TELEPHONE NETWORKS

The telephone network, as we know it today, began in 1876 when Alexander Graham Bell was granted a patent for inventing the first telephone. Today's modern and advanced telephone system has evolved from a few directly connected subscribers to an extensive network of cable, radio, and satellite transmission systems interconnected through high-speed digital switching centers capable of connecting any two subscribers in the world in a matter of seconds. While transmission technology and advances in end instruments have played a large part in the expansion of the telephone network, it has really been the advances in computer-controlled switching elements that have propelled the telephone system into the fast, feature-rich, and reliable network that exists today.

TELEPHONE NETWORK EVOLUTION

At first, telephones were directly connected on a point-to-point basis, but it quickly became apparent that some form of multiple connection scheme was necessary to make the telephone practical. If a person needed a telephone for everyone that he or she wanted to talk to and a wire to that location, we would have a house full of phones and the skies would be darkened with telephone wires. The first form of switching was a manual process. Telephone customers were connected to a switchboard and would call into the operator and ask to be connected to one of the other customers on that switchboard. The earliest operational system of this type was placed in service in 1878 with 21 customers (1). This concept worked well for small communities where there were few telephone customers. There was no numbering plan and customers were known by their names. The method of making a phone call started by turning the ringer crank to signal the operator and then asking to be connected to another customer (i.e., Doc Jones). Each customer on the switchboard was connected to a patchcord (receive) and a socket (send). To make the connection, the operator would plug the calling person's patchcord into the socket of the person being called, and ring the line. The operator would monitor the call and pull the patchcords when done. Eventually switchboards were connected, allowing expansion of the calling area. Later, in 1879, an alphanumeric system of two letters and five numbers was used to identify each customer. While this was a highly personalized system, and sometimes the center of town gossip, it was completely manual, very slow, and cumbersome. Figure 1 shows an example of an early manual switchboard. Some of the early switchboards were operated by boys, but eventually phone companies changed to female operators (2).

The limited capabilities of an operator switchboard network forced the development and introduction of a more automatic capability for establishing phone calls. In 1892, Almon B. Strowger introduced the first commercial form of electronic switching called "step-by-step," which formed the basis of the automatic telephone system of the 1900s (3). This mechanical system, known as the Strowger switch, was based on the use of a dial-type telephone connected to a series of stepping relays that allowed the customer to dial a number representing that of the person being called. While this system became the foundation of switching for many years, improvements were made with the introduction of other forms of mechanical switching systems such as crossbar, panel, and XY, all designed to improve on the speed and efficiency of network connections.

Figure 1. Example of an early manual switchboard. (Copyright 1997. Reprinted by permission of Classic P10 Partners.)

The invention of the transistor made the electronic switches the next generation of modern switching. During the late 1950s, many experimental systems were developed, but it was not until 1965 that the first stored program control switch was put into service. This system was the American Telephone and Telegraph (AT&T) electronic switching system (ESS), which significantly improved the speed of call handling and became the basic building block for all switching systems through the 1970s and 1980s (4). As the computer came of age, advances in processor-controlled switching systems set forth a new era of stored program controlled switching. This new form of switching introduced the high-speed intelligent network of today, which offers advanced features such as calling party identification, voice activated calling, call waiting, and call forwarding.

NUMBERING SYSTEM

Once the concept of an automatic telephone network began to take shape, some form of structured hierarchy and numbering needed to be developed. Customers now became subscribers and were assigned unique four-digit numbers representing their line number on the automatic switch providing their service. The early numbering schemes used a two-alpha, five-digit numbering notation (the alpha representation was converted into actual numbers when being handled by an automatic switching system). During this era, the two-letter alphas were assigned to represent the name of the place where the switching system was located. When the limited alphanumeric numbering system became too cumbersome and unable to support the vast amount of customers wanting telephone service, the all number calling (ANC) plan was introduced in 1958. Each of the switching offices was assigned a unique address code of three digits to represent its office code, and several switching offices were grouped into areas (usually by state) and provided with a unique area code of three digits. This numbering plan became the North American Numbering Plan (NANP) in use today in the United States, Canada, Puerto Rico, Guam, and most Caribbean Islands. Under this plan, every telephone is assigned a unique 10-digit address consisting of a three-digit area code, three-digit office code, and four-digit subscriber number. Special dialing arrangements within modern switches allow dialing of only seven digits within the same area code and local calling area. In addition, special access codes also allow for dialing international subscribers almost as easily as calling across town. With the proliferation of telephone numbers, driven by the introduction of pagers, fax machines, cellular phones, and computer connections, the NANP has undergone major revisions in recent years to add more area codes and reduce the geographical size of these area codes in order to provide more customer numbers. Now many states and some larger cities are using multiple area codes (5).

TELEPHONE NETWORK HIERARCHY

As the number of subscribers grew in the telephone network, along with their need to call more and more people across the nation, it became apparent that it would be impractical and costly to connect all switching offices so that every person could call any telephone subscriber in the country (and eventually the world). This evolving telephone industry became a national system under the control of the Bell telephone operating system. In the early 1900s there were only 10 million subscribers in the Bell System, but by the 1980s there were close to 180 million subscribers. The Bell System introduced a hierarchical network architecture to reduce the amount of network connections and improve the speed of completing calls. This network hierarchy is shown in Fig. 2 and consists of class 5 end offices (EOs) or central offices (COs) at the lowest level. These EOs are the switching systems that terminate all the subscriber lines in the area. In many cases, in large metropolitan areas using several EOs, the class 5 offices were connected to a local tandem office to provide interconnectivity to all class 5 EOs in the area. The tandem office's sole function was to connect calls from one EO to another. It did not terminate subscribers. The next level in the hierarchy is the class 4 toll office. Each class 5 EO is connected to a class 4 toll office for access to the long-distance toll network interconnecting the country. The toll network has three levels of connectivity through class 3 primary centers, class 2 sectional centers, and class 1 regional centers. The whole design of the system was to allow calls to complete by using the fewest number of switching centers. The routing logic was designed to complete the call using the least number of connections, but alternate routing used the longer path, if needed, to ensure completion of the call. Figure 2 shows the different routing paths available for call completion. Table 1 lists the number of each class offices in operation in the United States in 1982. Tandem offices are not listed in the table because they were not part of the toll network (6).

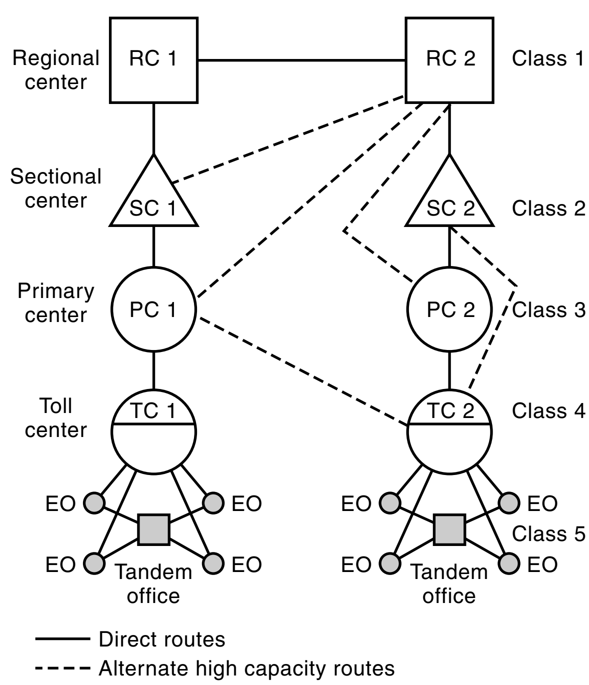

Figure 2. Bell Telephone toll hierarchy.

