bewteen bandwidth and quality. More bandwidth usually improves signal quality, but bandwidth costs money. Service providers, whether using TDM or *Voice over IP* (VoIP) for voice transmission will always have to choose between quality and bandwidth. A process known as nonuniform quantization involves the usage of smaller

[5]Shepard, Steven. *SONET/SDH Demystified.* New York: McGraw-Hill, 2001. p. 15–21.

quantization steps at smaller signal levels and larger quantization steps for larger signal levels. This gives the signal greater granularity or quality at low signal levels and less granularity (quality) at high signal levels. The result is to spread the signal-to-noise ratio more evenly across the range of different signals and to enable fewer bits to be used compared to uniform quantization. This process results in less bandwidth being consumed than for uniform quantization.[6]

Encoding The fourth and final process in PCM is encoding the signal. This is performed by a *codec* (coder/decoder). Three types of codecs exist: waveform codecs, source codecs (also known as vocoders), and hybrid codecs. Waveform codecs sample and code an incoming analog signal without regard to how the signal was generated. Quantized values of the samples are then transmitted to the destination where the original signal is reconstructed, at least to a certain approximation of the original. Waveform codecs are known for simplicity with high-quality output. The disadvantage of waveform codecs is that they consume considerably more bandwidth than the other codecs. When waveform codecs are used at low bandwidth, speech quality degrades markedly.

Source codecs match an incoming signal to a mathematical model of how speech is produced. They use the linear predictive filter model of the vocal tract, with a voiced/unvoiced flag to represent the excitation that is applied to the filter. The filter represents the vocal tract and the voice/unvoiced flag represents whether a voiced or unvoiced input is received from the vocal chords. The information transmitted is a set of model parameters as opposed to the signal itself. The receiver, using the same modeling technique in reverse, reconstructs the values received into an analog signal.

Source codecs also operate at low bit rates and reproduce a synthetically sounding voice. Using higher bit rates does not result in improved voice quality. Vocoders (source codecs) are most widely used in private and military applications.

Hybrid codecs are deployed in an attempt to derive the benefits from both technologies. They perform some degree of waveform matching while mimicking the architecture of human speech. Hybrid codecs provide better voice quality at low bandwidth than waveform codecs. Table 2-1 provides an outline of the different ITU codec standards and Table 2-2 lists the parameters of the voice codecs.

[6]Collins, Daniel. *Carrier Grade Voice Over IP.* New York: McGraw-Hill, 2001. p. 95–96.

Table 2-1

Descriptions of voice codecs (ITU)

ITU Standard	Description
P.800	A subjective rating system to determine the *Mean Opinion Score* (MOS) or the quality of telephone connections
G.114	A maximum one-way delay end to end for a VoIP call (150 ms)
G.165	Echo cancellers
G.168	Digital network echo cancellers
G.711	PCM of voice frequencies
G.722	7 kHz audio coding within 64 Kbps
G.723.1	A dual-rate speech coder for multimedia communications transmitting at 5.3 and 6.3 Kbps
G.729	Coding for speech at 8 Kbps using *conjugate-structure algebraic code-excited linear-prediction* (CS-ACELP)
G.729A	Annex A reduced complexity 8 Kbps CS-ACELP speech codec
H.323	A packet-based multimedia communications system
P.861	Specifies a model to map actual audio signals to their representations inside the human head
Q.931	Digital subscriber signaling system number 1 ISDN user-network interface layer 3 specification for basic call control

Table 2-2

Parameters of voice codecs

Standard	Data rate (Kbps)	Delay (ms)	MOS	Codec
G.711	64	0.125	4.8	Waveform
G.721, G.723, G.726	16,24,32,40	0.125	4.2	
G.728	16	2.5	4.2	
G.729	8	10	4.2	
G.723.1	5.3, 6.3	30	3.5, 3.98	

Popular Speech Codecs Codecs are best known for the sophisticated compression algorithms they introduce into a conversation. Bandwidth costs service providers money. The challenge for many service providers is to squeeze as much traffic as possible into one "pipe," that is one channel. Most codecs allow multiple conversations to be carried on one 64 kbps channel. There is an inevitable trade off in compression for voice quality in the conversation. The challenge for service providers is to balance the economics of compression with savings in bandwidth costs.

G.711 G.711 is the best-known coding technique in use today. It is a waveform codec and is the coding technique used in circuit-switched telephone networks all over the world. G.711 has a sampling rate of 8,000 Hz. If uniform quantization were to be used, the signal levels commonly found in speech would be such that at least 12 bits per sample would be needed, giving it a bit rate of 96 Kbps. Nonuniform quantization is used with eight bits used to represent each sample. This quantization leads to the well-known 64 Kbps DS0 rate. G.711 is often referred to as PCM. G.711 has two variants: A-law and mu-law. Mu-law is used in North America and Japan where T-Carrier systems prevail. A-law is used everywhere else in the world. The difference between the two is the way nonuniform quantization is performed. Both are symmetrical at approximately zero. Both A-law and mu-law offer good voice quality with a MOS of 4.3, with 5 being the best and 1 being the worst. Despite being the predominant codec in the industry, G.711 suffers one significant drawback; it consumes 64 Kbps in bandwidth. Carriers seek to deliver voice quality using little bandwidth, thus saving on operating costs.

