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SYSTEMS AND APPLICATIONS

On the CD-ROM

SECTION 17

TELECOMMUNICATIONS

Telecommunication systems and applications deal with the actual technology by which devices communicate with each other. This includes transmission systems, switching systems, local area networks (LANs), wide area networks (WANs), broadband systems, cellular and mobile communication systems, and wireless digital communications. It is interesting to note that in areas of the world where the infrastructure for wire-based communication systems does not exist, it seems that development of such systems will be skipped and they will go directly to wireless.

Future trends dictate that we must communicate more information much faster, much more reliably, and send more transmissions over the same system. We will continue to see wireless technology becoming the dominant communication system. A very good example is the increase in ways of wireless connection to the Internet such as cell phones and PDAs. Clearly, security will also be a dominant force in determining how we communicate. Such security will need to be achieved without degrading the communication process and without making communicating more difficult.

Chapter 17.5, Wireless Networks, deals with technologies that combine Internet access with cell phone communication, with picture and email transmission, and walkie-talkie communication. These technologies must allow us to communicate and use information seamlessly, across a wide variety of devices and platforms.

In Chap. 17.6, we look at data networks and the Internet. LANs, metropolitan area networks (MANs), and WANs must be able to cope with the explosion in volume of information communicated at extremely high speeds over hard wire, fiber, and wireless systems.

This explosion in wireless technology and all the resulting capabilities it enables requires more and more sophisticated micro-miniaturized circuits and systems, thus becoming a major driver for MEMS technology. This is fully covered in Chap. 17.8. C.A.

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CHAPTER 17.1
TRANSMISSION SYSTEMS

A. B. Brown, Jr., Virgil I. Johannes

INTRODUCTION

A. B. Brown, Jr.

Service Networks

Communication entails the transfer of information from one point to another; the information may be in any of several forms, including voice, data, video, and facsimile. The facilities used for any particular form of service are known as a *service network*. A service network is made up of *terminals*, through which the information enters and leaves the network; *transmission facilities*, which provide the transfer of information from place to place, and *switching*, which connects the appropriate transmission facilities to cause the information to be delivered to the desired place. Some service networks do not include terminals, and some do not include switching. Terminals are of many types, including telephones, teletypewriters, facsimile terminals, and computer ports.

Interfaces

The boundary of the service network, where the user interacts with it (as by means of some piece of terminal equipment) or connects equipment to it is the *interface*. If the interface is digital, the use of digital transmission and switching obviates the need for digital-to-analog (D/A) and analog-to-digital (A/D) conversion of the signals. Contemporary networks with digital interfaces are used for digitally encoded video or voice as well as for high-speed data. Some older networks are used only for data.

Codes and Protocols

Communication of digital data is usually done in the form of characters, each of which is represented as a sequence of bits. Less commonly, the signals may be a bit stream generated and interpreted by the terminal equipment. The number of bits per character and the correspondence between the bit sequence and the character represented is called a *code*. Several codes have been used at various times for telegraphic and computer communications, starting with the Morse dot-dash codes used by telegraphers. The most common standard code at the present time is International Alphabet no. 5 (known in the United States as the American Standard Code for Information Interchange, ASCII). Other codes have been in use for older types of transmission equipment (teleprinters), and some codes have been introduced by individual manufacturers.

In coded transmission, two methods of maintaining synchronism between the transmitting and receiving points are commonly used. In *start-stop transmission*, the interval between characters is represented by a steady 1 signal; the transmission of a single 0 bit signals the receiving terminal that a character is starting. The information bits follow the start bit and are followed by the stop pulse. It is the same as the signal between characters and has a minimum length that is part of the specification of the terminal: 1.0, 1.42, 1.5, or 2.0 b are commonly used. In the synchronous method, the bits are sent at a uniform rate; if there is no character ready to be sent when it is time to send one, a synchronous idle pattern is used to maintain the timing. The synchronous method is used at higher speeds.

Protocols are standard procedures for the operations of communication; their purpose is to coordinate the equipment and processes at interfaces and at the ends of the communication channel. The International Telegraph and Telephone Consultative Committee (CCITT) recommendations in the X series, which apply to public data networks, include X.1, speeds; X.2, user options; X.20, interface for start-stop transmission services; X.21, interface for synchronous operation; X.22, multiplex interface for synchronous terminals; X.24, definitions of interface circuits; and X.121, international numbering plan for public data networks. Additional recommendations apply to packet networks.

Network Planning Objectives

The objectives that form the basis of network planning are derived from the service the network is to provide. There may, however, be different objectives for a network corresponding to different intended uses by the customers. For example, most of the objectives of the public telephone network are derived from the needs of voice communication; however, some objectives, e.g., the limits on delay distortion and impulse noise, concern its use for data communication. In turn, network objectives are the bases for the design objectives of the facilities which will compose the network. In practice networks evolve while they are in service, usually with more stringent objectives to meet the rising expectations of the users and with improved performance made possible by improved technology and designs. Thus, additions to the network may be designed according to objectives different from those of the earlier parts.

Packet Networks

Among the data networks provided up to the mid-1960s, both the circuit-switched and message-switched types were in use. In the first, a channel was assigned full time for the duration of a call. In the second, a message or section of a serial message was transmitted to the next switch if a path (loop or trunk) was available; if not, it was stored until a path became available. The use of trunks between the message switches was often very efficient. In many circuit-switched applications, however, data were transmitted only a fraction of the time the circuit was in use. In order to make more efficient use of facilities and to make it possible to charge users only according to the amount of data transmitted, packet networks were developed. In a packet network, a message from one terminal to another is divided into *packets* of some definite maximum length, often 128 octets (corresponding to bytes in a computer) of data; the packets are sent from the origination point to the destination individually. Each packet contains a header, which provides the network with necessary information to handle the packet. The packets transmitted by one terminal to another are interleaved on the facilities between the packets transmitted by other users to their addresses, so that the idle time of one source can be used by another. At the destination switching center, the message is reassembled and formatted before delivery to the called station. In general, a network has an internal protocol to control the movement of data within the network. The internal speed of the network is in general higher than any of the terminal speeds, so that there is a change of speed from source to network and from network to destination.

The same physical interface circuit can be used for communication with more than one other terminal or computer at the same time by the use of *logical channels*. At any given time, each logical channel is used for communication with some particular addressee; each packet includes the identification of its logical channel, and the packets for the various logical channels are interleaved on the physical-interface circuit.

Three methods of handling messages are in common use. *Datagrams* are one-way messages sent from an originator to a destination; the packets are delivered independently, not necessarily in the order they were sent. Delivery and nondelivery notifications may be provided. In *virtual calls*, packets may be exchanged between two users of the network; at the destination they are delivered to the addressee in the same order as they were originated. *Permanent virtual circuits* also provide for exchange of packets between two users of the network; each assigns a logical channel, by arrangement with the provider of the network, for exchange of packets with the other. No setup or clearing of the channel is then needed. Some packet networks support terminals that are not capable of formulating the data in packets, by means of a packet assembler and disassembler included in the network.

Other packet networks have been set up since ARPANET (the earliest major packet network in the United States) to provide packet communication service to commercial users, in the United States, Canada, France, Spain, Japan, and elsewhere. An early commercial network in the United States provides service at speeds up to 56 kb/s. The service is either common-user or with closed user groups; the latter service prevents communication into or out of a given group of stations, the equivalent of a private network. The network is arranged in a hierarchy of switching nodes, the higher nodes being redundantly connected by 56-kb/s trunks. The transit delay for a packet is less than 200 ms. The network operates in the virtual call mode.

Protocols for packet networks are the subject of several Recommendations of the CCITT. Recommendations X.3, X.28, and X.29 cover start-stop mode terminal handling; X.25 covers the interface for packet-mode data terminal equipment, X.75 covers the interface for packet-network interconnection, and X.244 covers exchange of protocol identification during virtual call establishment.

In early packet networks, routing of each packet in a message was independent; each packet carried the address of its destination, as well as a number to permit arranging the packets in the proper order at the destination. Later networks use a *virtual circuit*, which is set up at the beginning of a call and which contains the routing information for all the packets of that call. The packets after the first contain the designation of the virtual circuit. In some networks, the choice of route is based on measurements, received from all other nodes, of the delay to every other node in the network. It is sufficient for a node to measure and transmit to the other nodes the delay to nodes to which it is directly connected.

Integrated Service Digital Networks (ISDNs)

An ISDN uses a single digital transmission network to provide a wide variety of services, such as voice, text, facsimile, videotex, and video, both switched and nonswitched, and both circuit- and packet-made. See Chap. 17.4.

TRANSMISSION SYSTEM OPERATIONS

Virgil I. Johannes

Transmission System Principles

Transmission Systems. A telecommunications transmission link can be either a *loop*, which connects a subscriber with a switch, or a *trunk*, which connects two switches. A loop may consist of a *feeder* portion, connecting the switch at the central office with a terminal near the subscriber, and a *distribution* portion connecting the terminal to the subscriber. Transmission can be at voice frequency over wire pairs, or a number of voice-frequency channels can be multiplexed together using frequency-division techniques (analog carrier) or time-division techniques (digital carrier). The multiplexed signal can then be transmitted over guided wave media, such as wire and optical fibers, or through free space, as in radio systems.

Digital carrier provides higher quality transmission than analog, as the signal can be completely regenerated when necessary without the addition of noise and distortion that is characteristic of analog carrier. Digital carrier over fiber-optic cables is also more economical than analog carrier on cable. As a result, new carrier systems being installed throughout the world are almost exclusively digital, and in the United States at least, most analog carrier systems have been removed from service, leaving only vestiges of a once ubiquitous analog carrier network.

Transmission at voice frequency is commonly used in loops, and for some short trunks between analog switches, such as the 1A ESS, as the cost of converting the signal to and from digital form can be avoided. Loops to customers with digital PBX (Private Branch Exchanges) are often digital carrier, and there is increasing use of digital carrier on loops longer than 3 to 5 km. In this latter technique, signals are converted to and from digital form in equipment near the customer, and the final connection to the customer is at voice frequency over wire pair. Direct digital connection to the residential and small business customer is not widely available, but there are a few installations, and many proposals, both technical and regulatory, for bridging this "last mile." These include use of wire pairs from the central office to provide a 144 kb/s digital link, in connection with the ISDN, "fiber to the home" schemes with much higher capacity, and "fiber to the curb" schemes in which the feeder portion of the loop is fiber, while the distribution part is copper.

Voice-Band Transmission and Analog Carrier. Voice-band (200 to 3500 Hz) transmission over a pair of wires in a cable is widely used for loops, and for trunks between nearby switches. For lengths of a few miles or more, the cable is often *loaded* to improve voice-band performance.

In a loop, wire gauge and loading are normally selected to keep the loop resistance below 1300 Ω, which permits proper operation of the supervisory and transmission circuits. Loops up to about 3 mi can be 26-gauge pairs. Loops up to 5 mi can be 24-gauge, and longer loops are often of mixed gauges with the thinner wire closer to the exchange. An alternative approach (*long-route design* or *unigauge*) is to use 26-gauge only (up to 2800 Ω loops) and compensate for the additional resistance with audio gain, equalization, and increased signaling power at the office.