Table 1. Public Network Hierarchy of the Bell System (1982)

Switch Class	Functional Designation	No. in Bell System	No. in Independent	Total
1	Regional center	10	0	10
2	Sectional center	52	15	67
3	Primary center	148	20	168
4	Toll center	508	425	933
5	End office	9803	9000	18,803

POSTDIVESTITURE NETWORK

The Bell hierarchical network was modified in January 1984, when divestiture of AT&T was decreed by the courts. No longer was AT&T the sole provider of telephone service from phone to phone, but competition was allowed in virtually every aspect of the business. The divestiture broke AT&T into seven smaller Bell operation companies (BOCs) as well as many other independent telephone companies. AT&T itself provided the long distance service, but now came into competition with other companies such as MCI and Sprint. AT&T replaced the three level toll network with a flat (i.e., single) level network consisting of 142 tandem switches. The new network is shown in Fig. 3.

At the lowest level are local access and transport areas (LATA) established mainly along existing networks being operated by incumbent service providers known as local exchange carriers (LEC). The number of LATAs at the time of divestiture was 164. Toll calls between exchanges within the

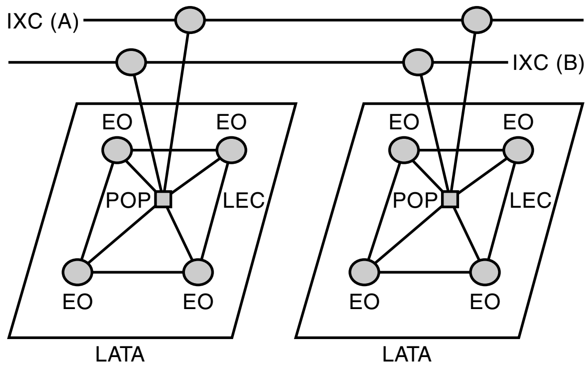

Figure 3. Postdivestiture network.

LATA are the responsibility of the LEC. For inter-LATA traffic, the LECs connect to interexchange carriers (IXC) whose sole responsibility is to connect the long-distance toll calls. Today, customers are allowed to pick which IXC they use for their long-distance service, and that part of the market has become a multi-billion-dollar industry. LECs interface with the IXCs at a single point of entry into and out of the LATA called a point of presence (POP). This POP is nothing more than a designated point, like a tandem switch or class 4 toll office, within the LATA (7).

ANALOG AND DIGITAL SIGNALS

The basic premise that makes the telephone work is the conversion of the spoken word into an electrical signal. The vibrations created by our speech are converted into electrical variations of voltage for transmission over wires. This signal is an analog representation of the voice pattern and is continually varying in amplitude and frequency. This analog signal is based on the frequency of our voice in the range of 0–3 kHz. Figure 4(a) shows the basic method of speech transmission for the analog signal. Transmission over long distances causes the signal to fade, and therefore it needs to be amplified at various points along the path. This amplification process unfortunately introduces noise into the original signal so that when it arrives at its intended destination it is of lower quality than the original. Noise is introduced into this signal from unwanted sources such as power lines, motors, switching equipment, and electrical storms. The amplifier has no way of distinguishing between the real signal and the noise. Special care is taken to ensure that the amplifiers are built with noise-eliminating filters and circuitry, but the noise can never be totally eliminated. The need to improve on service led telephone engineers to develop methods of digitally encoding the human voice into a series of pulses that would represent the analog signal. One form of encoding, known as pulse code modulation (PCM), is the technique used in modern switching systems to provide the high-speed and high-quality networks in use today. The uniqueness of digital transmission is that it reduces the introduction of noise. Signals are not amplified (along with noise), but rather they are regenerated (repeated) as pulses of "1s" and "0s." Figure 4(b) shows how the noise is eliminated in transmission with the use of repeaters. Digital signals are also processed more efficiently in modern digital switching and transmission systems using computer-based principles. The higher speed at which the digital signals are processed allows them to be placed in different time slots

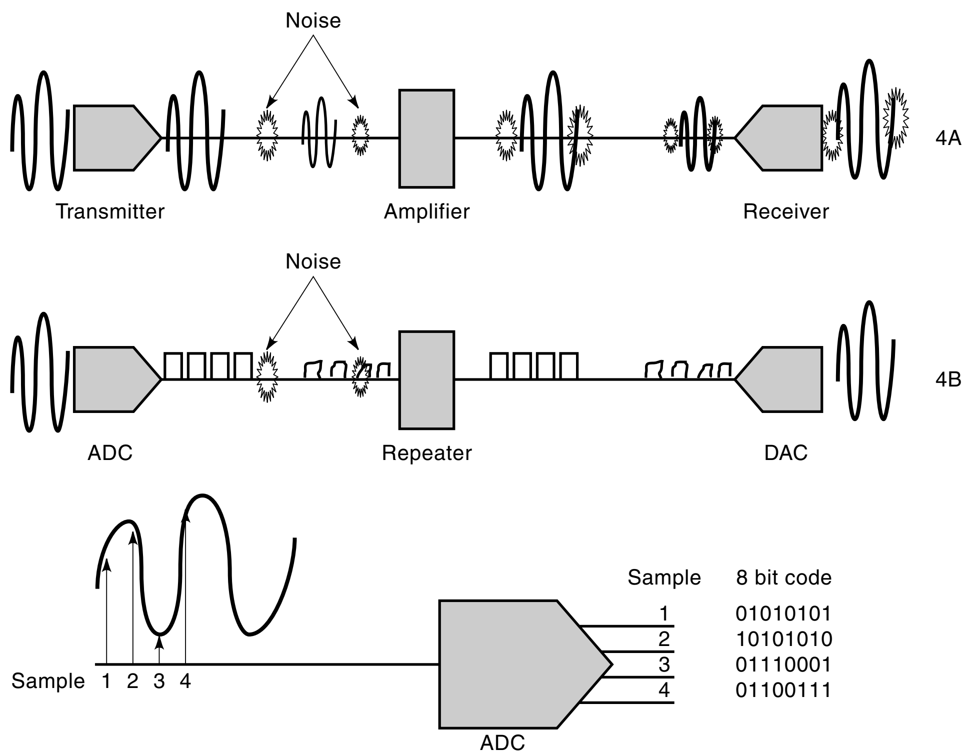

Figure 4. Comparison of analog and digital signals and the affect of noise.

(multiplexing) in transmission systems, thus allowing many conversations to take place over the same pair of wires.

For the conversion of an analog signal to digital pulses, the analog signal is processed by an analog to digital converter (ADC) also known as a coder/decoder or codec. Figure 4(c) shows the basic ADC process. The ADC samples the analog signal and converts it into a series of digital pulses. Experimental design determined that sampling an analog signal 8000 times per second was sufficient enough to represent the signal adequately. Therefore, at an 8 kHz sampling speed, the ADC samples the analog signal every 125 μs and converts it into a number representing the amplitude and slope (direction) of the signal. This number is then converted into a binary eight-digit code for use in switching and transmission. At the receive end, the binary code is converted back into a voltage level and slope in the digital to analog converter (DAC), which then provides the reproduced analog signal to the receiver.

As with analog, a digital signal also becomes degraded due to signal loss and noise in the transmission media and switching equipment. However, the digital signal is regenerated rather than amplified through use of regenerative repeaters. These repeaters determine if the incoming signal is a 1 or 0 and then generate a new pulse of 1 or 0. Any noise on the circuit is lost since it is a new signal. The signal received at the destination is still the same 8 bit code sent at the originating end and is used to reproduce the original signal using the DAC process (8).