G.728 LD-CELP *Code-Excited Linear Predictor* (LD-CELP) codecs implement a filter and contain a codebook of acoustic vectors. Each vector contains a set of elements in which the elements represent various characteristics of the excitation signal. CELP coders transmit to the receiving end a set of information determining filter coefficients, gain, and a pointer to the chosen excitation vector. The receiving end contains the same code book and filter capabilities so that it reconstructs the original signal. G.728 is a backward-adaptive coder as it uses previous speech samples to determine the applicable filter coefficients. G.728 operates on five samples at one time. That is, 5 samples at 8,000 Hz are needed to determine a codebook vector and filter coefficients based upon previous and current samples. Given a coder operating on five samples at a time, a delay of less than 1 millisecond is the result. Low delay equals better voice quality.

The G.728 codebook contains 1,024 vectors, which requires a 10-bit index value for transmission. It also uses 5 samples at a time taken at a rate of 8,000 per second. For each of those 5 samples, G.728 results in a transmitted bit rate of 16 Kbps. Hence, G.728 has a transmitted bit rate of 16 Kbps. Another advantage here is that this coder introduces a delay of 0.625 milliseconds with an MOS of 3.9. The difference from G.711's MOS of 4.3 is imperceptible to the human ear. The bandwidth savings between G.728's 16 Kbps per conversation and G.711's 64 Kbps per conversation make G.728 very attractive to carriers given the savings in bandwidth.

G.723.1 ACELP G.723.1 ACELP can operate at either 6.3 Kbps or 5.3 Kbps with the 6.3 Kbps providing higher voice quality. Bit rates are contained in the coder and decoder, and the transition between the two can be made during a conversation. The coder takes a bank-limited input speech signal that is sampled a 8,000 Hz and undergoes uniform PCM quantization, resulting in a 16-bit PCM signal. The encoder then operates on blocks or frames of 240 samples at a time. Each frame corresponds to 30 milliseconds of speech, which means that the coder causes a delay of 30 milliseconds. With a look-ahead delay of 7.5 milliseconds, the total algorithmic delay is 37.5 milliseconds. G.723.1 gives an MOS of 3.8, which is highly advantageous in regards to the bandwidth used. The delay of 37.5 milliseconds one way does present an impediment to good quality, but the round-trip delay over varying aspects of a network determines the final delay and not necessarily the codec used.

G.729 G.729 is a speech coder that operates at 8 Kbps. This coder uses input frames of 10 milliseconds, corresponding to 80 samples at a sampling rate of 8,000 Hz. This coder includes a 5-millisecond look-ahead, resulting in an algorithmic delay of 15 milliseconds (considerably better than G.723.1). G.729 uses an 80-bit frame. The transmitted bit rate is 8 Kbps. Given that it turns in an MOS of 4.0, G.729 is perhaps the best trade-off in bandwidth for voice quality. The previous paragraphs provide an overview of the multiple means of maximizing the efficiency of transport via the PSTN. We find today that TDM is almost synonymous with circuit switching. Telecommunications engineers use the term TDM to describe a circuit-switched solution. A 64 Kbps G.711 codec is the standard in use on the PSTN. The codecs described in the previous pages apply to VoIP as well. VoIP engineers seeking to squeeze more conversations over valuable bandwidth have found these codecs very valuable in compressing VoIP conversations over an IP circuit.[7]

Signaling

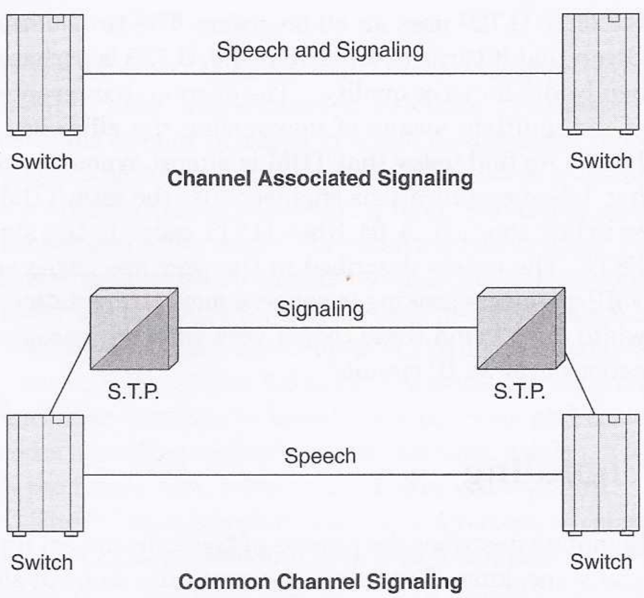
Signaling describes the process of how calls are set up and torn down. Generally speaking, there are three main functions of signaling: supervision, alerting, and addressing. Supervision refers to monitoring the status of a line or circuit to determine if there is traffic on the line. Alerting deals with the ringing of a phone indicating the arrival of an incoming call. Addressing is the routing of a call over a network. As telephone networks matured,

[7]Ibid.

individual nations developed their proprietary signaling systems. Ultimately, there become a signaling protocol for every national phone service in the world. Frankly, it is a miracle that international calls are ever completed given the complexity of interfacing national signaling protocols.