Voice-band trunks were once widely used among local and tandem offices but with digital switches, digital carrier connection is generally preferred. Voice-band trunks are still used for shorter connections among analog switches. Such trunks can be either two-wire, in which both directions of transmission are carried on a single wire pair, or four-wire, preferred for longer trunks, in which the two directions of transmission are carried on separate pairs. Use of amplifiers (repeaters) along the transmission path to compensate for the loss of longer cables is now rare, any gain required being included in a terminal near the switch, which also incorporates any other functions required, such as two-wire to four-wire conversion, and equalization of the frequency response. Voice-band trunks are typically designed to have a loss of about 6 dB (after any amplification) between local offices, and less for tandem office connections.

Typical analog carrier transmission involves single-sideband suppressed-carrier modulation of a voice-band signal, thus shifting the signal to a higher frequency. Such shifted signals, each typically occupying a 4-kHz bandwidth, are then combined into groups of 12 or 24 voice channels, and then further modulated and combined into 300 or 600 channel mastergroups. Short haul systems that carry groups, and long-haul systems carrying many mastergroups on a single coaxial, microwave radio, or satellite facility, while no longer being installed, are still in service in some areas, mostly outside North America. The SG submarine cable system, for example, carries 4200 two-way voice circuits in a single coaxial cable with dielectric diameter of 1.7 in. and has been in service across the Atlantic since 1976.

Digital Transmission Formats and Hierarchies

The first telecommunications digital transmission system, (T1) put into service in the United States in 1962, was designed to carry 24 voice channels over distances up to 50 miles. Each voice channel was sampled 8000 times per second, and each sample represented by an 8-bit byte, thus occupying 64 kb/s. The line signal was assembled by taking 1 byte from each channel in sequence (for a total of 192 bits) and then adding a single bit, the framing bit, to make a 193-bit frame, with a line rate of 1.544 Mb/s. At the receiving end, the framing bit was used to identify the other bytes so that they could be associated with the appropriate voice channel. In this initial design, an alarm was given, and the system was taken out of service, if more than 1 percent of the received framing bits failed to match the transmitted 1-0 pattern for about 300 ms. T1 systems were point-to-point, independent of each other, and the exact line rate of each system was determined by its own crystal clock, with an accuracy of about ±50 ppm. This system became widespread within a few years in the United States, Canada, and Japan. Subsequently, a similar system, with 8 kHz sampling, and 8 bits per sample, but carrying 30 channels at a line rate of 2.048 MHz, and differing in coding technique as well, was developed extensively in Europe. These are referred to as *primary rate* systems.

As the use of the primary rate systems increased, higher bit-rate systems were developed for longer distances and more economical transmission. The line signals for these higher rate systems are assembled from the primary rate signals in one or more steps using bit-by-bit time division multiplexing, and pulse stuffing. Two separate hierarchies of rates, each based on one of the two primary rates, along with rules for multiplexing and demultiplexing have been standardized. These are shown in Table 17.1.1, along with a variant used only in Japan. In the United States, transmission systems at rates other than the hierarchical rates are more the rule than the exception, and the

TABLE 17.1.1 Hierarchical Rates in the Plesiochronous Digital Hierarchy—kb/s

North America	Japan	Europe
64	64	64
1544	1544	2048
6312	6312	8448
44736	32064	34368
	97728	139264

6.312 Mb/s rate is little used. Higher rate signals for transmission are assembled from whatever number of DS-1 or DS-3 signals may be appropriate for the application at hand, using the pulse-stuffing technique. Thus the existing U.S. network has transmission systems carrying one, two, and four primary rate signals, in which the standardized primary rate signals appear only at the ends, as well as systems operating at various multiples of 44.736 Mb/s in which again the standardized signals appear only at the ends. The termination of these nonhierarchical rate systems in standardized signals allows for ready interconnection and rearrangement. All these rates and variations in both hierarchies are sometimes referred to, in the aggregate, as the *plesiochronous digital hierarchy* (PDH), in reference to the derivation of the various rates from independent timing sources that are not necessarily synchronized.

While systems above the primary rate generally include parity check bits for monitoring of the performance, the primary rate systems were initially monitored by examining the signal for violations of the line code and these violations (but not the errors!) disappear when the signal is multiplexed to a higher rate. To provide some means of determining the end-to-end performance of a DS-1 signal that may be multiplexed and demultiplexed many times between source and destination, newer DS-1 sources use the 193rd bit not only for framing, but also for error detection using a cyclical redundancy check code. A similar provision has been added to the 2.048 Mb/s signal. Even with these additions, the provisions for monitoring, maintenance, and network control provided in the PDH are not well suited to the worldwide digital network, and the existence of two separate hierarchies makes interconnection cumbersome.

In 1988, a new set of rates and multiplexing rules for higher bit-rate systems was standardized by the CCITT.* This new hierarchy includes a single set of rates intended for worldwide use, to transport signals from either of the two older hierarchies. There is also extensive provision of additional bits for monitoring and network operation. Recognizing the present availability of synchronized timing sources, the new hierarchy is nominally synchronous (hence the name, *synchronous digital hierarchy*, or SDH) although there is provision for small frequency variations, to account for the possibility that synchronism among parts of the network may be lost from time to time. In the United States, this hierarchy is part of a more extensive set of standards, referred to as the Synchronous Optical Network (SONET). SONET is intended also to encourage use of line systems and a standardized set of optical interfaces at the established rates of the hierarchy. The lowest rate envisioned is 51.84 Mb/s, appropriate for carrying a DS-3 signal.

Media for Transmission

Paired Cable. Paired telephone cables have been described earlier. Trunk and loop cables are similar, except that to reduce loss trunk cables often use thicker wire and therefore include fewer pairs (900–pair cable with 22 gage pairs is typical). The performance of carrier systems on paired cable is generally limited by crosstalk. Near-end crosstalk is more significant than far-end crosstalk. Some trunk cables are available with an internal shield (screen) to separate the directions of transmission and thus reduce near-end crosstalk, and in some applications a separate cable is used in each direction (two-cable operation).

Coaxial Cable, Waveguide. Coaxial cable for telecommunication trunks has been made with as many as 22 coaxial tubes, stranded and sheathed as for paired cable, and often including some wire pairs for maintenance. Tube diameters have ranged from 2.9 mm (in Europe), through 0.375 in. (once widespread in the United States), to 1.7 in. (for single tube submarine cable). Such cables were widely used for medium- and long-haul transmission before the advent of optical fiber, and many remain in service outside the United States. Circular waveguide systems operating at mm wavelengths had been used in some experimental installations before being obsoleted by optical fiber. (See also Chap. 17.3.)

Optical-Fiber Cables. Optical fibers for telecommunications are covered in Chap. 17.3 and they are typically about 125 μm thick and are made of very pure silica with dopants added to control the index of refraction. They consist of a doped silica glass core with a glass cladding having a 0.3 to 1 percent lower index of refraction, which

*International Consultative Committee on Telephone and Telegraph, recently renamed the Standards Sector of the ITU (International Telecommunications Union). The CCITT formerly issued its standards in the form of “Recommendations” every four years in volumes identified by their color. Current practice is to issue or revise individual Recommendations from time to time.

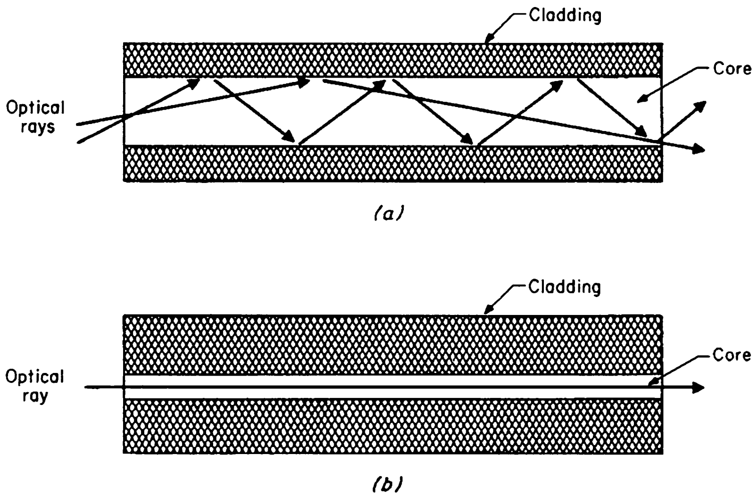


FIGURE 17.1.1 Optical fibers: (*a*) Multimode fiber; (*b*) single-mode fiber. (*From Paul S. Henry, "Introduction to Lightwave Transmission," IEEE Commun. Mag., Vol. 23, No. 5, p. 13. © 1985 IEEE. Used by permission*)

serves to confine the light within the fiber. If the core is thick enough (50 μm), light at the wavelengths used (0.8 to 1.55 μm) will propagate in several modes, as shown in the multimode fiber of Fig. 17.1.1*a*. A thinner core (0.85 μm) can restrict propagation to a single mode as shown in Fig. 17.1.1*b*. Single-mode fibers are more difficult to splice and couple, but usually have lower loss because of lower dopant concentration in the core.

Radio and Free Space. Microwave frequency bands allocated to telecommunications within the United States are shown in Table 17.1.2. (Except for mobile applications there is little present use of other frequency bands.) At these frequencies propagation is line-of-sight, so is limited by the curvature of the earth. At typical relay tower spacings of 15 to 30 mi, the free space loss is about 140 dB in the 4- and 6-GHz bands. With antenna gains of about 40 dB, the normal loss from antenna input to output is about 60 dB. Loss well above the normal can occur because of propagation effects, as discussed in Sec. 16. At frequencies above 10 GHz, the additional attenuation resulting from heavy rainfall is often the limiting factor. Path lengths 10 to 15 mi in the 11-GHz band, and 1 to 3 mi in the 18-GHz band are typical. The deployment of microwave systems is often limited by the need to avoid interference with existing systems using the same frequencies. These same microwave bands are used for satellite links.

Rainfall and other propagation problems have mitigated against the use of optical, or near optical, wavelengths in free-space transmission.

TABLE 17.1.2 Selected United States Common-Carrier Microwave Frequency Allocations

Band, GHz	Allotted frequencies, MHz	Bandwidth, MHz
2	2,110–2,130	20
	2,160–2,180	20
4	3,700–4,200	500
6	5,925–6,425	500
11	10,700–11,700	1,000
18	17,700–19,700	2,000
30	27,500–29,500	2,000

Voice Encoding, Circuit Multiplication, Echo Control

Voice Encoding. To prepare a voice signal for digital transmission or switching, it is band-limited to about 3500 Hz and sampled at 8 kHz; each sample is encoded into 8-b PCM, producing a 64-kb/s signal. The quantization of the signal to one of $2^8 = 256$ levels that is inherent in the coding process produces noise, termed *quantizing noise* or *quantizing distortion*, in the decoded signal. (This is the major impairment suffered by a voice signal transmitted digitally.)