BASIC SWITCHING TYPES

The first telephone switches were manual switchboards, but they used the same basic process for call completion used by all automatic systems today. This basic process has the following steps:

1. *Initiation.* The subscriber notifies the switch that a call needs to be placed (off hook).
2. *Signaling.* The subscriber tells the switch where the call is to go (dialing).
3. *Switching.* The switch determines where to connect call (routing).
4. *Connection.* The switch sets a path to the destination and connects (ringing).
5. *Disconnect.* The call is dropped when completed (on hook)

Starting with the introduction of the first automatic switch in 1892 by Almon B. Strowger, all switching systems for the next 50 years or so were electromechanical. These systems were progressive control switches and were either under direct control, as in the step-by-step Strowger type, or under common control, as with the crossbar. These switches could only do basic call completion and had no other features to offer the subscribers. It was not until the introduction of stored program controlled (SPC) switching with the AT&T No.1 electronic switching system (ESS) in 1965 that advanced features were possible (9).

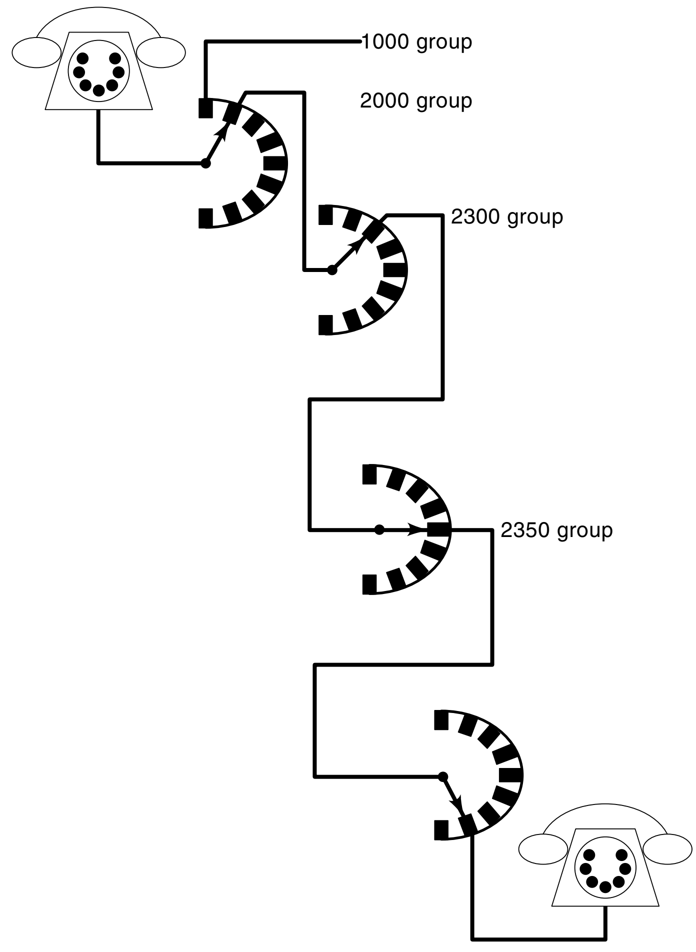

Figure 5. Step-by-step switch operation example that shows the completion of a call to a subscriber with the number 2359.

Step by Step

The Strowger step-by-step switch is a direct control form of switching. The principle behind the step-by-step switch is the use of dial pulses generated by the dialing telephone to control the "stepping" of a switch relay. Strowger's telephone was designed to generate a series of pulses by opening and closing a set of contacts on the phone a prescribed number of times for each digit dialed. The opening and closing of the contacts create a voltage and no-voltage condition, causing a current flow on the line to step the relay at the other end. Thus one pulse is a 1, two for the number 2, and so forth. Zero is actually 10 pulses. Figure 5 shows the basic concept of the Strowger step-by-step switch.

The incoming line is connected to the first "stepper" and, based on the number of pulses dialed, the stepper steps to one of ten positions. Thus a dialed number of 2 would step to the second contact, setting up a path to all subscribers in the 2 "thousands" group of numbers. This contact is connected to another "stepper" consisting of another set of 10 contacts and will move to the position determined by the next dialed number. The call progresses as the dialed digits step each successive set of "steppers." The example in Fig. 5 shows how the

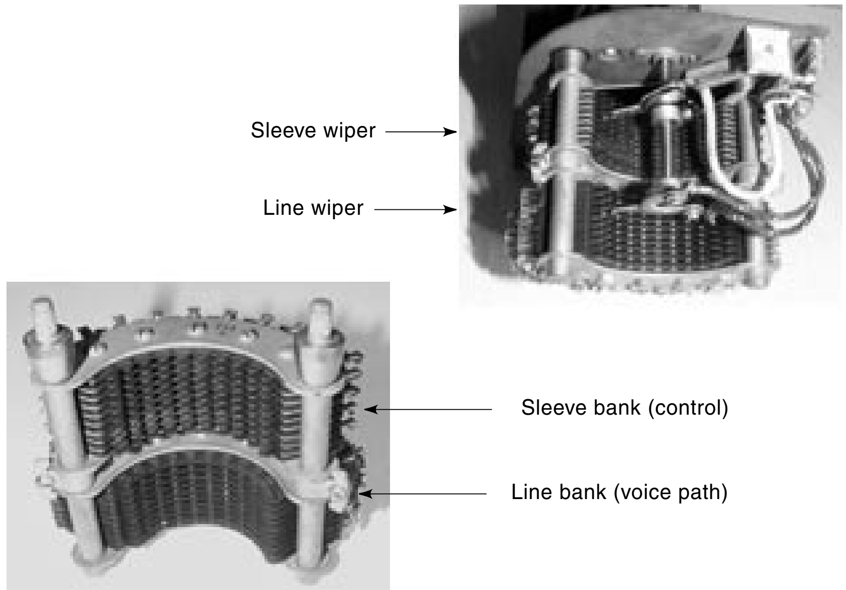

Figure 6. A Strowger-type step-by-step switch with 10 levels of 10 contacts.

dialed number 2359 will complete to a specific subscriber line. The “0” level as the first number dialed is reserved for the operator. When dialing to another exchange, the first three digits of a seven-digit number determine which trunk to connect. Once connected to the trunk, the last four digits are used to complete the call to the specific subscriber. As advances were made in the step-by-step process, contacts were stacked in a vertical bank and ganged together so that more than one call could be in progress at one time. As shown in Fig. 6, the most popular Strowger switch was stacked in a 10×10 matrix. Thus each level of contacts are connected to 10 more banks. Consider the set of “steppers” in Fig. 5 to be repeated or stacked in 10 vertical planes. As the first call is initiated, the stepper switch automatically moves up to the first bank of contacts. The second call would use the second level, the third would use the third level, etc. Once the level is established, the dialed pulses step to the appropriate contact. The process is repeated until the call is connected (10).

Step-by-step switching is progressive, under direct control of the dial pulses from the calling phone. As each set of pulses is dialed and the switching relay is stepped to appropriate contacts, the next stepping relay must be ready to accept the next set of pulses. If anything happens to the call progress due to equipment failure or blocked routes, the call is not completed. There is no backing up to try another route. The step-by-step switch is also a mechanical nightmare to maintain, and because of the current-driven relays, it becomes a major source of unwanted noise. Figure 7 shows a bank of step-by-step switches along with the relays and wiring as they were installed in a telephone control office. A typical control office would have thousands of these “steppers” connected together in multiple rows, stacked from floor to ceiling.