Signaling System 7 (SS7) For much of the history of circuit-switched networks, signaling followed the same path as the conversation. This is called *Channel-Associated Signaling* (CAS) and is still in wide use today. R1 *Multifrequency* (MF) used in North American markets and *R2 Multi-Frequency Compelled* (RFC) used elsewhere in the world are the best examples of this. Another name for this is in-channel signaling. The newer technology for signaling is called *Common Channel Signaling* (CCS), also known as out-of-band signaling. CCS uses a separate transmission path for call signaling and not the bearer path for the call. This separation enables the signaling to be handled in a different manner to the call. This enables signaling to be managed by a network independent of the transport network. Figure 2-4 details the difference between CAS and CCS.

Figure 2-4
CAS and CCS



[8]Stallings, William. *ISDN and Broadband ISDN with Frame Relay and ATM.* New York: Prentice Hall, 1995. p.292.

Signaling System 7 (SS7) is the standard for CCS with many national variants throughout the world (such as Mexico's NOM-112). It routes control messages through the network to perform call management (setup, maintenance, and termination) and network management functions. Although the network being controlled is circuit switched, the control signaling is implemented using packet-switching technology. In effect, a packet-switched network is overlaid on a circuit-switched network in order to operate and control the circuit-switched network. SS7 defines the functions that are performed in the packet-switched network but does not dictate any particular hardware implementation.[8]

The SS7 network and protocol are used for the following:

- Basic call setup, management, and tear down
- Wireless services such as *personal communications services* (PCS), wireless roaming, and mobile subscriber authentication
- *Local number portability* (LNP)
- Toll-free (800/888) and toll (900) wireline services
- Enhanced call features such as call forwarding, calling party name/number display, and three-way calling
- Efficient and secure worldwide telecommunications

Signaling Links SS7 messages are exchanged between network elements over 56 or 64 Kbps bidirectional channels called signaling links. Signaling occurs out of band on dedicated channels rather than in-band on voice channels. Compared to in-band signaling, out-of-band signaling provides faster call setup times (compared to in-band signaling using MF signaling tones), more efficient use of voice circuits, support for *Intelligent Network* (IN) services that require signaling to network elements without voice trunks (such as database systems), and improved control over fraudulent network usage.

Signaling Points Each signaling point in the SS7 network is uniquely identified by a numeric point code. Point codes are carried in signaling messages exchanged between signaling points to identify the source and destination of each message. Each signaling point uses a routing table to select the appropriate signaling path for each message. Three kinds of signaling points are used in the SS7 network: *service switching points* (SSP), *signal transfer points* (STP), and *service control points* (SCP), as shown in Figure 2-5.

SSPs are switches that originate, terminate, or tandem calls. An SSP sends signaling messages to other SSPs to set up, manage, and release voice

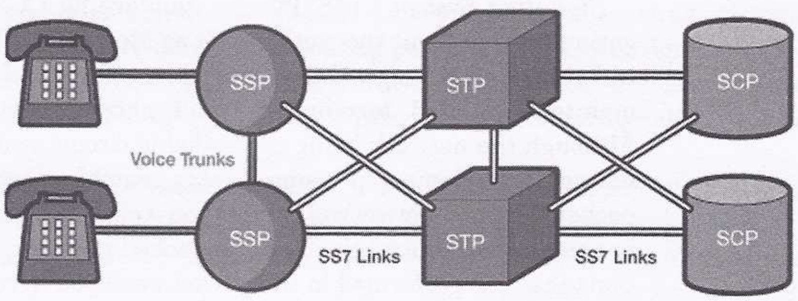


Figure 2-5
SS7 signaling points (Source: Performance Technologies)

circuits required to complete a call. An SSP may also send a query message to a centralized database (an SCP) to determine how to route a call (such as a toll-free 1-800/888 call in North America). An SCP sends a response to the originating SSP containing the routing number(s) associated with the dialed number. An alternate routing number may be used by the SSP if the primary number is busy or the call is unanswered within a specified time. Actual call features vary from network to network and from service to service.

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

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

SS7 Signaling Link Types Signaling links are logically organized by link type (A through F) according to their use in the SS7 signaling network (see Figure 2-6 and Table 2-3).