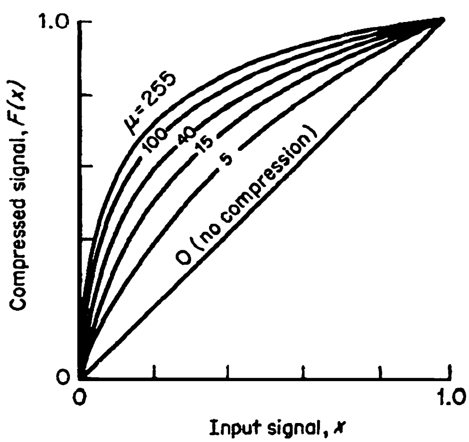


FIGURE 17.1.2 Logarithmic compression characteristics. (*Copyright 1982, Bell Telephone Laboratories, Inc. Reprinted by permission*)

To improve the signal-to-distortion ratio at small-signal levels, a logarithmic encoding law is employed in order to use a greater portion of the available levels for weak signals. In this way, a certain *percentage* change in the instantaneous signal corresponds closely to a change of one encoding level, and the signal-to-distortion ratio is approximately constant over a wide range of talker volume. This process can be thought of as compressing the signal in amplitude according to one of the curves of Fig. 17.1.2 and then performing a linear A/D conversion. In current practice, however, a piecewise linear approximation to the logarithmic curve is incorporated in the coder proper and, of course, in the decoder as well. The two approximations in current use (A-law in Europe, μ-law in North America and Japan) are shown in Table 17.1.3. Both meet the CCITT requirement of 33-dB signal-to-quantizing-distortion ratio for sine waves from 0 to –30 dBm0, with overload point at about 3 dBm0. Idle channel noise for the μ-law coder is specified as less than 23 dBrnc0. (In this, as in other sine-wave specifications of coders, exact submultiples of the 8-kHz sampling frequency are excluded.) In the μ-law code, the all-zero code word is not used, slightly increasing the quantizing noise.

Some coding schemes for voice which use less than 64 kb/s found use in special applications even prior to the CCITT standardizing a 32-kb/s ADPCM (adaptive differential pulse code modulation) method. In differential PCM, the *difference* between sampled values of the voice signal is coded rather than the values themselves. The adaptive feature, an improvement on the logarithmic coding law, changes the code in accord with a complex algorithm dependent on past signal amplitudes. A major problem in finding an acceptable 32-kb/s coding method was providing acceptable performance for voice-band data signals from data modems. This is even more of a problem with 16-kb/s coding, and along with decreasing cost of high-capacity transmission systems, and increasing nonvoice traffic on the network, has limited the application of these lower bit-rate coding schemes.

TABLE 17.1.3 CCITT Recommended Coding Laws for Voice

Name	Basic law $0 \le \lvert x\rvert \le 1$	Parameter value	Segments in linear approximation	Primary multiplex rate, Mb/s
μ law	$y = \dfrac{\ln(1+\mu\lvert x\rvert)}{\ln(1+\mu)}$	$\mu = 255^*$	15	1.544
A law	$\dfrac{1+\ln A\lvert x\rvert}{1+\ln A} \quad \dfrac{1}{A} \le x \le 1$	$A = 87.6$	13	2.048
	$\dfrac{A\lvert x\rvert}{1+\ln A} \quad 0 \le x \le \dfrac{1}{A}$			

*μ = 100 was used in original T1 system.

Circuit Multiplication, Time-Assignment Speech Interpolation. In a typical telephone conversation, only one party is talking at a time, so each direction of transmission is idle half the time. When pauses are also taken into account, the actual utilization is about 40 percent. Equipment taking advantage of this phenomenon to increase the number of voice channels carried by a transmission link is widely used on long distance international transmission links on submarine cable and satellite. It was first introduced on analog submarine cable systems, and termed TASI (time assignment speech interpolation), as the analog channels available were assigned to various talkers as they became active. The TASI systems achieved a 2:1 increase in capacity in practice. Similar systems, sometimes called DSI (digital speech interpolation) have been deployed on digital satellite links as well, with similar results. The combination of interpolation techniques with a code conversion from the 64 kb/s PCM signal to 32 kb/s ADPCM allows an increase of as much as 5:1 in capacity for digital voice applications. (The signal is restored to 64 kb/s at the receiving end.) This combined function is referred to as *circuit multiplication*, and is in general used on satellite and intercontinental submarine cable digital links.

Circuit multiplication introduces some slight degradations to voice signals, introduced by the conversion to and from ADPCM, the finite time to recognize the beginning of speech (processing clip), and the occasional need for more channels than are available. (The effect of the latter can be mitigated by reducing the number of bits for all conversations, rather than cutting off some conversations for short intervals.) Circuit multiplication equipment must detect nonvoice signals, and pass them without alteration, as they would not be restored correctly. This detection may be based on the 2000–2200 Hz tone emitted by voice-band data modems.

Echo Control. If the round trip delay is less than 45 ms, acceptable echo performance can be achieved by designing all trunks to have a loss that increases the return loss of the echo. Voice-band and other analog trunks were traditionally assigned losses in the range of 0.5 to 8.9 dB, according to an overall plan called the Via Net Loss plan. With digital transmission and switching, which are inherently lossless, 3 dB is typically inserted at both ends of a connection, and no attempt is made to add loss to intermediate trunks. Above a 45-ms delay, the loss required for acceptable echo performance would be excessive, and echo cancelers are used. An echo canceler incorporates an internal delay line with taps that are adjusted to produce a signal duplicating the echo, which is then subtracted from the received signal, canceling the effect of the echo. The adjustment is done rapidly and continuously. One echo canceler is necessary for each voice channel, but economical integrated circuit realizations are available.

The Digital Transmission Network

Network Structure. The structure of the existing digital transmission network, based on the plesiochronous digital hierarchy, is illustrated in Fig. 17.1.3, which shows a cross-section of equipment arrangements and signal flows applicable to any of the hierarchies shown in Table 17.1.1.* Primary rate signals may be coded voice signals multiplexed together, with the encoding and multiplexing done in a unit called a *channel bank* or *Primary PCM Multiplex* as described above for the original T1 system. Digital switches typically include the coding and multiplexing functions as well, and primary rate signals can originate from other sources, such as computer data and coded video. Primary rate signals destined for a remote location may either be transmitted directly, typically over a wire pair, or further combined into higher-rate signal by the multiplexes for more economical transmission, as illustrated in Fig. 17.1.3. The *Line Terminating Equipment* of the figure include such functions as power feeding for a repeated line, error monitoring, switching from a failed line to an operating one (protection switching), and conversion of the signal to the line code. (These functions may actually be packaged with the multiplex.) In the United States, the 6.312 Mb/s line rate and crossconnect capability have not found much use, so direct multiplexing of DS-1 signals into DS-3 signals in a single piece of equipment without access to the DS-2 signal is the norm.

Figure 17.1.3 is representative of equipment found in telephone company offices. The offices are connected by transmission line systems using cable or radio, and often include regularly spaced repeaters, which serve to amplify, retime, and regenerate the received digital signal. The line systems do not necessarily operate at a hierarchical rate. They do, however, have interfaces at hierarchical levels (in the United States at DS-1 and DS-3), and at standardized signal levels and format, and thus are readily interconnectable with other line systems,

*Interconnection between the two hierarchies is based on the common 64 kb/s rate. Conversation between A and μ-law coding and remultiplexing into appropriate format is performed at a connection point in the μ-law country, and often associated with circuit modulation equipment. Some United States–Europe links use a "hybrid hierarchy" in which the multiplexing sequence is 64 kb/s (A-law), 2.048 Mb/s, 6.312 Mb/s, 44.736 Mb/s, and 139.264 Mb/s.

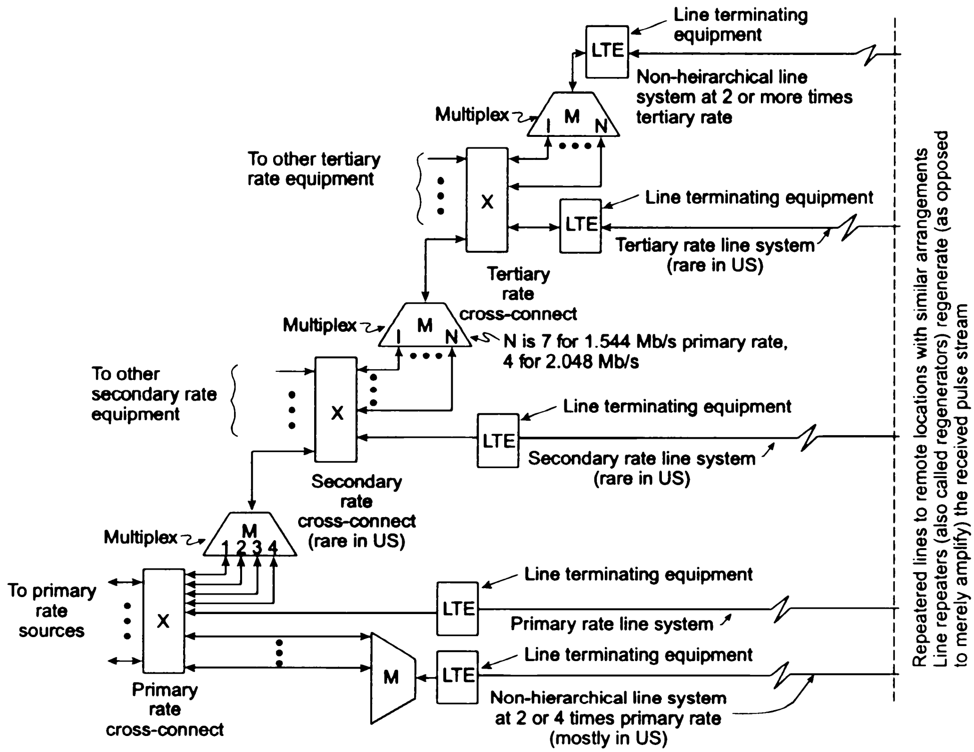


FIGURE 17.1.3 Functions and equipment in the plesiochronous digital hierarchy.

signal sources, and so forth. The line systems proper are proprietary, and terminal equipment at both ends, as well as intermediate repeaters, are generally from the same manufacturer, or at least a cooperating group of manufacturers. The repeaters may be powered locally, if they are located in a telephone company office, for example, or via power fed over the cable. Paired cable systems are typically powered over the pairs that carry the signals, and conductors for the purpose may be provided in optical fiber cables.

Crossconnects. To facilitate interconnection and rearrangements, all signals of a given hierarchial rate in a telephone company office are brought together at the *crossconnects* of Fig. 17.1.3. These crossconnects may be simple wiring panels at which the various sources, multiplexes, and line terminating equipments are interconnected and accessed for testing by manually placed wiring. In the United States, such a crossconnect is referred to, for example, as a DSX-1 for a crossconnect at the primary level, and so on. A more elaborate alternative to simple wiring panels is the digital crossconnect system (DCS, or DACS for digital access and crossconnect system). These provide electronic switching of the signals and include multiplexes so that *components* of the signals may be interchanged. Thus a primary level DCS can reroute entire 1.544 Mb/s signals, and interchange 64 kb/s channels freely among such signals. (This latter function is similar to that performed by digital switches, but the DCS is too slow to be used on a call-by-call basis, and does not incorporate the signaling capability of a digital telephone switch.) This requires synchronism of the various signals, which is achieved by synchronizing all connected switches and channel banks from a master clock.

Equipment incorporating the line terminating functions, multiplex, and electronically switched crossconnect in a single unit is also available, offering space, cost, and operational savings.