Common Control Crossbar

The deficiencies of the step-by-step Strowger switch led to development of the crossbar switch using a process known as common control. With common control, all the digits are dialed into a memory logic, called a register or marker, where they are maintained for use until the call is connected. The basic building blocks of a common control switch are as follows (11):

1. *Line Equipment.* Determines request for service and accepts digits into memory.
2. *Switching Network.* Used to connect calling party to called party (i.e., crossbar).
3. *Common Control.* Performs control of switching process.
4. *Trunk Equipment.* Provides connectivity to adjacent switches and toll network.

Figure 7. A bank of step-by-step switches used in a telephone central office.

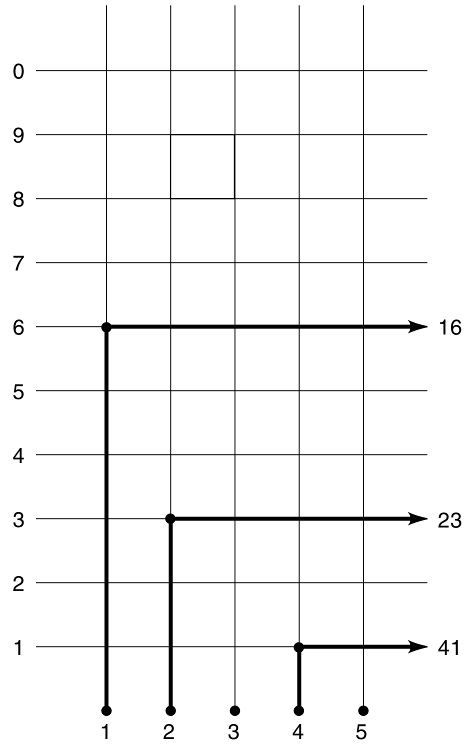

Figure 8. An example of a crossbar switch matrix showing the connection of 3 different call paths.

The crossbar switch differs from the step-by-step in implementation of the method of connecting two points together. Figure 8 shows a simplified example of how a crossbar matrix works. The example shows a 5 by 10 matrix. Each crosspoint in the matrix is activated by the intersection of two bars, thus the name crossbar. A path through the matrix is established by activating the intersection point of each bar through a set of magnetic contacts. Thus three different connections can be made at the same time for numbers 16, 23, 41. Several crossbar matrixes are staged together to provide for several simultaneous yet independent paths through the network. The common control network uses the stored digits in the register to determine the final address as it establishes a path through the matrix to the endpoint. If a path is blocked, the common control will try alternate paths, thus improving the probability of call completion. Crossbar switch matrix elements were used for many years until being replaced in the 1970s and 1980s by electronic switching systems using stored program control.

Electronic Switching Systems

The advent of modern-day telephone networks, with feature-rich service, began with the introduction of electronic switching systems (ESSs) using stored program controlled (SPC) logic to control switch call handling. Step-by-step and crossbar switches were fixed in their architectures and the services that they could provide. Once the architecture was "wired" in, it could not be changed. Such switches basically only connected one subscriber to another and did not offer any special services, like call forwarding, call transfer, and call waiting. As the electronic age emerged in the 1970s and 1980s, control of telephone switches migrated from fixed, hardwired logic to processor controlled systems. The early SPC switches used writeable memory, ferrite core cards to store the programs for controlling of switch functions. Later, computers and software took over that role.The basic operation of a SPC switch allocates call completion and management to one of five program functions (12).

1. *Input Programs.* Scan incoming and outgoing lines and trunks to recognize requests for service and maintain status information.
2. *Operational Programs.* Analyze received information (i.e., digits dialed) to determine what action to take.
3. *Subroutines.* Perform digit analysis, determine routing information, and establish connectivity through switch.
4. *Output Programs.* Make and release calls to lines and trunks.
5. *Executive Control.* Oversee all other program actions and direct the process as the call is handed off between programs.

Switch Matrix. The modern-day ESS digital-type switches have replaced the mechanical-type stepping and crossbar switching arrangements with faster electronic devices based on transistor-type electronics. Switching is done digitally and is fast and reliable. The functional area of the switch that performs this switching is called a digital matrix. There are two common methods of connecting calls through the matrix: space division and time division. Several combinations of space and time division switching are used by the switch manufacturers in their architecture designs.

The analog automatic switches, such as the Strowger step-by-step and the crossbar, used space division switching as the method of connecting one point to the other using shared facilities. These switches were designed with the realization that not everyone would be using the telephone at the same time. Therefore, the equipment to cross connect the two subscribers is used for the duration of the call and then released, for someone else to use, when the call is completed. For instance, when designing a space division switch to support 100 subscribers, the equipment would be engineered to support 10 calls at once and would be wired to support 10 separate calling paths. Thus the allocation of calls is done in the physical realm of established paths (space).

Space division switching is also used in digital switching networks when connecting one digital path to another. The difference between analog and digital is that pulses are being moved instead of analog signals. In all ESS digital-type switches, the analog signal produced by the calling party at the instrument is converted to a digital signal at the switch. This signal is handled digitally throughout the switching process and transmission, until it is connected to the called party at the final destination. The call is then converted back to analog. A digital space division matrix switches the digital signal from one digital time slot to another, as shown in Fig.

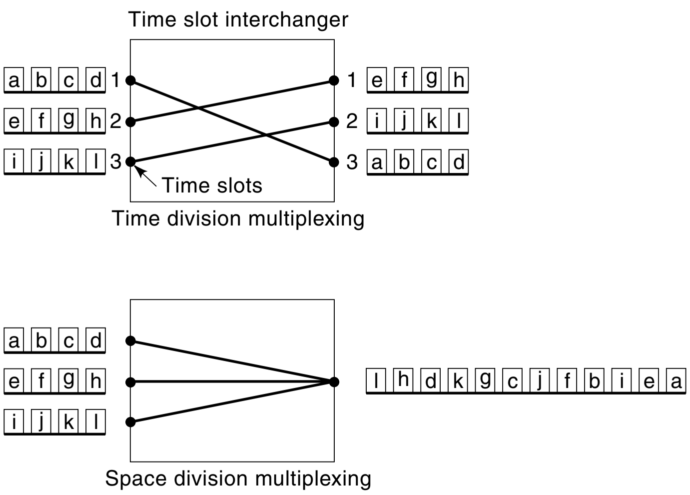

Figure 9. Comparison of digital space division and time division switching.

9(a). With time division switching, the digital signal is switched in time slot order in a digital pulse chain, as shown in Fig. 9b. With combinations of space division and time division switching, many simultaneous calls are able to be processed though an ESS-type switch. Figure 10 is a simplified example of the use of a time-space-time (TST) matrix to connect three calls.

As an example of a modern-day switch, the AT&T 5ESS®-2000 switch (Fig. 11), is an example of a system that is based on a distributed SPC architecture. It is a digital time division switch of modular design and can be configured to serve up to 150,000 subscribers lines and 35,000 trunks simultaneously. Its call processing capabilities are not only fast, but it offers the advanced features of today's networks, such as caller identification, call waiting, call forwarding, and three-way calling. The 5ESS®-2000 switch is made up of three major functional areas (modules), each performing complicated tasks through distributed processors, all under control of the central control (CC) computer. Table 2 lists the three modules and the functions that they perform. During call processing within the switching module (SM/SM-2000), the line unit (LU) detects an incoming call and in coordination with the module control time slot interchanger (MCTSI) sets up the call, provides dial tone, and accepts the dialed digits. Once the MCTSI determines where the call is going, it sends a message to the communications module (CM) and an intermediary routing MCTSI to perform digit translation to determine final call routing. The CM sends a message to the terminating switching module (SM) containing the dialed digits and other information needed to handle the final call connection. The CC and the terminating SM coordinate the status of the called subscriber (busy or not) and then establish the time slot to be used to connect the call through the switch matrix. Once this is done, the terminating MCTSI sets up a path through the terminating LU and rings the called subscriber. When the call is completed by either party hanging up, the CM is notified and coordinates the disconnect of all equipment and sets the time slots to idle, ready for use in the next call. During the call process, a record is kept of the call information and recorded through the administrative module on to magnetic tape for billing information (13).