Figure 2-6
SS7 signaling link types (Source: Performance Technologies)

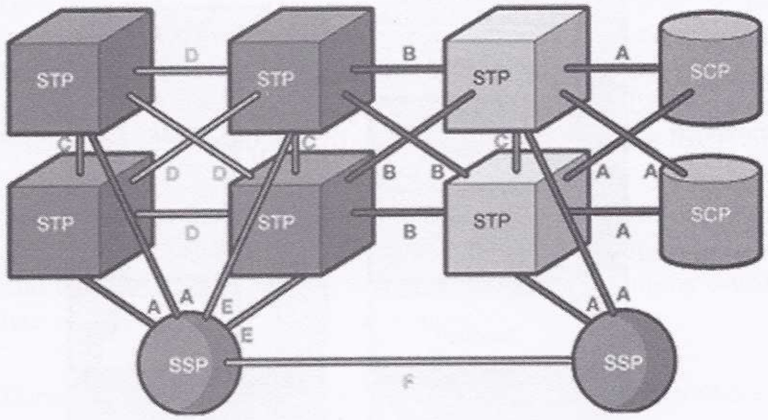


Table 2-3

Descriptions of SS7 links

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

Source: Performance Technologies

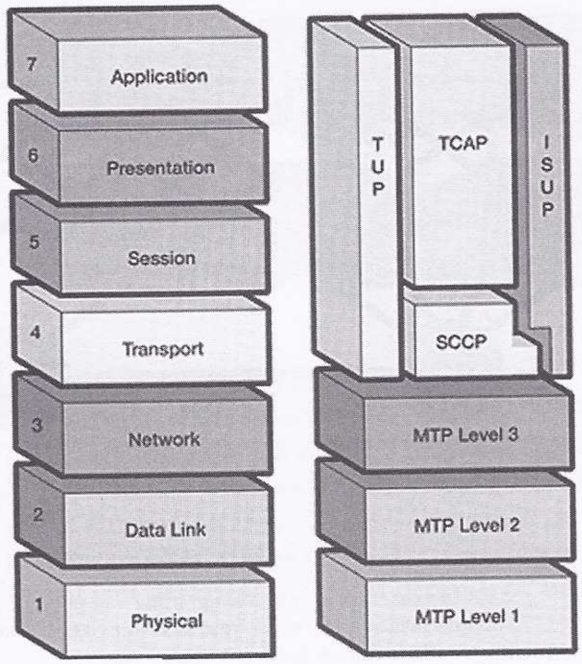


Figure 2-7 The OSI Reference Model and the SS7 protocol stack (Source: Performance Technologies)

SS7 Protocol Stack The hardware and software functions of the SS7 protocol are divided into functional abstractions called levels. These levels map loosely to the *Open Systems Interconnect* (OSI) seven-layer model defined by the *International Standards Organization* (ISO), as shown in Figure 2-7.

Message Transfer Part The *Message Transfer Part* (MTP) is divided into three levels. The lowest level, MTP level 1, is equivalent to the OSI physical layer. MTP level 1 defines the physical, electrical, and functional characteristics of the digital signaling link. Physical interfaces defined include E-1 (2,048 Kbps; 32 64-Kbps channels), DS-1 (1,544 Kbps; 24 64-Kbps channels), V.35 (64 Kbps), DS-0 (64 Kbps), and DS-0A (56 Kbps). MTP level 2 ensures accurate end-to-end transmission of a message across a signaling link. Level 2 implements flow control, message sequence validation, and error checking. When an error occurs on a signaling link, the message (or set of messages) is retransmitted. MTP level 2 is equivalent to the OSI data link layer.

MTP level 3 provides message routing between signaling points in the SS7 network. MTP level 3 reroutes traffic away from failed links and signaling points, and it controls traffic when congestion occurs. MTP level 3 is equivalent to the OSI network layer.

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

Telephone User Part (TUP) In some parts of the world (such as China and Brazil), the *Telephone User Part* (TUP) is used to support basic call setup and teardown. TUP handles analog circuits only. In many countries, ISUP has replaced TUP for call management.

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

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

The Advanced Intelligent Network (AIN)

How are features delivered? In one concept, features are made possible by the *Advanced Intelligent Network* (AIN) and SS7. The AIN is a telephone network architecture that separates service logic from switching equipment, enabling new services to be added without having to redesign switches to support new services. It encourages competition among service providers as it makes it easier for a provider to add services, and it offers customers more service choices. Developed by Bell Communications Research, AIN is recognized as an industry standard in North America.

The AIN was a concept promoted by large telephone companies throughout the 1980s to promote their architecture for the 1990s and beyond. Two consistent themes characterize the AIN. One is that the network can control the routing of calls within it from moment to moment based on some criteria other than that of finding a path through the network for the call based on the dialed number. The other is that the originator or receiver of the call can inject intelligence into the network and affect the flow of the call. That intelligence is provided through the use of databases in a network.