Time-Division Multiplexing: Framing. If two or more synchronous digital signals are to be multiplexed, they can be adjusted in phase and pulse width and combined directly, with some method for identification of the signals for later demultiplexing. A common method of identification is to group a number of input bits into a *frame* with an additional framing bit at the beginning of the frame. For example, in the 24-channel PCM channel bank, twenty-four 8-b words, each representing one sample, constitute 192 b of a 193-b frame, and the 193d bit is the framing bit. If no other features were desired, the framing bit could be a simple pattern such as 101010, which would allow frame to be recovered at the receiving channel bank by finding a position in the received pulse stream which alternates between 0 and 1 at intervals of 193 b. To provide additional functions while maintaining the T1 line rate, an *extended superframe format* involving a framing pattern that identifies groups of twenty-four 193-b frames is used. In this format, only six of the 24 slots allocated to framing bits have a fixed pattern used for framing, while six slots are used for cyclic redundancy check bits (e.g., an error-detecting code covering the entire superframe of 24 frames) and 12 slots are used for 4-kb/s data link for maintenance communication.

This use of a single framing bit at the beginning of a frame is common in the United States at all rates. Another method, common in Europe, is to use a multibit framing alignment word at the beginning of a much longer frame. In Synchronous Digital Hierarchy, framing bytes 8 b are included in section overhead, and distributed throughout the frame.

Synchronization, Pulse Stuffing. Multiplexing to the primary level (1.544 or 2.048 Mb/s) is synchronous, and entire 8-b bytes are multiplexed. In channel banks and switches the locally encoded voice signals are synchronous and appear 8 b at a time, so that this is a natural mode of operation. Where 64-kb/s signals from more than one source are to be intermingled, e.g., in a digital switch, they are synchronized to a central reference which is distributed via designated digital facilities. In a digital channel bank connected to a digital switch, e.g. via a digital line, the digital switch drives the line toward the channel bank at a rate synchronous with the references and the channel bank is loop-timed; i.e., it derives its transmit frequency from the signal it receives from the switch, which in turn is timed from the network reference.

Above the primary level, signals normally arrive at the multiplex a bit at a time, and it is not easy to assure that all signals that arrive at a multiplex will be synchronous. Present practice, therefore, is to stuff pulses into each input digital stream to synchronize them all to a common, higher rate. The synchronous streams are then multiplexed bit by bit. This process, called *pulse stuffing* or *positive justification*, is illustrated in Fig. 17.1.4. The location of the stuffed pulses is signaled to the receiving end (to allow for removal) on still other bits added to the frames for the purpose.

The pulse stuffing-destuffing process introduces some jitter (undesired phase modulation) in the demultiplexed pulse stream, which is usually not troublesome as it is of low amplitude and frequency.

Superframes. The pulse-stuffing scheme described above requires a low-data-rate digital channel between multiplexer and demultiplexer to signal the presence or absence of stuffed time slots. Low-data-rate channels may also be required to provide communication between terminals for maintenance functions such as switching to a spare or for parity bits for detection of transmission errors. These low-data-rate channels are provided by adding bits that are located by establishing a superframe structure encompassing many information-bit frames. The extended superframe mentioned above is an example.

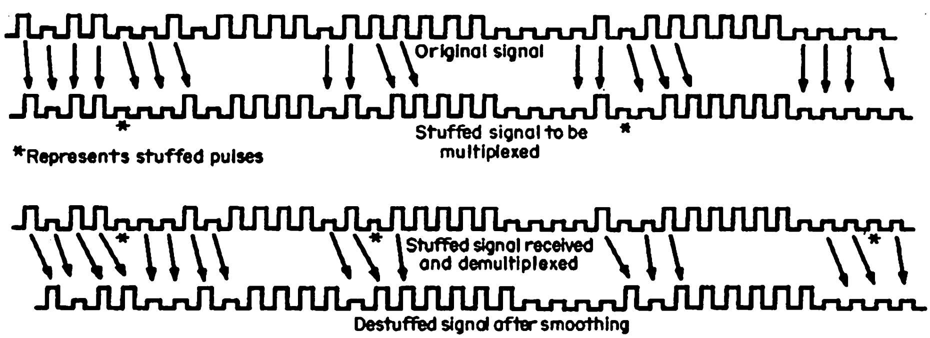


FIGURE 17.1.4 Pulse-stuffing synchronization. (*Copyright 1982, Bell Telephone Laboratories, Inc. Reprinted by permission*)

Regenerative Repeaters. A block diagram of a regenerative repeater is shown in Fig 17.1.5. The input signal is amplified and equalized, a timing wave is extracted in the clock-extraction circuit, and then the signal is regenerated; i.e., in each time slot a decision is made whether a 1 or 0 is present and a pulse accordingly applied (or not) to the output. The stylized waveshapes of Fig. 17.1.5 illustrate this action, and Fig. 17.1.6*a* shows an idealized "eye pattern" at the regenerator input, for a ternary repeater, i.e., one that can produce either a positive or negative pulse as well as a zero. An eye pattern is the superposition of waveshapes resulting from all possible pulse sequences (0 + 0, – + –, ⋯).

The frequency response of the equalized channel is deliberately rolled off, often with a cosine shape, to produce a pulse that may be as wide as two time slots at the base (as in Fig. 17.1.6*a*) rather than a close replica of the transmitted pulse. This reduces the noise and crosstalk on the equalized pulse. The adaptive equalizer acts to bring the peak of the received pulse to a standard amplitude and thus provides the proper equalization for any cable length within its range. (Several fixed equalizers may be required to span the complete range of cable lengths.) The adaptive equalizer also compensates for variations in cable temperature, and is usually analog, involving only a few singularities. Misequalization and other circuit imperfections can result in closing the eye (Fig. 17.1.6*b*), increasing the probability of error because of thermal or other noise.

Jitter in a repeatered line results from pattern variations in the signal transmitted. As a result of imperfect equalization and other idiosyncrasies of the repeater, these pattern variations appear as phase variations applied to the timing extraction filter (see below), and any components of the phase variation at a frequency within the bandpass of this filter appear as jitter on the timing signal and hence on the repeater output. The rms jitter of a line is approximately proportional to the square root of the number of repeaters. (For T1, for example, with a repeater tank Q of 80, the rms jitter for 10 repeaters is about 3°.) Jitter in the amounts usually encountered has little effect on 64-kb/s encoded voice signals. Jitter (unlike errors) can be reduced or eliminated completely at the endpoints by writing the receiving information into a buffer memory and reading it out under control of a stable clock. Thus the major concern in practice is to assure an adequate size of buffer memory where dejitterization intentionally or inadvertently takes place. (An example of inadvertent dejitterization would be the connection of a repeatered line to a terminal with an input repeater of higher Q.)

Clock extraction for a ternary repeater involves full-wave rectification of the signal, filtering the result in an LC tank or equivalent, in order to obtain a sine wave at the symbol rate, and then shaping the sine wave to obtain a sampling pulse or edge. The full-wave rectification produces a strong component at the symbol rate. The required Q of the tank circuit, or equivalent, depends on the line code selected. In any case, a higher Q in the filter reduces the jitter of the timing wave, which results from pattern variations in the digital stream. On the other hand, a higher Q increases the static offset of the sampling pulse because of temperature variations of the frequency of the tank. Arrangements used have included an LC tank with Q of 80 (T1), a monolithic crystal filter, a surface-acoustic-wave (SAW) filter, and phase-locked loop with crystal-controlled VCO (voltage-controlled oscillator).

The regenerator proper is a clocked flip-flop or similar arrangement with carefully controlled threshold. The output pulse is often clocked to about 60 percent of the time slot in ternary systems, but nonreturn to zero (NRZ) is usual in binary systems, such as fiber-optic systems.

Most regenerative repeaters have some provision for fault location, i.e., determining from an office which of the many repeaters in a failed line is at fault. This may be accomplished by feeding a small part of the output back to the originating station through a filter that passes only a particular audio frequency assigned to that manhole. To locate a faulty repeater, a digital pattern containing the particular audio frequency associated with a certain manhole is applied to the line. If the audio frequency then appears on the fault-locating pair, the repeater in that manhole and all repeaters upstream of that line are known to be operating. The procedure is then repeated with other audio frequencies until the defective repeater is located. All repeaters at a given manhole share the same audio filter, and all manholes share the same voice-frequency fault-locate pair. Recent designs for high-capacity fiber-optic and radio systems incorporate error detectors (based on parity bits or line code redundancy) in each repeater, with a method for interrogation from the terminals.

The probability that an ideal digital repeater with gaussian noise added to its input will make an error is the probability that the noise will exceed half the eye height, which for m-level transmission is

$$P_E = \frac{m-1}{m}\operatorname{erfc}\left|\frac{1}{(m-1)\sqrt{2}}\,\frac{\text{peak signal}}{\text{rms value of noise}}\right|$$

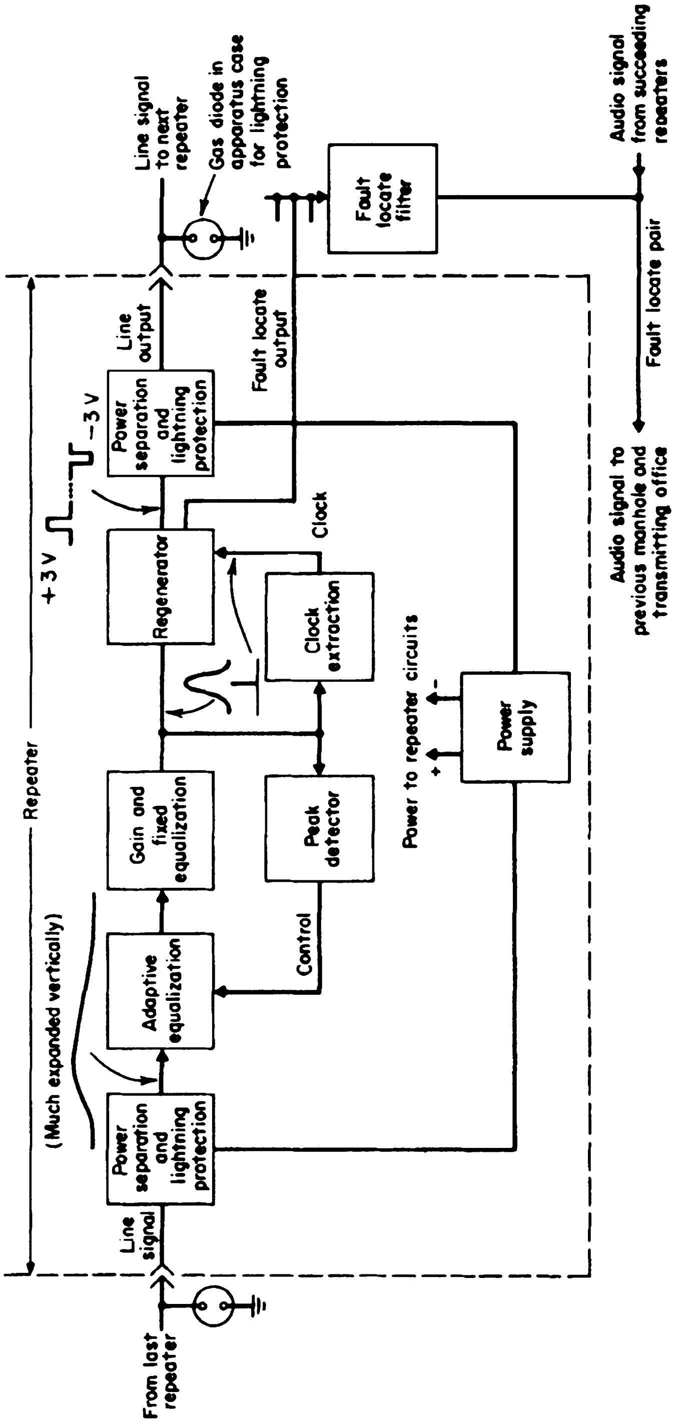


FIGURE 17.1.5 Representative regenerative repeater for cable system with associated external components.