The AT&T 5ESS is just one example of ESS SPC-type switching technology. Table 3 lists the most common digital telephone switching systems in use in North America over the last 20 years.

SIGNALING

Subscriber Signaling

For a telephone switch to operate and be able to perform the functions that it is designed to do, there needs to be some form of signaling scheme set up between the telephone and

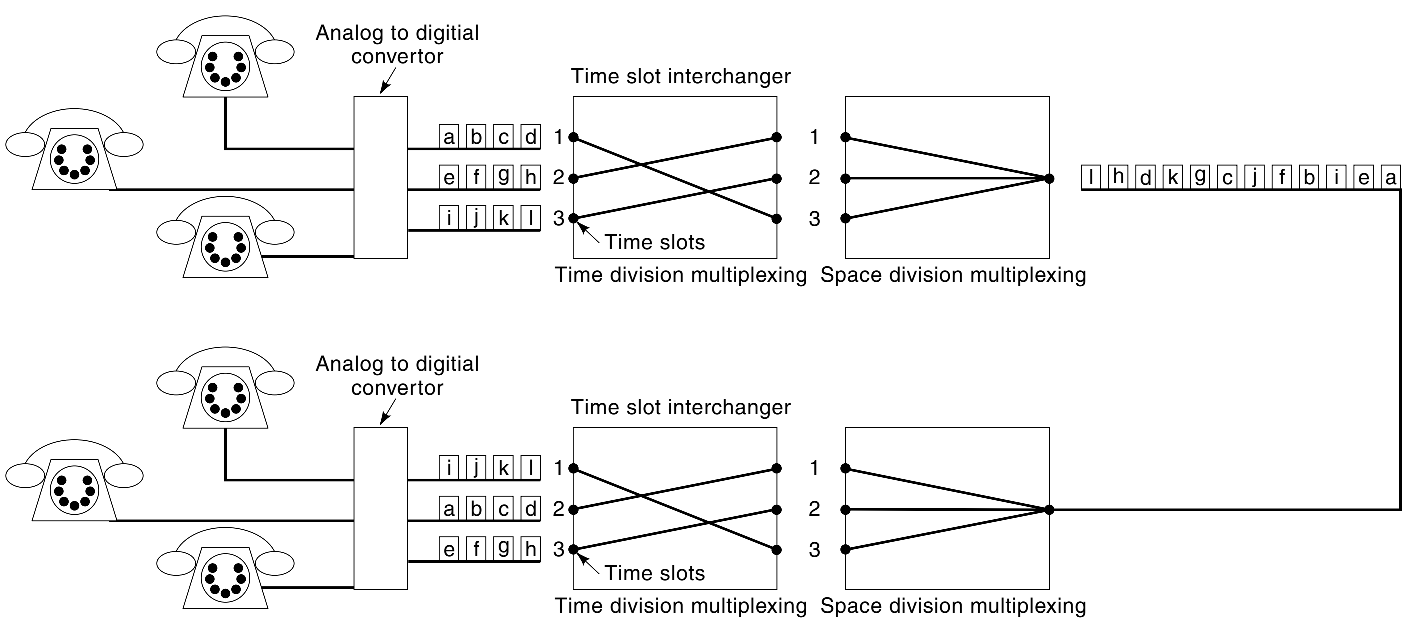

Figure 10. An example of a time-space-time switch.

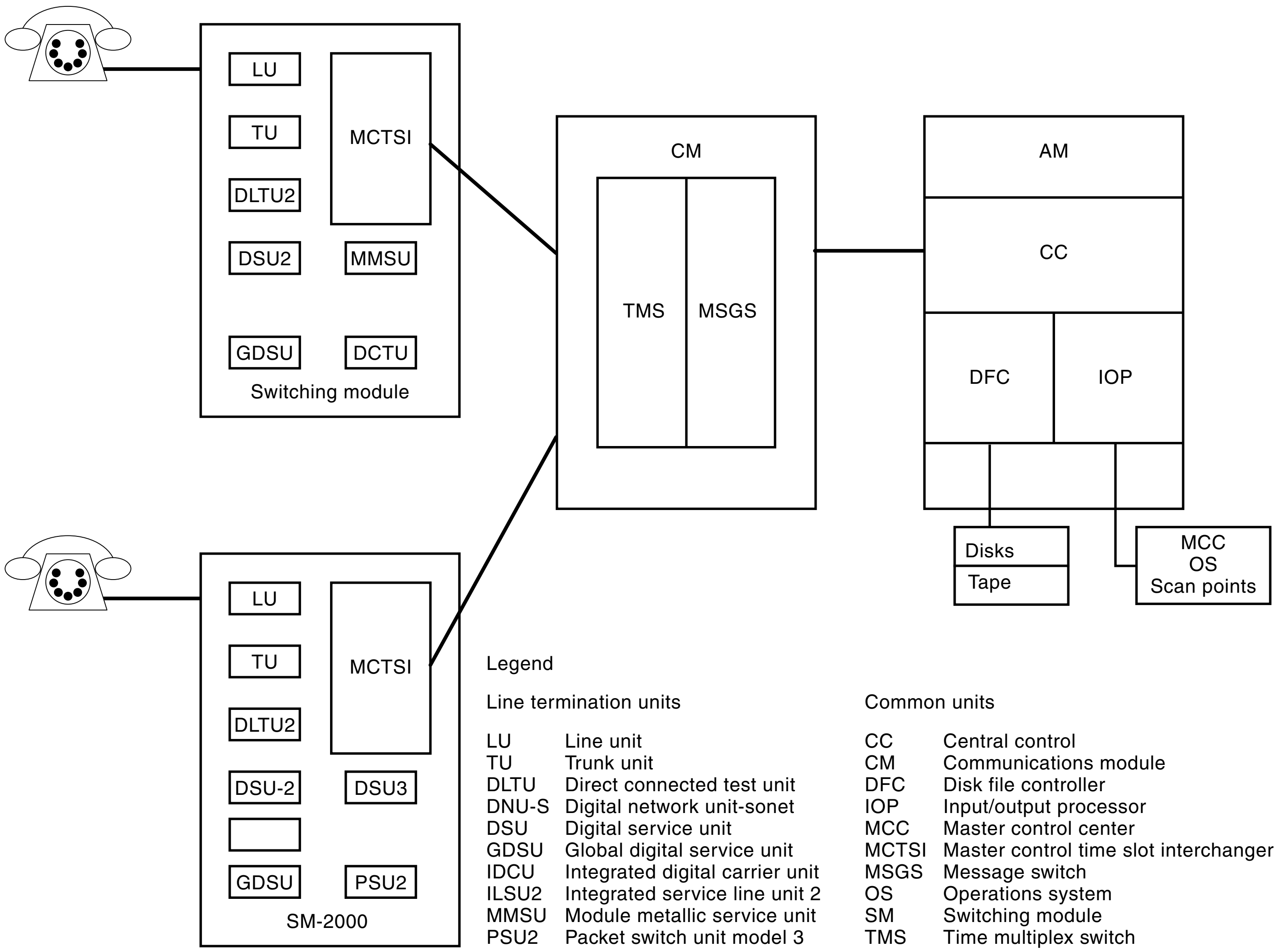

Figure 11. Functional diagram of a 5ESS®-2000 switch.

the switch to communicate when a call needs to be placed and to whom. The first automatic switch designed by Strowger used dial pulses to communicate to the switch. This form of signaling was used until the 1970s, when tone dialing was introduced and became the standard form of dialing. Although the dial pulse method can still be used on telephone switches, it can only convey limited information and cannot be used after call completion to control automatic answering systems and services.