The foundation of the AIN architecture is SS7 (see Figure 2-8). SS7 enables a wide range of services to be provided to the end-user. An SCP is a network entity that contains additional logic and that can be used to offer advanced services. To use the service logic of the SCP, a switch needs to contain functionality that will enable it to act upon instructions from the SCP. In such a case, the switch is known as an SSP. If a particular service needs to be invoked, the SSP sends a message to the SCP asking for instruction. The SCP, based upon data and service logic that is available, will tell the SSP which actions need to be taken.[9]

How does AIN work? A telephone caller dials a number that is received by a switch at the telephone company central office. The switch, also known as the signaling point, forwards the call over an SS7 network to an SCP where the service logic is located. The SCP identifies the service requested from part of the number that was dialed and returns information about how to handle the call to the signaling point. Examples of services that the SCP might provide include area number calling services, disaster recovery services, do not disturb services, and 800 toll-free and 5-digit extension dialing services.

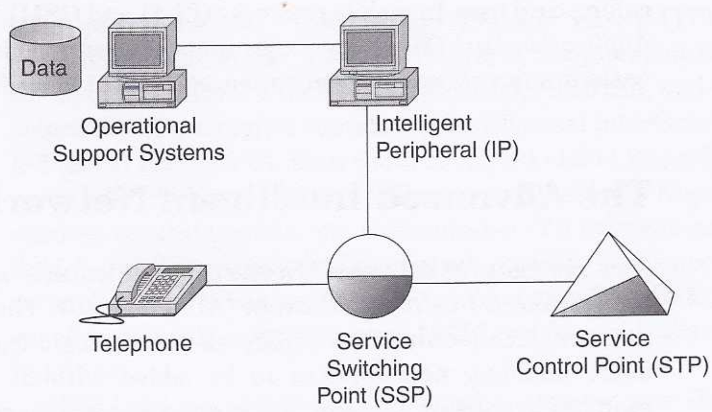


Figure 2-8
AIN release 1 architecture (Source: Telcordia)

[9]Collins, Michael. *Carrier Grade Voice Over IP.* New York: McGraw-Hill, 2001. p. 311.

In some cases, the call can be handled more quickly by an *intelligent peripheral* attached to the SSP over a high-speed connection. For example, a customized voice announcement can be delivered in response to the dialed number or a voice call can be analyzed and recognized. In addition, an adjunct facility can be added directly to the SSP for a high-speed connection to additional, undefined services.[10]

SCPs have two complementary tasks. First, they host the application functionality on which service logic is installed after services are created. Secondly, the SCP controls functionalities developed by SCP vendors. The SCP contains programmable, service-independent capabilities (or service logic) that are under the control of service providers. As a separate offering, the SCP can contain service-specific data that can be customized by either service providers or their customers (at the service provider's discretion). In addition to its programmable functionality, the SCP provides SS7 interface to switching systems.

A third element in the AIN architecture is the intelligent peripheral. Intelligent peripherals provide resources such as voice announcements, voice recognition, and DTMF digit collection. Intelligent peripherals support flexible interaction between the end user and the network.

The two main benefits of AIN are its capabilities to improve existing services and to develop new services as sources of revenue. To meet these goals, service providers must introduce new services rapidly, provide service customization, establish vendor independence, and create open interfaces. AIN technology uses an embedded base of stored program-controlled switching systems and the SS7 network. At the same time, AIN technology enables the separation of service-specific functions and data from other network resources. This feature reduces the dependency on switching system vendors for software developments and delivery schedules. In theory, service providers have more freedom to create and customize services.[11]

Features

Custom Local Area Signaling Service (CLASS) features are basic services available in each *local access and transport area* (LATA). The features and

[10]Search Networking. "Advanced Intelligent Networks," p.1, http://searchnetworking.techtarget.com. (This URL is no longer valid.)

[11]Telcordia Technologies. "Intelligent Networks (IN)." A white paper hosted by the International Engineering Consortium at www.iec.org.

the services they enable are a function of Class 5 switches and SS7 networks. The Class 4 switch offers no features of its own. It transmits the features of Class 5. With almost three decades of development, the Class 4 switch has a well-established history of interoperability with the features offered by the Class 5 and SS7 networks. Features often enable service provider systems to generate high margins that, of course, equate to stronger revenue streams.

Examples of features offered through the DMS-250 system can be grouped into two major categories: basic and enhanced services. The basic services include 1+, 800/900 service, travel cards, account codes, PIN numbers, operator access, speed dialing, hotline service, *automatic number identification* (ANI) screening, *virtual private networks* (VPNs), calling cards, and call detail recording. Enhanced services include information database services (NXX number services, authorization codes, calling card authorization, and debit/prepaid card services), and routing and screening (includes Carrier Identification Code [CIC] routing, time of day screening, ANI screening, and class of service screening). Enhanced features also include enterprise networks, data and video services (dedicated access lines, ISDN PRI services, dialable wideband services, and switched 56 Kpbs), and multiple dialing plans (full 10-digit routing, 7-digit VPN routing, 15-digit international dialing, speed dialing, and hotline dialing). Most of these features have been standards on the DMS-250 and other Class 4 switches for many years.

This long list of features is evidence of the importance of features in the legacy market in which they were developed. Service providers are reluctant to give up these features and the higher margins they generate. In the converging market, features are equally important to reliability because service providers don't want to offer fewer features to their customers and they will want to continue to offer high-margin features.