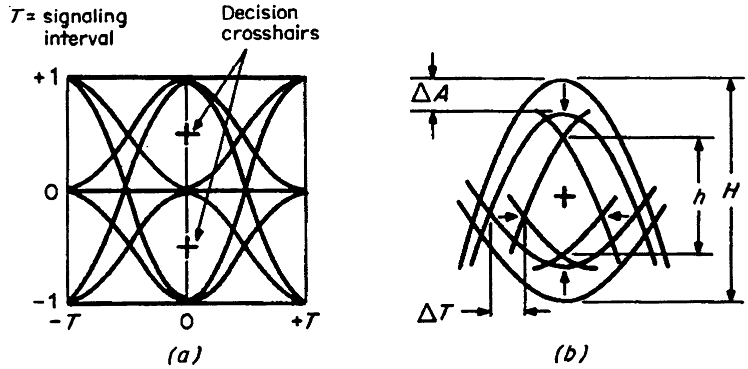


FIGURE 17.1.6 Eye diagrams for a ternary repeater: (*a*) ideal eye; (*b*) closing the eye to account for practical degradations (upper eye only). (*Copyright 1982, Bell Telephone Laboratories, Inc. Reprinted by permission*)

This is plotted in Fig. 17.1.7. It can be seen that in the range of usual interest, 10^{-6} or better, variation of a few decibels of signal-to-noise ratio produces a variation of many orders of magnitude in error probability. This is generally true in practice as well, even though crosstalk or some other interference, rather than gaussian thermal noise, may be the major cause of errors.

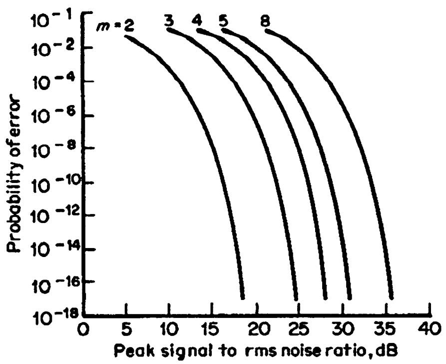


FIGURE 17.1.7 Probability of error vs. peak signal to rms gaussian noise for random *m*-level polar transmission. (*Copyright 1982, Bell Telephone Laboratories, Inc. Reprinted by permission*)

The features described above are typical of regenerative repeaters for use on cable systems. Repeaters on metallic cable systems normally are powered from a direct current carried on the same conductors as the signal, and they also require a power-separation filter, as shown in Fig. 17.1.5, while radio and fiber-optic repeaters do not. Fiber-optic regenerators are similar to regenerators for metallic cable, except that a light source and optical detector are used at the output and input. The light source may be a solid-state laser or light-emitting diode (LED). Early systems were restricted to wavelengths of 0.82 to 0.85 μm for which lasers and LEDs were available. (LEDs are less expensive but provide less light power to the fiber.) Once lasers at 1.3 μm (where the fiber has lower loss) with satisfactory lifetimes became available, they were widely incorporated in new systems, and some systems have appeared with 1.55 μm lasers, the wavelengths at which present fibers have minimum loss (but not minimum dispersion). Optical detectors are typically *pin* diodes or avalanche diodes. For both lasers and detectors, compound semiconductor alloys, such as InGaAsP are widely used. Coherent detection, in which the incoming signal is mixed or compared with a local light source of about the same wavelength, has not yet proved practical in optical systems (although it is the norm in digital radio systems) because of the lack of light sources with sufficient spectral purity and stability. Regenerative repeaters for radio use require modulators and demodulators as well as the functions described above. Figure 17.1.8 shows the block diagram of a regenerator for a 135-Mb/s (actually 3 × 44.736-Mb/s) system using 64 QAM (64 quadrature amplitude modulation). The symbol rate is 22.76 × 10 Mb/s. This regenerator operates on a 70-MHz i.f. signal, and is used on a 30-MHz channel with radio systems in the 6-GHz band, and on a 40-MHz channel with radio systems in the 11-GHz band. In 64 QAM, the transmitted signal has 64 possible states, that is, combinations of amplitude and phase as shown in Fig. 17.1.9. Other modulation methods, such as QPSK (quaternary phase shift keying) and 8PSK, are also in use. In QPSK the transmitted amplitude is constant, but

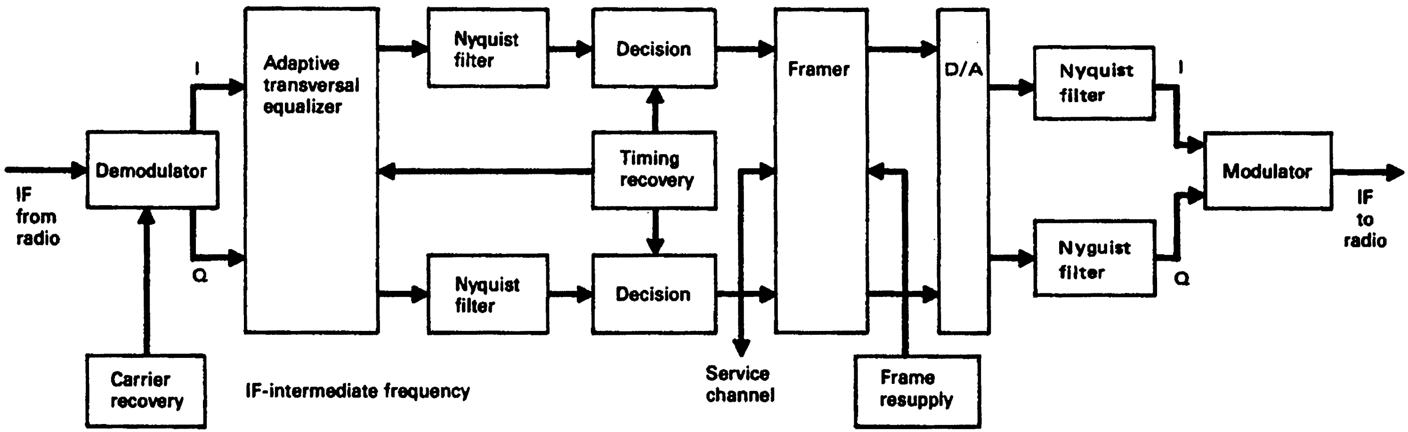


FIGURE 17.1.8 Digital regenerator for 135-Mb/s digital radio system using 64 QAM. (*Copyright 1986, Bell Telephone Laboratories, Inc. Reprinted with permission*)

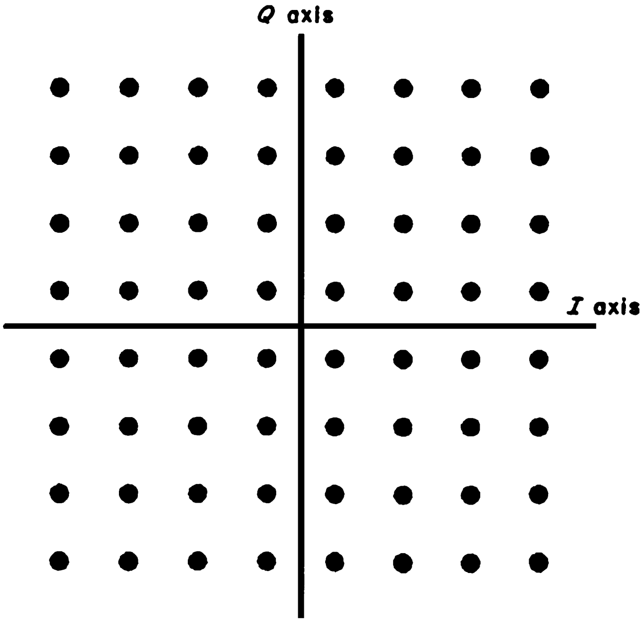


FIGURE 17.1.9 Phase plane representation of possible signals in 64 QAM. (The signals are sinusoids with amplitude given by distance from origin and phase given by angle with *I* axis.) (*From AT&T Bell Laboratories RECORD, January 1986, p. 31. Copyright 1986, Bell Telephone Laboratories, Inc. Reprinted with permission*)

the phase can take on one of four values. In regenerators for radio use, the equalizer is compensating for distortion introduced by propagation effects such as multipath, and it is the development of improved equalizers that has allowed the use of high-capacity modulation such as 64 QAM.

Scramblers and Line Codes for Digital Systems. Normally it is not possible to transmit an arbitrary digital stream directly as a binary signal because of the possibility that some patterns might cause the line system to misfunction. For example, too long a string of zeros could cause loss of timing signal in the repeaters, repetitive patterns could produce strong discrete frequencies in the output of radio systems that might violate emission limits, and a variation of the average density of 1s would cause a wander of the baseline of the signal after passing through a transformer or power-separation filter. To counter these effects, the signal is usually scrambled or coded (sometimes both).

A digital signal is scrambled by adding to it, in mod 2,* a long predetermined pseudo-random pattern of 1s and 0s. It can be unscrambled at the receiving end by adding (mod 2) the same pattern, in the same phase. The pseudo-random pattern is obtained from a shift register clocked at the signal rate with some taps added and fed back to the input. Appropriately selected taps produce a maximal-length pattern $2^N - 1$ long, where N is the number of stages in the register. The phase of the sequence is normally synchronized to the framing pattern, allowing ready descrambling at the receiving end.

Scrambling may reduce the probability (but cannot *guarantee* the absence) of any unfortunate pattern such as a long string of zeros. Therefore, if scrambling is to serve this purpose, high-Q timing recovery circuits such as phase-locked loops are required, and this is typical of fiber optic and radio repeaters. Another use of scrambling is encryption of a digital signal to maintain its confidentiality. This can be accomplished by scrambling

*In mod 2 (modulo 2) addition 0 + 0 = 0, 1 + 0 = 1, 1 + 1 = 0.

the signal with a very long sequence known only to the parties involved. This practice was formerly largely confined to government communications, but heightened commercial interest in privacy has led to a standard method of encrypting, the data encryption standard (DES) promulgated by the National Bureau of Standards, and reasonably priced silicon integrated circuits to implement it. Scrambling for encryption is, of course, best applied at the source and destination of the signal to be encrypted, while scrambling to control signal statistics is applied within a particular transmission system.

Coding, as opposed to scrambling, adds enough redundancy to the signal to guarantee some desired properties, regardless of the pattern to be transmitted. T1 uses a ternary (three-level) code called bipolar (or AMI, for alternate mark inversion) with all zero limitation, used on T1. In this code alternate 1s are transmitted as + and – pulses, assuring dc balance and avoiding baseline wander. Further, an average density of one pulse in eight slots with a maximum of 15 zeros between 1s is required.* This maintains timing signal sufficient for the inexpensive, low-Q (80) timing circuits in the repeaters.