A phone has two basic states of operation: on hook (idle) or off hook (in use). In the on-hook condition, the pair of wires to the switch are in an open condition. When a call is to be made, or answered, the receiver is taken off hook, closing the loop through a pair of contacts. This is normally a -48 V loop that draws loop current through the wires from the battery (central) source at the switching center. This flow of loop current tells the line termination device (LU for the 5ESS®-2000) that a call is about to be established (off hook) or a call has been answered. When placing a call, the switch applies dial tone to the line indicating to the caller that the switch is ready to accept dialed numbers. For dial pulse, the dialing mechanism (rotary) pulses the contacts the proper number of times for each number. Each pulse opens the contacts for 50 ms and closes them for 50 ms at a rate of 10 pulses per second. The time between digits is nominally 700 ms and is based on the minimum time to dial the next number on a rotary dial.

Because of the slow speed of dial pulses and limited information set, tone dialing was introduced. The tone keypad produces a discrete set of frequencies for each number key pressed. This method is called dual tone multifrequency (DTMF) because each frequency is comprised of two tones (one high frequency and one low) determined by the switch closure at the crosspoint, as shown in Fig. 12. For number 1, the composite tone is made up of 1209 and 697 Hz, number 2 is 1336 and 697 Hz, and so forth. When a call is connected, these same frequencies are used to access such things as bank accounts, reservation services, and subscriber extensions. These tones can be passed over the same channel as the voice call is using since these composite tones are within the voice frequency range (14).

Interoffice Trunk Signaling

Signaling between switches offices (interoffice signaling) is done in a similar manner as with loop signaling between the

Table 2. 5EES®-2000 Functions

Module	Functions	Submodules
Administrative Module (AM)	Provides a common interface to all of the 5ESS-2000 Switch Outputs reports for all the 5ESS-2000 Switch Coordinates common maintenance activities	Common Control (CC) Disk File Controller (DFC) Input/Output Processor (IOP) Master Control Center (MCC)
Communications Module (CM)	Switches messages between processors of the SM/SM-2000(s), CM, and AM Provides timing synchronization Switches data (voice) between SM/SM-2000(s) Is central space switcher Handles global or central resources	Time Multiplex Switch (TMS) Message Switch (MSGS)
Switching Module/Switching module-2000 (SM/SM-2000)	Terminates analog, digital and ISDN lines and trunks Converts analog to digital and digital to analog Contains highly intelligent processors that perform over 95% of the call processing functions of the 5ESS-2000 Performs time division switching of time slots Performs service functions	Line Unit (LU) Trunk Unit (TU) Digital Line Trunk Unit (DLTU) Digital Service Unit (DSU) Global Digital Service Unit (GDSU) Module Control Time Slot Interchanger (MCTSI) Modular Metallic Service Unit (MMSU) Directly Connected Test Unit (DCTU) Integrated Digital Carrier Unit (IDCU) Integrated Services Line Unit (ISLU) Packet Switching Unit (PSU)

subscriber and the switch. A series of on and off pulses (reversals of battery polarity) are used to signal the other end that a call is to be made or disconnected, and then the signaling is sent over the transmission media using a set of frequencies similar to the DTMF frequencies. Table 4 lists the set of multifrequency (MF) signaling codes used to signal between offices. The KP signal indicates the start of pulse sending and the ST signal indicates the end. In the process of signaling between offices, the calling office sends a connect or seizure signal to the other end indicating that it wants to set up a call. A seizure signal consists of a constant on-hook connection. The other end returns a momentary pulse called a wink to indicate it is ready to receive signaling information. Once the circuit (trunk) is set up, the calling office sends the digits for the number being called in a sequence such as KP9295521ST. This form of signaling is referred to as in-band or channel associated signaling (CAS) since the signaling information is sent between switches over the same path used for the voice call.

In the 1980s, a new form of signaling was introduced to telephone networks to improve on the older style CAS. By using a separate out-of-band, or common channel signaling (CCS) link, high-speed messages are passed between switches to handle call management. The higher-speed message traffic allows switches to communicate at a higher level of intelligence and make call connections extremely fast, even on

Table 3. Digital Central Office Switching Systems of North America

Manufacture	Designation	Date of Introduction	Application	Line Size
AT&T	4 ESS	1976	Toll	107,000
AT&T	5 ESS	1983	Local	100,000
CIT-Alcatel	E 10-five	1982	Local	100,000
GTE	3 EAX	1978	Toll/tandem	60,000
GTE	5 EAX	1982	Local	145,000
LM Ericsson	AXE10	1978	Local/toll	200,000
NEC	NEAX-61	1979	Local/toll	80,000
ITT	System 1210	1978	Local/toll	26,000
NorTel	DMS-10	1977	Local	7,000
NorTel	DMS-100	1979	Local	100,000
NorTel	DMS-200	1978	Toll	60,000
Siemens	EWSD	1981	Local	200,000
Sromberg Carlson	DCO	1977	Local	32,000
Vidar	ITS4	1977	Toll/tandem	7,000
Vidar	ITS4/5	1978	Local/toll	12,768

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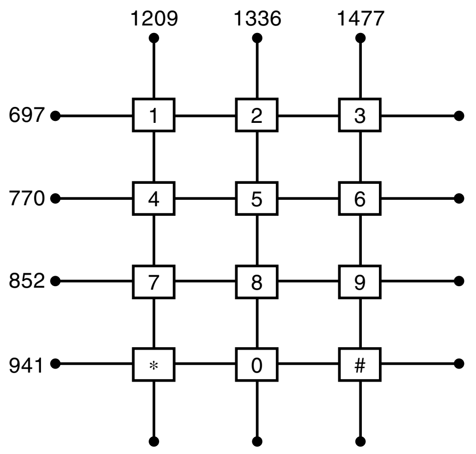

Figure 12. Dual tone multiple frequency touch pad.

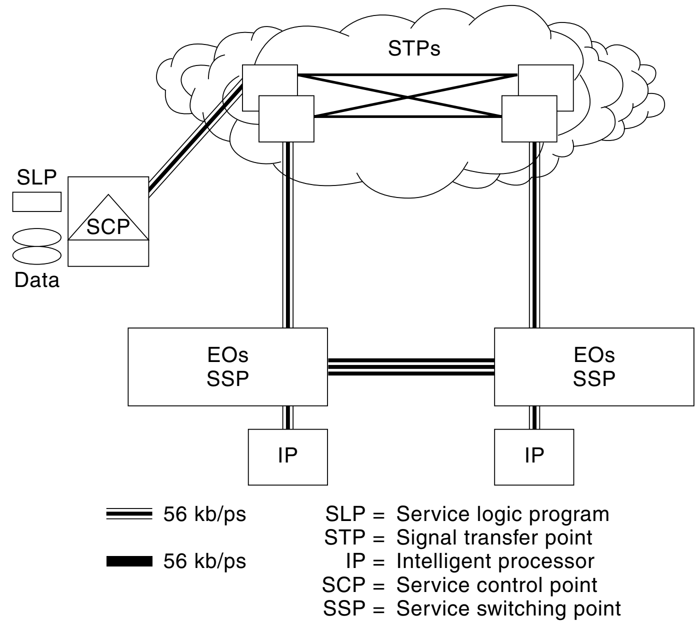

Figure 13. Simplified common channel signaling advanced intelligent network.

coast-to-coast or international calls. An in-band CAS call may take from 2 to 10 s to be established while a CCS call will be set up in under a second. Figure 13 shows the connectivity of the CCS network used in North America today. Each EO and other levels of switches are connected to a signaling transfer point (STP), which accepts a call setup message from an originating switch and relays that information the terminating switch. The messages coordinate the type of call, called number, calling number, status of the called party (busy), and which trunk circuit to use in the call establishment. In the case of 800 and 888 type numbers, the STP will look in a database (service control point, SCP) and determine what the real telephone number (area code and subscriber number) is for the call. The 800/888 numbers are really phantom-type numbers that do not change very often but can be assigned to any real number depending on the information in the database. When someone calls the 800/888 number, the CCS link gets the real telephone number from the SCP database. That way a company can have one 800/888 type service number but can assign the answer point wherever it wants merely by changing the database entry. One other key feature of CCS is that the STP can communicate with the terminating switch to determine if the number being called is busy. If so, it sends back a message to the originating switch and a circuit is never established between EOs. The originating switch provides the busy tone signal to the calling party. This saves expensive cross-country long-distance trunks from being tied up sending busy tones to the calling parties (that is not charged) instead of serving revenue-type calls (15). See SIGNALING.