Performance Metrics for Class 4 and 5 Switches

To date, the basis for choosing a Class 4 or 5 switch architecture over that of softswitch has been reliability, scalability, good QoS, and well-known features and applications. The question now becomes, what happens when competing technologies meet or exceed the standards set around the legacy of the Class 4 or 5 switch?

Reliability Not all businesses need to provide or rely on round-the-clock availability. For those that do, Table 2-4 from a 1998 Gartner Group study[12] illustrates how costly downtime can be within and across industries.

The telecommunications industry is a little different. If a Class 4 switch with 100,000 DS0s charging $0.05 per minute were to be down 1 hour, the service provider would lose $300,000 in revenue. Downtime is lost revenue. Five 9s of reliability is the standard for the legacy market. Service providers know their customers expect the same levels of reliability in any new market as they did in the legacy market. The focus of debate on this issue in the industry revolves around the question of how many nines of reliability a product can deliver. The PSTN or legacy voice network claims five 9s. What is meant by five 9s?

One to Five 9s Availability is often expressed numerically as a percentage of uninterrupted productive time containing from one to five 9s. For instance, 99 percent availability, or two nines, equates to a certain amount of availability and downtime, as does 99.9 percent (three nines), and so on. Table 2-5 offers calculations for each of the five 9s based on 24-hour, year-round operation.

How Does a Switch Achieve Five 9s? Five 9s are not the result of divine guidance given to Bellcore in the 1970s, but rather a process of engineering resulting in a high level of reliability. Reliability is enhanced when each component is replicated in a system. This is called redundancy. If one unit

Table 2-4
Costs of down time

Industry	Application	Avg. cost per hour of downtime
Financial	Brokerage operations	$6,500,000
Financial	Credit card sales	$2,600,000
Media	Pay per view	$1,150,000
Retail	Home shopping (TV)	$113,000
Retail	Catalog sales	$90,000

[12]Nelson, Gene. "Architecting and Deploying High Availability Solutions: Business Drivers and Key Considerations." Compaq white paper. October 1998. Available online at ftp://ftp.compaq.com/pub/supportinformation/papers/ecg0641198.pdf.

Table 2-5

Availability and downtime: How the five 9s are calculated

Availability	Downtime
90% (one 9)	36.5 days per year
99% (two 9s)	3.65 days per year
99.9% (three 9s)	8.76 hours per year
99.99% (four 9s)	52.55 minutes per year
99.999% (five 9s)	5.25 minutes per year

fails, its replicated unit takes over. Redundancy is usually expressed in terms of a ratio of 1:1 where one replicated unit exists for every primary unit or N11 redundancy where there are N(N>1) replicated units per primary unit. Another mechanism to enhance reliability is to ensure no *single point of failure* (SPOF) exists on the system. That is, every mechanism has a backup in the event of the failure of one unit. Hot standby refers to having a replicated unit take over the functions of its primary unit for either planned or unplanned outages. Continuously available (reliable) systems rely exclusively on active replications to achieve transparency in masking both planned and unplanned outages.[13]

Network Equipment Building Standards (NEBS) In addition to the five 9s, the other buzzword for reliability in the Class 4 market is *Network Equipment Building Standards* (NEBS). NEBS address the physical reliability of a switch. NEBS parameters are contained in Telcordia specification SR 3580, which contains the requirements for performance, quality, safety, and environmental metrics applicable to network equipment installed in a carrier's central office. Most North American carriers require equipment in their central offices or switching locations to be NEBS compliant. Tests include electrical safety, immunity from electromagnetic emissions, lightning and power faulting, and bonding and grounding evaluations. Between five 9s and NEBs, the Class 4 and 5 vendors have developed, through decades of experience, a reliable product that delivers superb uptime, and rapid recovery capabilities. As a result, service providers are often reluctant to experiment with new technologies.

[13]"The Five 9's Pilgrimage: Toward the Next Generation of High Availability Systems." A white paper from Clustra at www.clustra.com.

Scalability For financial reasons, a Class 4 or 5 switch must offer flexibility in scalability. Ideally, a service provider starts out with a chassis and a minimum capability in terms of DS0s (one DS0 is one phone line) and then adds more capacity as demand increases. This is considered a scalable solution and is preferable to a solution that requires either another chassis or a whole new system (known as a forklift upgrade) when demand grows beyond the initial installation. Nortel's DMS-250 Class 4 switch, for example, scales up to approximately 100,000 DS0s. The other metric for scalability is the call-processing power of the switch. The DMS-250, for example, can process 800,000 BHCAs.[14]

Another financial reason for scalability is to achieve a low cost per DS0 in purchases and operations. By buying a large switch with many DS0s, a service provider can negotiate a lower price per DS0 and improve the odds of being profitable in that market. For these financial reasons, scalability is important to service providers in both the legacy and converging markets. In terms of BHCAs, a Class 4, such as DMS-250, offers call-processing power at 800,000 BHCAs. The first softswitches offered no more than 250,000 BHCAs.