Another scheme to guarantee timing density with a ternary code is to replace strings of zeros with a pattern with two successive pulses of the same polarity, allowing its identification and removal at the receiving end. The arrangement, called BNZS (for bipolar with N zero substitution) is also in considerable use. In Europe, it is called HDB or CHDB, for high-density bipolar or compatible high-density bipolar. BNZS codes all carry one information bit per ternary symbol, and codes with N = 3, 4, and 6 are in use (see Tables 17.1.4 and 17.1.5). Greater information capacity can be obtained using block codes such as 4B3T (each block consists of 4 b on three ternary symbols). Generally, as the information capacity is increased, the reduction in redundancy results in poorer timing and baseline-wander performance. There are may variants, but all have spectra that are zero at both zero frequency and the symbol rate, as do the bipolar and BNZS codes.

TABLE 17.1.4 B3ZS Choice of Sequences to Substitute for Three Zeros

Last pulse transmitted	Last substitute sequence used: 00 + or + 0 +	Last substitute sequence used: 00 – or – 0 –
+	– 0 –	00 +
–	00 –	+ 0 +

Another type of code is epitomized by the *duobinary* code, which is a 1-b-per-symbol ternary code in which adjacent pulses of opposite polarity cannot occur. Thus the redundancy is used to reduce the transmitted energy at high frequencies rather than to control baseline wander and assure timing. Scrambling, high-Q tanks, and circuit design to minimize baseline wander are normally required. The duobinary spectrum is maximum at zero frequency and zero at half the symbol rate. The reduction of high-frequency energy makes it attractive for sharply band-limited media such as radio channels or, on wire pairs, to minimize crosstalk. Variants with larger numbers of transmitted levels have been dubbed *partial-response codes* and can also limit the transmitted spectrum in a similar fashion.

Various ternary codes are illustrated by example in Table 17.1.5. The redundancy added in any of these codes is sufficient to allow reasonably accurate estimates of errors in transmission by detection of violations

TABLE 17.1.5 Coding Examples

Code name	Coding rule	Example
Binary		10111001100000001111
Bipolar, AMI	Invert alternate 1s	+0 – + –00 + –0000000 + – + –
B3ZS, CHDB2	Invert alternate 1s, but for three 0s substituting according to Table 17.1.4	+0 – + –00 + –00 – + 0 + 0 – + – +
4B3T	Code Table	+00 + –00 + –0 – + – – –
Duobinary	From binary input sequence a first form sequence* C	
	$C_n = a'_n \oplus C_{n-1}$ then output =	01111011101010100000
	$C_n + C_{n-1} - 1$	–0 + + + 00 + + 0000000 – – – –

*$\oplus$ = addition modulo 2, and a'_n is the logical inverse of a_n, that is, substitute 0 for 1 and vice versa. The first step avoids a long string of decoding errors resulting from a single transmission error.

*This is readily obtained in voice-band coding by simply not using the all-zero word, and the standard μ-law coder has this feature.

TABLE 17.1.6 CCITT Error Performance Objectives for a 27,500-km Connection at 64 kb/s

Performance classification	Objective
Errored seconds	Fewer than 8% of one-second intervals to have any errors (equivalent to 92% error-free-seconds)
Severely errored seconds	Fewer than 0.2% of one-second intervals to have a bit error ratio worse than 10^{-3}

of the constraints of the particular code. In uncoded systems parity bits may be added to accomplish this purpose. Forward-acting error correcting codes, in which redundancy is added to allow detection and correction of errors at the receiving end, have not been widely used on cable systems. They are, however, widely used on radio and satellite systems, where the capacity is limited by the transmitter power, rather than the bandwidth.

Performance Issues. Digital transmission systems consist of terminals connected by cable or radio links that include regenerative repeaters. Each repeater detects, regenerates, and retimes incoming pulses, so the only impairments introduced are errors and jitter (unwanted phase modulation). Error performance is sometimes characterized in terms of average error rate (or ratio) which is the number of errors divided by the number of bits received, with the implicit assumption that errors are more or less randomly distributed. This has not proved to be a satisfactory measure, as most systems have enough design margin against thermal noise and crosstalk (the normal causes of randomly distributed errors) that the errors on actual systems in service occur in infrequent bursts, as a result of noise impulses or other disturbances. Consequently, the CCITT states error performance objectives for 64 kb/s in terms of error free seconds and severely errored seconds as shown in Table 17.1.6.* Objectives for performance at higher rates are shown in Table 17.1.7. This table is stated in terms of blocks of bits, intended to correspond to block sizes used for error monitoring.

Jitter can be removed by reading the signal into a buffer memory, and reading it out at a constant rate. It is only necessary to organize the network so that the jitter does not exceed the capacity of the buffer memories, and the CCITT has recommended jitter limits to this end.

Maintenance Techniques in Digital Systems. Faults in the basic T1 system are detected by the inability of a channel bank to find framing pulses in its 1.544-Mb/s input signal. When such a condition persists long

TABLE 17.1.7 CCITT Error Performance Objectives for a 27,500-km Path at or Above the Primary Rate

Rate Mb/s	1.5 to 5	>5 to 15	>15 to 55	>55 to 160	>160 to 3500
Bits/block	2000 to 8000	2000 to 8000	4000 to 20,000	6000 to 20,000	15,000 to 30,000
Errored seconds	4%	5%	7.5%	16%	not specified
Severely errored seconds	0.2%	0.2%	0.2%	0.2%	0.2%
Background errored blocks	0.03%	0.02%	0.02%	0.02%	0.01%

An errored block is a block in which 1 or more bits are in error.
An errored second is a 1-s period with one or more errored blocks.
A severely errored second is a 1-s period which contains 30 percent or more errored blocks, or four contiguous blocks each of which has more than 1 percent of its bits in error, or a period of loss of signal.
A background errored block is an errored block not occurring as part of a severely errored second.

*An additional requirement based on a one-minute interval will likely be deleted as redundant in practice.

enough (perhaps 2 s), the channel bank causes the trunks it serves to be declared "busy," lights a red alarm light, and sounds an office alarm. It also transmits a special code that causes the other channel bank to take the trunks out of service, light a yellow alarm light, and sound an alarm. Maintenance personnel then clear the trouble, perhaps by patching in a spare T1 line, and proceed with fault location and repair. Channel banks can be checked by looping, i.e., connecting digital output and input. Repeatered line-fault location techniques were discussed earlier.

In higher-speed systems, automatic line and multiplex protection switching is often provided. A typical line-protection switch monitors and removes violations of the redundancy rules of the line signal on the working line at the receiving (tail) end. When violations in excess of the threshold are detected, a spare line is bridged on at the transmitting (head) end and if a violation-free signal is received on this line, the tail-end switch to spare is completed. If the spare line also has violations, there is probably an upstream failure and no switch is performed. A multiplex-protection switch is typically based on a time-shared monitor that evaluates each of several multiplexers and demultiplexers in turn by pulse-by-pulse comparison of the actual output with correct output based on current input. Multiplex monitors also usually check the incoming high- and low-speed signals using the line-code redundancy.

As the digital network has grown, there has been an increasing use of maintenance centers, to which all alarms in a geographic region are remoted, and which are responsible for dispatching and supervising maintenance personnel, directing restoration of failed facilities over other facilities or routes, and rearrangements for other purposes as well. There is also increasing provision, in network design, of alternative routes, sometimes by routing lines to form a ring, so that there are two physically separate paths between any two offices.

Synchronous Digital Hierarchy and SONET. The existing plesiochronous digital network has grown piecemeal over decades, with the parameters of new systems reflecting the technology and needs at the time of their development. In the late 1980s, a worldwide effort brought forth a new hierarchy for higher rate systems to provide capabilities not possible in the existing network. In this new hierarchy, multiplexing is by interleaving of 8-b bytes, as in the primary rate multiplexes, as opposed to the bit interleaving used elsewhere in the existing network. Further, similar formats are used for multiplexing at all levels, and it is intended that new transmission systems will be at hierarchical levels. Another important feature of the new hierarchy is an overall plan for monitoring and controlling a complex network, and the inclusion of enough *overhead* in the formats to support it. In spite of the name, the new hierarchy allows nonsynchronous signals based on the existing hierarchy to enter, and multiplexing throughout includes enough justification capability to accommodate the small frequency deviations characteristics of reference clocks.

The new hierarchy starts at 51.84 Mb/s, and all higher rates are an integral multiple of this lowest rate. Multiples up to 255 have been envisioned, and structures for several rates have been standardized within the United States. The rates of most interest are shown in Table 17.1.8.

A single frame of the STS-1 signal consists of 810 bytes, as shown in Fig. 17.1.10. The bytes appear on the transmission line read from left to right, starting with the first row. The transported signal occupies 774 bytes of the frame, with the remainder of the frame dedicated to overhead. The transported signal plus the path overhead, the *payload*, are intended to be transported across the network without alteration as the signal is multiplexed to, and recovered from, higher levels. In order to accommodate frequency and phase variations in the network, the 783 bytes of payload can start anywhere within the 783 byte locations allocated to the payload, and continue into the next frame. The starting point is signaled by a *payload pointer* included in the line overhead, so the proper alignment can be recovered at the receiving end. This pointer, as well

TABLE 17.1.8 Major Rates in the Synchronous Digital Hierarchy

Line Rate Mb/s	Designation US (SONET)	Designation CCITT	Comment
51.84	STS-1		Used to carry one 44.736 Mb/s (DS-3) signal
155.52	STS-3	STM-1	Used to carry one 139.254 Mb/s signal
622.08	STS-12	STM-4	Used for fiber optic systems
2488.32	STS-48	STM-16	Used for fiber optic systems

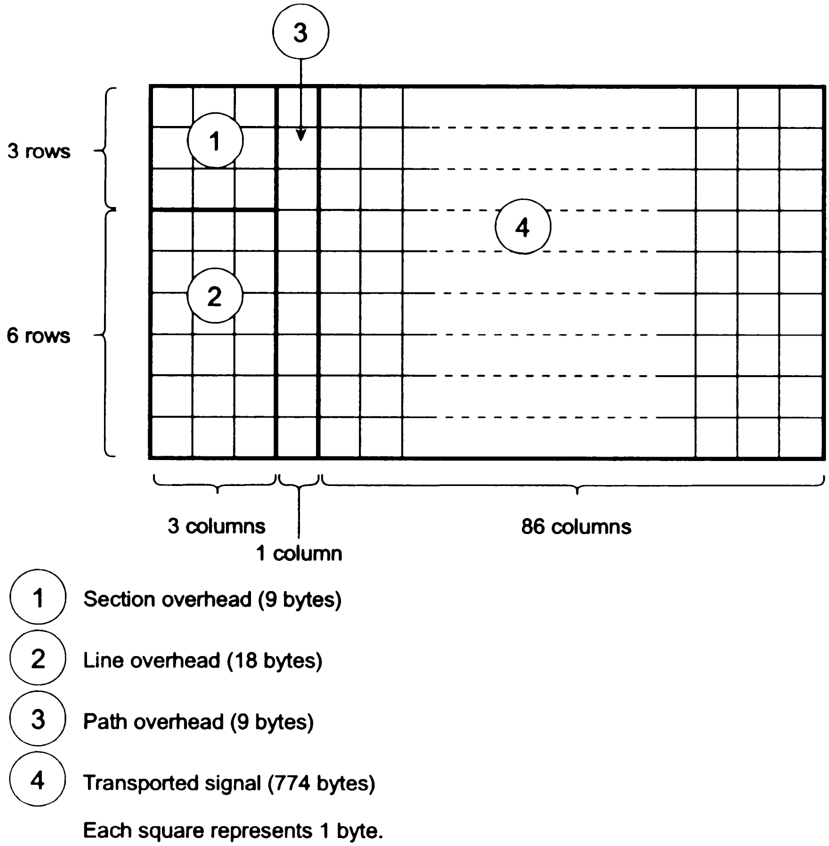


FIGURE 17.1.10 An STS-1 frame: (1) Section overhead (9 bytes); (2) line overhead (18 bytes); (3) path overhead (9 bytes); (4) transported signal (774 bytes). Each square represents 1 byte.

as the remainder of the line and section overhead bytes, are provided for the use of multiplex and line equipment, and will normally be changed several times as the frame passes through the network.