Table 4. Multifrequency Tones

Number	Low Tone	+	High Tone	Number	Low Tone	+	High Tone
1	700		900	7	700		1500
2	700		1100	8	900		1500
3	900		1100	9	1100		1500
4	700		1300	0	1300		1500
5	900		1300	KP	1100		1700
6	1100		1300	ST	1500		1700

MULTIPLEXING

The simplest method of establishing a telephone voice call from one location to the other is to utilize a single path of wires between each end of the circuit. With increased long-distance traffic, more and more wires were required to carry traffic between switching centers across the country. To improve the efficiency of this operation, multiplexing schemes were introduced to facilitate transmitting simultaneous voice conversations over the same path. Frequency division multiplexing (FDM) allocates each voice call a different frequency slot on the transmission media. A pair of wires can actually carry frequency signals higher than the one 3 kHz voice call. The basic FDM scheme allocates voice channels to one of 12 frequency slots between 60 and 108 kHz at 4 kHz spacing. The voice signal is modulated with the carrier frequency (i.e., 64 kHz) for transmission over the path and then is demodulated through electronic filters at the receiving end. These 12 voice channels are a "group" of signals used in the Bell System as a building block for multiplexing signals in a FDM hierarchy of increasing size. Five groups are combined into a supergroup, 10 supergroups are combined into a mastergroup, and 6 mastergroups into a jumbogroup. Finally, 3 jumbogroups are combined into a jumbogroup mux, providing for the transmission of 10,800 individual voice calls over one single transmission media. As the number of channels increases and the frequency bandwidth reaches 60,000 kHz, different types of transmission media are used, from twisted wire to coaxial cable and microwave radio.

As with the FDM networks, digital networks are also multiplexed but are based on time division techniques instead of frequency. Time division multiplexing (TDM) hierarchy is

Table 5. Fiber Optic Carrier (OC) Rates

OC Rate	Megabytes per Second	OC Rate	Megabytes per Second
OC 1	51.840	OC 24	1244.160
OC 3	155.520	OC 48	2488.320
OC 12	622.080	OC 96	4976.640
OC 18	933.120	OC 192	9953.280

based on using the sampled digital signals derived in the pulse code modulation (PCM) technique used to convert analog voice signals to a series of digital samples. As was discussed earlier, an analog voice signal is sampled at 8 kHz rate (125 μs) and converted into an 8 bit code word. The multiplexing of voice channels in the time division spectrum provides a time slot every 125 μs for a new sample. By designing a transmission system that is fast enough to transmit pulses much faster than 125 μs, many channels can be sent over the transmission system virtually simultaneously. If a single voice call is used, the rate of transmission would need to be 8 bits every 125 μs or 8000 bits per second (b/s) (for two voice channels, 16 bits every 125 μs or 16 kb/s). The basic multiplexing scheme divides the time slots into 24 channels and operates at a rate of 1,544,000 bits per second or 1.544 megabits/s. At that rate, 193 bits of information can be sent over the transmission media every 125 μs as derived from the following formula:

$$
\begin{aligned}
(24 \text{ channels} \times 8 \text{ bits per sample}) &= 192 \text{ bits} \\
&\quad + 1 \text{ bit for alignment} \\
&= 193 \text{ bits}
\end{aligned}
$$

$$193 \text{ bits}/0.000125 = 1{,}544{,}000 \text{ bps}$$

In North America (and Japan) this digital transmission system is called the T carrier system. The basic rate of 1.544 megabits/s is called the T1 rate or DS1 (digital signaling rate 1). The system is then multiplexed up at higher rates at T2/DS2 at 6.312 megabits/s (4 T1s), T3/DS3 at 44.736 megabits/s (28 T1/DS1s), or T4/DS4 at 274.176 megabits/s (6 T3/DS3s). As can be seen by this hierarchy scheme, up to 4032 voice channels can be sent over the maximum T4/DS4 transmission media (16).

To gain more efficiency and higher capacity of telephone transmission highways, the introduction fiber optic transmission technology has launched circuit capacity into another realm. The capacity of wire and radio transmission systems is physically limited by the media itself. The spectrum of electrical frequencies is limited to wire or radio transmission systems. Fiber optics allows the use of the higher-speed light spectrum for transmission over an optical path at literally the speed of light. By modulating this light into discrete channels, much higher capacity can be realized. Therefore, optical transmission media can provide the multiplexing of voice signals into optical channel rates, as shown in Table 5.

NETWORK AND TRAFFIC MANAGEMENT

The advanced sophistication of modern switching systems has brought about increased emphasis on network and traffic management. With the amount of calls being processed each hour of each day, system failures can prevent millions of people from completing their calls. Lost calls mean lost revenue. It is extremely important that subscribers complete their calls and that the quality of service is high. Most telephone companies employ elaborate network management systems to report on the health and welfare of switches and transmission equipment. Using highly sophisticated processors and software, they extract live performance data and analyze it to determine trouble spots. The basic functional areas of a network management systems are:

1. *Fault Management.* Detection, isolation, correction, and prevention of faults affecting service.
2. *Performance Mmanagement.* Monitoring and controlling the quality and integrity of the system.
3. *Trouble Administration.* Administration of trouble calls from customers reporting service problems.
4. *Configuration Management.* Maintenance of records on software, hardware, network database, and subscriber configurations.
5. *Accounting/Billing Management.* Collection of usage data and generation of billing information.
6. *Service Provisioning.* Adding, deleting, and changing customer service.
7. *Traffic Engineering.* Data collection, analysis, and service planning to ensure that system performance meets design goals.

A key area of telephone network performance is in traffic engineering. Telephone systems are not built to provide for 100% call completion for all subscribers at once. Calling pattern studies indicate that not all people call at the same time and there are peak periods of calling based on day of the week and time of day. Traffic engineers operate on the premise that a small percentage of the total number of subscribers will be using the telephone at any given time. The percentage varies between business area locations and residential locations as well as by regions. The average number of callers at any given time is on the order of 10–20% of the total number of subscribers. Networks are built to handle the predicted average busy hour/busy day traffic. Traffic engineering uses a unit of measurement called a CCS (100 call seconds) to indicate the load on a system. Because calls vary in number and duration, the CCS helps to quantify the amount of traffic in a single unit of measurement. For instance, 10 calls at 6 minutes duration each presents the same load to a network as 60 calls at 1 minute duration. The total time is still 60 minutes (3600 seconds), or 36 CCS (36-100 call seconds). The term 36 CCS is also referred to as an Erlang and is used to determine the number of trunks needed to support call traffic between switching offices. The average Class 5 End Office is engineered for 3.3 to 3.6 CCS for residential phones and 6.0 for a commercial environment.