Quality of Service (QoS) The voice quality of the PSTN is the standard for telephone service. The Class 4 switch was the first deployment of TDM utilizing a 64 Kbps circuit (G.711 codec), which remains the standard to this day. Service on the PSTN is known for the absence of echo, crosstalk, latency, dropped or blocked calls, noise, or any other degradations of voice quality. Mainstream service providers in North America are reluctant to deploy equipment that offers voice quality at a lesser standard than the Class 4 or 5 switch.

One advantage service providers have had in delivering good QoS is that their legacy networks were designed specifically to deliver excellent voice quality and dependability in call setup and teardown. Historically, service providers have owned and operated their own proprietary networks over which they have had total control. Given end-to-end control, this has ensured good QoS for their subscribers. Good voice quality has long been a selling point for long-distance service providers.

Good QoS has been important in the legacy market. Service providers fear the loss of market share if they introduce a product in a converging market that does not deliver the same QoS as subscribers experienced in the legacy market. QoS is vitally important in legacy and converging markets.

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[14]Nortel Networks. "Product Service Information-DMS300/250 System Advantage." www.nortel.com, 2001.

Transport

The PSTN was built over the course of a century at a great expense. Developers have been obsessed over the years with getting the maximum number of conversations transported at the least cost in infrastructure possible. Imagine an early telephone circuit running from New York to Los Angeles. The copper wire, repeaters, and other mechanisms involved in transporting a conversation this distance were immense for its time. Hence, the early telephone engineers and scientists had to find ways to get the maximum number of conversations transported over this network. Through much research, different means were developed to wring the maximum efficiency from the copper wire infrastructure. Many of those discoveries translated into technologies that worked equally well when fiber optic cable came on the market. The primary form of transport in the PSTN has been TDM (described earlier in this chapter). In the 1990s, long-distance service providers (*interexchange carriers* or IXCs) and local service providers (*local exchange carriers* or LECs) have migrated those transport networks to *Asynchronous Transfer Mode* (ATM). ATM is the means for transport from switch to switch.

Asynchronous Transfer Mode (ATM)

IXCs use high-speed switching systems to interconnect transmission lines. The key high-speed switching system used in IXC networks is ATM. ATM is a fast packet-switching technology that transports information through the use of small, fixed-length packets of data (53-byte cells).

The ATM system uses high-speed transmission facilities (155 Mbps/OC-3 and above). OC-3 is the entry-level speed for commercial ATM. Higher speeds (such as OC-192) are used in backbone networks of IXCs and other specialized service providers. ATM service was developed to enable one communication technology (high-speed packet data) to provide for voice, data, and video service in a single offering. When an ATM circuit is established, a patch through multiple switches is set up and remains in place until the connection is completed.

The ATM switch rapidly transfers and routes packets to the predesignated destinations. To transfer packets to their destinations, each ATM switch maintains a database called a routing table. The routing table instructs the ATM switch as to which channel to transfer the incoming packet to and what priority should be given to the packet. The routing table is updated each time a connection is set up and disconnected. This enables the ATM switch to forward packets to the next ATM switch or destination point without spending much processing time.

The ATM switch also may prioritize or discard packets that it receives based on network availability (congestion). The ATM switch determines the prioritization and discard options by the type of channels and packets within the channels that are being switched by the ATM switch.

Three signal sources go through an ATM network to different destinations. The audio signal source (signal 1) is a 64 Kbps voice circuit. The data from the voice circuit is divided into short packets and sent to the ATM switch 1. ATM switch 1 looks in its routing table and determines the packet is destined for ATM switch 4, and ATM switch 4 adapts (slows down the transmission speed) and routes it to its destination voice circuit. The routing from ATM switch 1 to ATM switch 4 is accomplished by assigning the ATM packet a *virtual circuit identifier* (VCI) that an ATM switch can understand (the packet routing address). This VCI code remains for the duration of the communication.

The second signal source is a 384 Kbps Internet session. ATM switch 1 determines the destination of these packets is ATM switch 4 through ATM switch 3. The third signal source is a 1 Mbps digital video signal from a digital video camera. ATM switch 1 determines this signal is destined for ATM switch 4 for a digital television. In this case, the communication path is through ATM switches 1, 2, and 4.

Optical Transmission Systems

At the physical layer, carriers use microwave or fiber optic cable to transport ATM packets containing voice and data from switch to switch. IXC backbone carrier facilities primarily use microwave and fiber transmission lines. Microwave systems offer a medium capacity of up to several hundred Mbps communications with a range of 20 to 30 miles between towers. Fiber optic communication systems offer a data transmission capacity of over one million Mbps (one million in a million bits per second).

Microwave transmission systems transfer signal energy through an unobstructed medium (no blocking buildings or hills) between two or more points. In 1951, microwave radio transmission systems became the backbone of the telecommunications infrastructure. Microwave systems require a transducer to convert signal energy of one form into electromagnetic energy for transmission. The transducer must also focus the energy (using an antenna dish) so it may launch the energy in the desired direction. Some of the electromagnetic energy that is transmitted by microwave systems is absorbed by the water particles in the air.