The *path* overhead is placed on the signal at the *path terminating equipment*, where the transported signal is assembled and embedded in the frame. It is not intentionally modified as the STS-1 frame passes through subsequent multiplexes and line systems, but can be read and used at intermediate points. This overhead is, from the point of view of the hierarchy, *end-to-end* information. It contains signals identifying the structure of the frame to aid in retrieving the embedded signal, status of maintenance indications, such as loss of signal for the opposite direction of transmission, a parity check on the previous frame, as well as provision for a message channel for use by the path terminating equipment.

Line overhead information may be inserted or modified when the STS-1 signal is multiplexed to a higher rate, or transferred between higher rate signals. Normally the capability to switch the higher rate signal to a standby facility in case of failure will be provided at such points, so the line overhead includes signaling for coordinating the operation of this protection switching, as well as functionality similar to the path overhead, but for use over the shorter "line."

The *section* overhead includes the framing alignment pattern by which the frame is located, and functionality similar to that of the path overhead, but for use and modification within individual sections, which end at regenerators or multiplexes.

TABLE 17.1.9 Representative North American Digital Systems for Paired Cable

Line rate, Mb/s	64-kb/s voice channel capacity	Widely used designation	Line formats	Usual medium	Typical repeater spacing, mi	Typical section loss, dB
1.544	24	T1 T10S (T1-outstate)	Bipolar with 1-in-8 1s density, 15 0s maximum	Wire pairs in single cable (but two directions in different units)	1 (on 22 gauge)	32
3.152	48	T1C, T1D, T148	Bipolar, 4B3T; duobinary; modified duobinary	As for T1, but also with shielded (screened) units	1 (on 22 gauge)	48
6.443	96	T1G	Quaternary	As for T1C		48

Frames for higher rate signals are generally similar. An STM-N frame has nine rows, and $N \times 270$ columns of which $N \times 3$ are for section overhead. The rather complex structure is based on *virtual containers, tributary units*, and *administrative units*, which are combinations of user signal with the overheads defined above appropriate to the administration of various types of paths.

Line Systems for Transmission

Systems on Wire Cable. Systems providing trunks on wire pair are generally designed to operate on the same cable types used for voice trunks, and to share such cables with voice trunks. Large numbers of such systems that have characteristics indicated in Table 17.1.9 are in service in North America and Japan, although the fiber systems are increasingly being used in new installations. Wire pair systems at 2.048 Mb/s are common in Europe. All the above are four-wire systems, using one pair for each direction of transmission. Two-wire systems providing 144 kb/s in both directions on a single pair have been specified and developed for use as ISDN loops, but little deployed as yet. These two-wire systems use echo cancelers, or time compression multiplexing in which the pair is used alternately in each direction (at about twice the average bit rate) with buffering at the ends.

Systems on Fiber Optic Cable. Fiber-optic systems, operating digitally, and using one fiber for each direction of transmission have developed extremely rapidly since their introduction in 1977, with steadily increasing capacity and, correspondingly, decreased per-channel cost. Systems for trunks and loops have been installed at many of the hierarchical rates (Table 17.1.1), but systems at even higher rates are most prevalent. The characteristics of such systems are summarized in Table 17.1.10, and some specific systems are shown on in Table 17.1.11. A branching unit, including 296 Mb/s regenerative repeaters, used in TAT-8 (Trans ATlantic cable 8) is shown in Fig. 17.1.11. Similar branching units in TAT-9 operate at 591 Mb/s, and include some multiplexing functions as well.

All terrestrial and submarine systems have customarily used intermediate regenerators when the system length requires gain between the terminals. Systems using optical amplifiers instead of regenerators have recently appeared, and the characteristics of one of these is also included in Table 17.1.10. In such systems, erbium doped optical amplifiers are used at intermediate points to overcome loss of the dispersion-shifted fiber with regeneration only at the ends. Figure 7.1.12 shows an amplifier designed for the system shown in the table. Typical output power of such amplifiers is +1 to +3 dBm.

Even these systems do not come close to exploiting the theoretical capacity of the fibers, and further developments are to be expected. Wavelength-division multiplexing, in which two or more transmitter-receiver pairs operate over a single fiber but at different wavelengths, is one way of tapping this capacity, and has seen limited use. The use of solitrons is being explored, and the record as of early 1993 for simulated long-distance transmission in the laboratory, 20 Gb/s over 13,000 km, used this technology.

TABLE 17.1.10 Parameters of Fiber-Optic Systems

Wavelength, nm	Fiber type	Bit rate, Mb/s	Maximum regenerator spacing, km*
850	Graded index	2–140	15–20
1300	Graded index	2–140	45–60
1300	Single mode	140–1700	25–60
1550	Single mode	1.5–2500	50–150

*Lower spacings generally correspond to higher bit rates.

TABLE 17.1.11 Some Fiber-Optic Systems

Name	Primary application	Bit rate per fiber	Wavelength, nm	Repeater spacing, km	Comment
TAT-10	Long undersea routes	591.2 Mb/s	1550	110	
FT-2000	Short and long terrestrial routes	2488.32 Mb/s	1310 or 1550	60 at 1310 nm 84 at 1550 nm	A variety of terminals is available
TAT-12	Long undersea routes	5 Gb/s	1550	33–45	Uses optical amplifiers

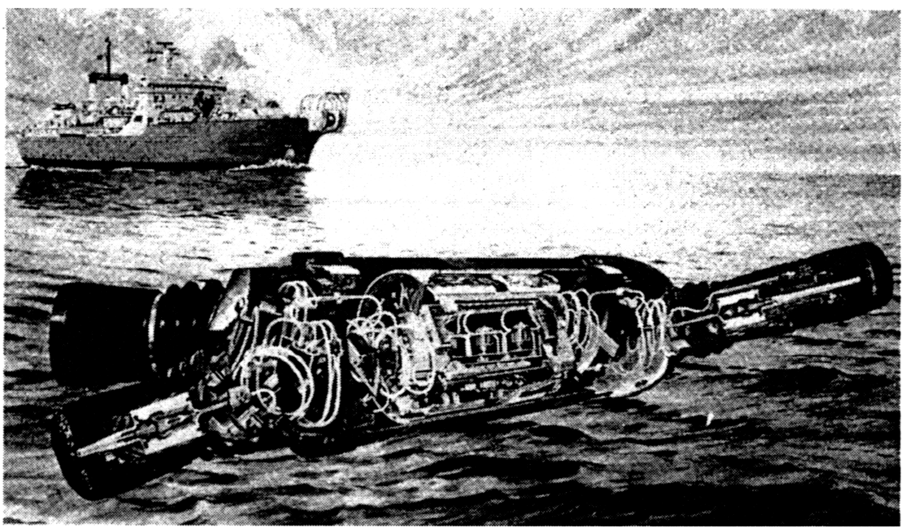


FIGURE 17.1.11 TAT-8 (transatlantic telephone cable no. 8) branching repeater with cable-laying ship in background. (*From AT&T. Used with permission*)

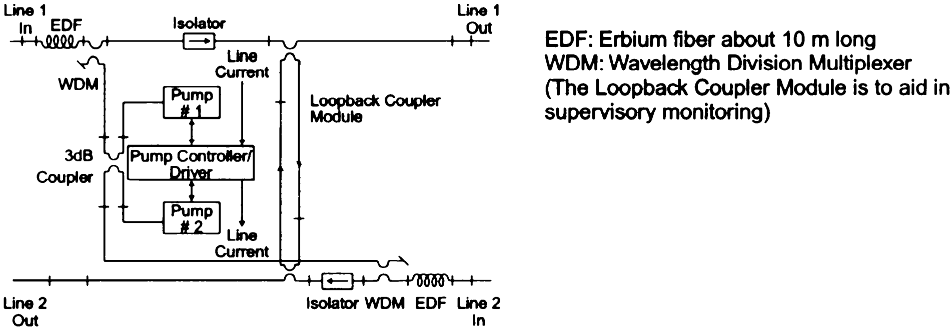


FIGURE 17.1.12 Amplifier pair for TAT-12 (AT&T section).

Terrestrial Radio. Frequencies allocated to telecommunications in the United States are shown in Table 17.1.2. Typical systems for analog signals modulate a carrier using low index FM with a signal consisting of one or more multiplexed mastergroups, and occupy a bandwidth of two or more times 4 kHz for each voice channel. Systems with very linear amplifiers have also used single-sideband AM, with a resulting bandwidth of closet to 4 kHz per voice channel.

While analog microwave radio once carried the bulk of long-haul telecommunications in the United States and many other countries, it has been mostly displaced in long-haul applications by optical fiber, particularly in the United States, and in short-haul applications by digital radio, owing to the need to interconnect with digital switches and other digital transmission systems.

A block diagram of a digital radio regenerator is shown in Fig. 17.1.8. The regenerator operates at i.f., and a complete station involves frequency conversion to and from rf, as well as receiving and transmitting antennas. An end station uses the transmitting and receiving portions of the regenerator separately to create and demodulate the transmitted and received signals.

Communications Satellite Systems. While the first experimental communications satellites were in low earth orbit, commercial satellites have almost uniformly been in a geostationary orbit, 22,300 miles above the equator. (The exceptions were in polar orbits, for better visibility from the northern polar regions.) In such an orbit, the satellite appears stationary from the earth, so a permanent communication link can be established using a single satellite, and earth stations with very directive stationary (or almost stationary) antennas. The disadvantages of this orbit are the high loss and delay resulting from the long distance the signal must travel. Table 17.1.12 lists representative applications of communication satellites, including current proposals for new low earth orbit systems.

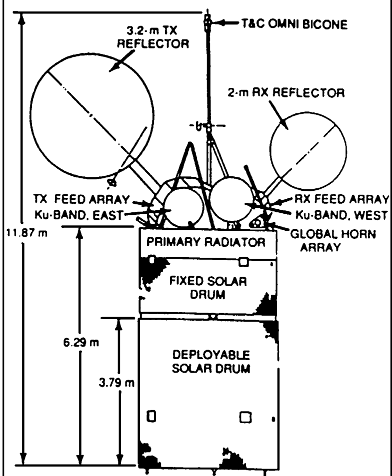
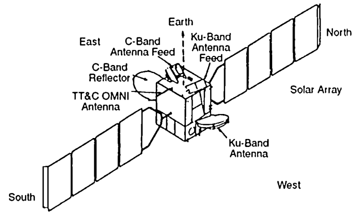
Communications satellites receive signals from an earth station, and include *transponders*, which amplify and translate the signal in frequency and retransmit it to the receiving earth station, thus making effective use of the line-of-sight microwave bands without requiring erection of relay towers. The transponders are powered from solar cells, with batteries for periods of eclipse. *Spin-stabilized* satellites are roughly cylindrical and spin at about 60 r/min, except for a "despun" portion, including the antennas, that is pointed at the earth. *Three-axis-stabilized* satellites have internal high-speed rotating wheels for stability, and solar cells on appendages which unfold after they are in orbit. Adjustments in the position and orientation of a satellite in orbit are accomplished under control of the telemetry tracking and control (TTC) station on the earth, and the exhaustion of fuel for this purpose is the normal cause of end of life of the satellite, typically 10 to 12 years. (At end of life, the TTC moves the satellites to an unused portion of the orbit, where it remains, an archeological resource for future generations.) High reliability is necessary, and on-board spares for the electronics, switchable from the TTC, at a ratio of 50 to 100 percent, are typically provided. Table 17.1.13 gives the characteristics of two current satellites.