Another important unit of measurement in traffic engineering is the grade of service (GOS) provided to the customer, which is the probability of a call being blocked (busy tone). Blockage can occur in either the switch or between switches if all trunks are being used. A key area of blockage in switches is the matrix. There are only a limited number of paths engineered into the design, and when they are fully utilized, no more calls can be made. Modern switches are designed to provide virtual nonblocking service, which means that enough capacity has been engineered into the switch to handle the normal amount of calls plus some level of extra capacity for stress conditions. However, in cases of local emergencies, when everyone wants to call at once, the demand can exceed design capacities and calls will be blocked. With blockage between switches, the same principle exists. There are only so many trunks available, and when that amount is being fully utilized, the next call will be blocked. Traffic studies examine the busy hour calling volume, quantified in CCS or Erlangs, and determine the number of trunks needed to support the normal service load to meet the desired blocking probability. If the probability that 1 call in 100 will be blocked (get a busy signal), the GOS is said to be P.01. If 5 calls are blocked out of 100, the GOS is P.05. Telephone companies consider the cost of trunking and equipment to determine the optimum level GOS needed (nominally in the range of P.01 to P.02) to provide the best service at the least cost (17).

ADVANCED TELEPHONE NETWORK SERVICE

Integrated Service Digital Network

Integrated services digital network (ISDN) was introduced in the late 1980s as a means of providing to subscribers improved end-to-end digital connectivity at speeds up to 128 kb/s using the existing two-wire cable plant. It maximizes the transmission capability of the existing local loop (connection between subscriber and EO) for the simultaneous transmission of voice and data. Basic rate ISDN subscribers have access to two bearer (B) channels operating at 56 or 64 kb/s each and a data signaling (D) channel operating at 16 kb/s. This basic rate interface (BRI) is normally called 2B + D. With this capability, users have the ability to make a voice and data call at the same time or a single data call at speeds up to 128 kb/s. Explosion of Internet service and telecommuting markets have made ISDN a popular residential and small business offering. Figure 14 shows a typical BRI application. The network terminating device (NT1) converts the four-wire 2B + D signal into a special encoded signal capable of being transmitted over a standard two-wire twisted pair local loop. ISDN also offers higher bandwidth service known as primary rate interface (PRI), which offers 23 bearer channels and 1 signaling channel and is known as 23B + D. This offering provides wideband incremental 56/64 kb/s digital service up to 1.472 megabytes/s and in most cases is the ISDN equivalent to a 24-channel T1 trunk used to interconnect switching systems. The most common use of PRI is for video teleconferencing systems that dial up bandwidth at multiple rates of 64/56 kb/s.

Advanced Intelligent Networks

An advanced intelligent network (AIN) is a service-independent network that takes the intelligence currently imbedded in switch manufacturers' software and outboards it to separate intelligent processors (IP). This provides the network operator with the capability to develop and control services more effectively and severs the tie between switch manufacturers and service providers (telephone companies) in the development of feature offerings. In the mid-1980s, commercial operating companies began asking for more control over how and when features were implemented. In many cases, one switch manufacturer would not have the correct mix of fea-

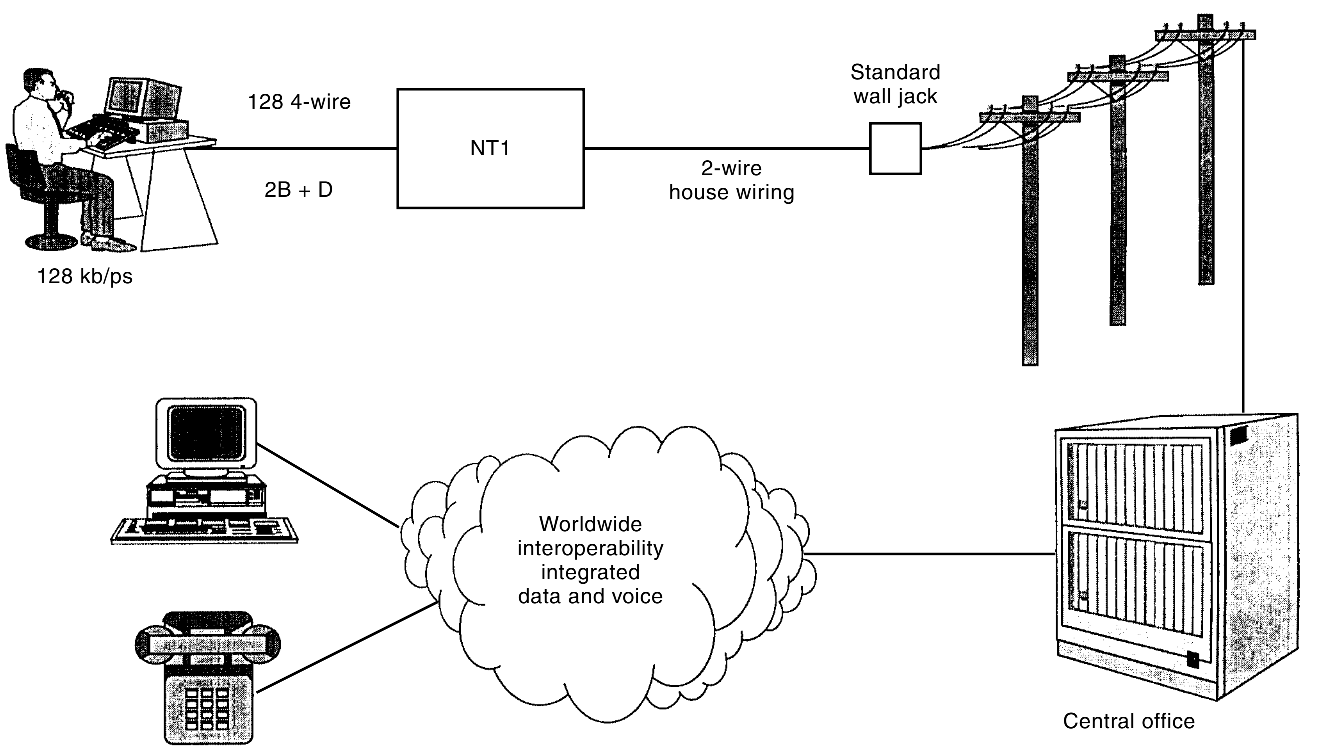

Figure 14. Basic ISDN application.

tures that a telephone company wanted to provide to its customers. This AIN concept introduces standard trigger detection points (TDP) into the call processing software to deflect the call process to an IP for further processing when special features are invoked. Figure 13 shows how the intelligent processors are tied in through the common channel signaling system to obtain the full intelligent power of the network.

The development of programs for the IPs is done by a separate service creation environment (SCE) operated by an independent vendor or by the telephone company. With this capability, telephone companies can develop and control their own special features and offer them without waiting for the switch manufacturers to provide them in a future software release. Some examples of the expanded feature sets that could be provided by AIN based systems are

1. Local number portability
2. Voice-activated dialing
3. Zip code routing
4. Area-wide networking
5. Intelligent routing, time of day, day of week, etc.
6. Calling party pays (wireless)
7. Do not disturb
8. Follow-me routing

Advances in switching technology are focused on improving call processing time, with experimental work being done with fiber optic switching systems. Future advanced intelligent networks promise to be feature rich and reliable, connecting subscribers at the speed of light. Wideband, global network switching will allow subscribers to call around the world using voice, data, and video at economical rates.

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TELEPHONE SIGNALING. See TELECOMMUNICATION SIGNALING.