Although the extensive deployment of fiber optic cable has removed some of the need for microwave radio systems, microwave radio is still used in places that are hard to reach or are not cost effectively served by fiber cable, such as in developing countries. Fiber optic transmission is the transfer of information (usually in digital form) through the use of light pulses. Fiber optic transmission can be performed through glass fiber or through air. Fiber optic transmission lines are capable of extending up to 1,200 kilometers without amplifiers. Each fiber optic strand can carry up to 10 Gbps of optical channels and a fiber can have many optical channels (called DWDM). Each fiber cable can have many strands of fiber.

Fiber cable is relatively light and low cost, and it can be easily installed in a variety of ways. It does not experience distortion from electrical interference and this enables it to be installed on high-voltage power lines or in other places that have high levels of electromagnetic interference. Optical transmission systems use strands of glass or plastic fiber to transfer optical energy between points. For most optical transmission systems, the transmitting end-node uses a *light amplification through stimulated emission of radiation* (laser) device to convert digital information into pulsed light signals (amplitude modulation). The light signals travel down the fiber strand by bouncing (reflecting) off the sides of the fiber (called the cladding) until they reach the end of the fiber. The end of the fiber is connected to a photodetector that converts these light pulses back into their electrical signal form.

Synchronous optical transmission systems use a specific frame structure and the data transmission through the transmission line is synchronized to a precise clock. This eliminates the signaling overhead requirement for framing or timing alignment messages. The basic frame size used in optical transmission systems is 125 usec frames.

Optical transmission systems are characterized by their carrier level (OCX) where the basic carrier level 1 is 51.84 Mbps. Lower-level OC structures are combined to produce higher-speed communication lines. Different structures of OC are used in the world. The North American optical transmission standard is called *Synchronous Optical Network* (SONET) and the European (world standard) is *Synchronous Digital Hierarchy* (SDH).

Signals are applied to and are extracted from optical transmission systems using an *optical add/drop multiplexer* (OADM). The OADM is a network element that provides access to all or some subset *synchronous transport signal* (STS) line signals contained within an *optical carrier level N* (OC-N). The process used to direct a data signal or packet to a payload of an optical signal is called mapping. The mapping table is contained in the OADM. A copy of the OADM mapping is kept at other locations in the event of equipment failure. This allows the OADM to be quickly reprogrammed.

SDH is an international digital transmission format used in optical (fiber) standardized networks that is similar (but not identical) to SONET. SDH uses standardized synchronous transmissions according to ITU standards G.707, G.708, and G.709. These standards define data transfer rates, defined optical interfaces, and signal structure formats.

Some of the key differences between SONET and SDH include differences in overhead (control) bits and minimum transfer rates. The first level available in the SONET system is OC1 and is 51.84 Mbps. The first level in the SDH system starts at STM-1 and has a data transmission rate of 155.52 Mbps. SONET also multiplexes *synchronous transport signal level 1* (STS-1) to form multiple levels of STS. The SDH system divides the channels into multiple DS0s (64 Kbps channels). This is why the overhead signaling structures are different.

Table 2-6 shows the optical standards for both SONET and SDH. This table shows that the first common optical level between SONET and SDH

Table 2-6

Optical transmission systems

OC level	Signal level	Sonet STS level	SDH STM level	DS0 (64 Kbps) channel
OC-1	51.84 Mbps	STS-1		672
OC-2	103.68 Mbps	STS-2		1,344
OC-3	155.52 Mbps	STS-3	STM-1	2,016
OC-4	207.36 Mbps	STS-4	STM-3	2,688
OC-9	466.56 Mbps	STS-9	STM-3	6,048
OC-12	622.08 Mbps	STS-12	STM-4	8,063
OC-18	933.12 Mbps	STS-18	STM-6	12,096
OC-24	1.24416 Gbps	STS-24	STM-8	16,128
OC-36	1.86624 Gbps	STS-36	STM-12	24,192
OC-48	2.48832 Gbps	STS-48	STM-16	32,256
OC-96	4.976 Gbps	STS-96	STM-32	64,512
OC-192	9.953 Gbps	STS-192	STM-64	129,024
OC-256	13.219 Gbps	STS-256		
OC-768	52.877 Gbps	STS-786		

is OC3 or STS-1. STS-x and STM-x are the standards that specify the electrical signal characteristics that are input to the respective optical encoding/multiplexing processes.

Conclusion

This chapter has described the major components of the PSTN. By categorizing the diverse components of the PSTN into three simple elements, access, switching, and transport, a framework is provided for understanding how comparable elements of a softswitch solution replace those of the PSTN, enabling bottlenecks to be bypassed in the PSTN when delivering voice services to subscribers. Many concepts deployed in the PSTN have been translated into softswitched networks, including signaling, voice codecs, and transport. When this technology can be duplicated by startup technology providers and implemented by competitive service providers, competition to the local loop becomes possible.