Most civilian communication satellites have used the common-carrier bands of 5925 to 6425 MHz in the uplinks, and the 3700 to 4200-MHz band in the downlinks. Now, direct broadcast satellites (DBS) use the 11- and 14-GHz bands for down- and uplinks, respectively. Since these bands are not so widely used in terrestrial

TABLE 17.1.12 Representative Applications of Communication Satellites

Application	Type of Service	Technical Characteristics	Status
Intercontinental telephone trunking	Point-to-point, 2-way	Earth station antennas to 30 m. FDMA, TDMA, and FM with a single channel per transponder used	The Intelsat system. Widely used where fiber optic cables are not available, also for handling peak loads on cables, and during repair of failed cables
Intercontinental TV transmission	Point-to-point, 1-way	Analog TV signals using FM with a single channel per transponder	Carried along with voice on the Intelsat satellites. Primary way of providing this service
National telephone trunks	Point-to-point, 2-way	Wide variety of antenna sizes, access methods have been used	No longer used in the United States, primarily because voice users don't like delay. Still used in countries with difficult terrain, long distances between population centers, or sparse networks
Distribution of TV signals to local broadcast stations or CATV distribution centers	Point-to-multipoint, 1-way	Smaller receiving antennas. Analog TV signals using FM with a single channel per transponder	Major provider of this service. Economics generally favorable compared to cable and microwave radio
Business and educational TV distribution, typically directly to viewing site	Point-to-multipoint, 1-way	Originally analog TV using FM, but increasingly digital, using coders, which, by removing redundancy encode the signal into 6 Mb/s or less, allowing multiple channels per transponder	Major provider of this comparatively new service
Data links, international and domestic	Point-to-point, 2-way	Low rate data channels can be multiplexed to a high rate to fill a transponder, or FDMA or TDMA can be used	Has seen considerable use, as with proper protocols, delay is not a problem in most applications. Fiber optic cables are eroding market
Maritime Mobile telephone	Fixed-point to mobile, 2-way	Operates at 1.5 GHz, with geosynchronous satellite	Via the INMARSAT system, the major modality for ship-to-shore telephony
Paging, short message	Fixed-point to mobile, 1-way	Would operate at 150 MHz with a total bandwidth of 1 MHz,using low-earth-orbit satellite	Proposal. Intent is to provide paging and limited message capability to personal receivers
Terrestrial mobile	Fixed-point to mobile, 2-way	Would operate at about 1.5 GHz, with a total bandwidth of about 20 MHz. Would use from 12 to 30 satellites in low earth orbit, using circular polarization and low directivity antennas on the mobile stations	Proposal. Intent is to provide mobile service roughly comparable to cellular, but available without the necessity for local terrestrial construction and network access

TABLE 17.1.13 Representative Communications Satellites

Satellite	Intelsat VI	Telstar 4
		
Type	Spin stabilized	Three-axis stabilized
Mass	4000 kg	4212 lb
Size	11.6 m high, 3.6 m diameter	Body 7.25 × 8.33 × 13.4 ft, extended length 80.4 ft
First launch	1989	1993
Launch vehicle	Ariane 4, Titan III	Atlas Centaur II AS
4/6 GHz Transponders and bandwidths	12 @ 36 MHz, 2 @ 41 MHz, 26 @ 72 MHz	24 @ 6 MHz
11/14 GHz Transponders and bandwidths	1 @ 36 MHz, 6 @ 72 MHz, 2 @ 77 MHz, 2 @ 150 MHz	8 @ 54 MHz, 16 @ 27 MHz (Can be used in pairs as 54 MHz also)
Primary power, W	2204	3744
Main applications	International telephony, TV, data	US domestic TV, data
Transponder output power, W	1.3–16 @ 4 GHz, 10 @ 11 GHz	12 @ 4 GHz, 60 @ 14 GHz (Can be doubled on some transponders)
Other features	SSTDMA (6×6 switch at microwave frequencies)	Can switch to a smaller receive spot beam to reduce and locate interference Compensates for uplink rain attenuation by increasing transponder gain

microwave relay, interference problems are less although rain attenuation is much higher. All frequencies are subject to sun-transit outage when the satellite is directly between the sun and the receiving earth station, so that the receiving antenna is pointing directly at the noisy sun. This occurs for periods of up to $^1/_2$ h/d for several days around the equinoxes. The effect can be avoided by switching to another distant satellite during the sun-transit period. The propagation delay between earth stations is about 0.25 s in each direction for geostationary

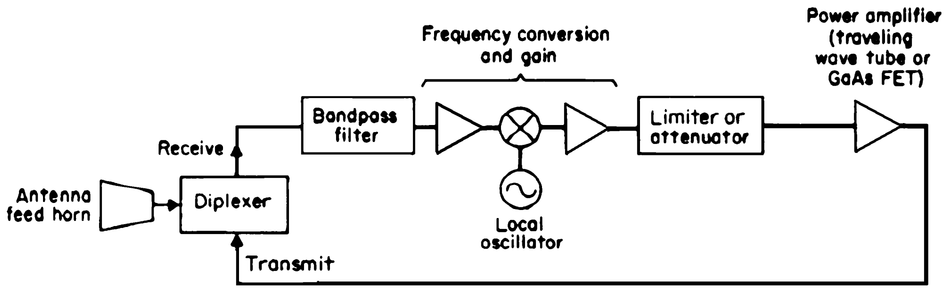


FIGURE 17.1.13 Satellite transponder.

satellites. This delay is of no consequence for one-way television transmission, but is disturbing to telephone users.

Some satellites use each frequency band twice, once in each polarization. Earth antenna directivity permits reuse of the same frequencies by different satellites as long as satellites are not too close in orbit (2° is the limit for domestic U.S. satellites in the 4- to 6-GHz band). Further frequency reuse is possible in a single satellite by using more directive satellite antennas which direct separate beams to different earth areas. Although Intelsat has made use of spot beams, most present satellite antennas have beam widths covering upward of 1000 mi on earth. A simplified transponder block diagram is given in Fig. 17.1.13.

Transponder utilization and multiple access. A transponder can be used for a single signal (single carrier operation) which may be either frequency- or time-division-multiplexed. Such signals have included a single TV signal, two or three 600-channel analog master groups multiplexed together and used to frequency modulate a carrier, thirteen 600-channel master groups using compounders and single-sideband amplitude modulation, and a digital signal with rates up to 14.0 Mb/s used to modulate a carrier using quaternary phase-shift keying (QPSK), or coded octal phase-shift keying. Single-carrier operation can be either point-to-point (as for normal telecommunication) or broadcast (as for distributing TV programs). Transponders can also be used in either of two multiple-access modes in which the same transponder carries (simultaneously) signals from several different earth stations.

In frequency-division multiplex access (FDMA) the frequency band of each transponder is subdivided and portions assigned to different earth stations. Each station can then transmit continuously in its assigned frequency band without interfering with the other signals. All earth stations receive all signals but demodulate only signals directed to that station. In the limit of subdivision, one voice channel can be placed on a single carrier (single channel per carrier or SCPC). As the high-power amplifiers (HPA) in the earth station and the satellite are highly nonlinear, power levels must be reduced considerably ("backed off") below the saturation level to reduce intermodulation distortion between the several carriers. It is also possible to use *demand assignment* in which a given frequency slot can be reassigned among several earth stations as traffic demands change.

In TDMA (time-division multiple access) each earth station uses the entire bandwidth of a transponder for a portion of the time, as illustrated in Fig. 17.1.14. This arrangement implies digital transmission (such as QPSK) with buffer memories at the earth stations to form the bursts. A synchronization arrangement that controls the time of transmission of each station is also required. As at any given time only a single carrier is involved, less backoff is required than with FDMA, allowing an improved signal-to-noise ratio. Demand assignment can be realized by reassigning burst times among the stations in the network. Satellite-switched TDMA (SSTDMA) in which a switch in the satellite routes bursts among spot beams covering different terrestrial areas is a feature of Intelsat VI.

Transmission considerations. The free-space loss between a geostationary satellite and the earth is about 200 dB. To overcome this large loss, earth stations for telecommunications trunks have traditionally used large parabolic antennas (10 to 30 m in diameter), high output power (up to several kilowatts), and low-noise receiving amplifiers (cryogenically cooled in some cases). Transponder output power is limited to the power available from the solar cells, and therefore, downlink thermal noise often accounts for most of the system noise with intermodulation in the transponder power amplifier a significant limiting factor. Consequently, the capacity of

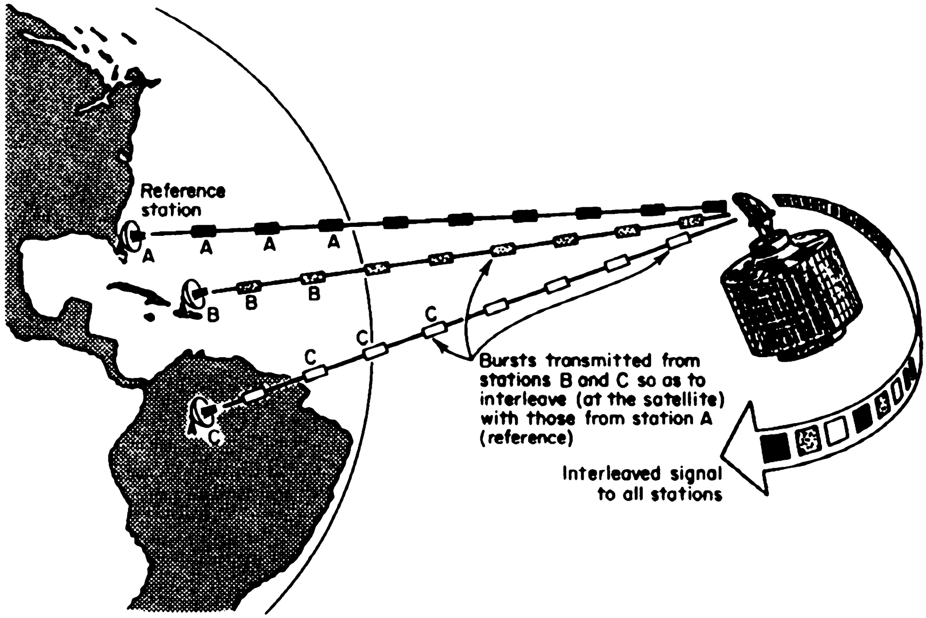


FIGURE 17.1.14 Satellite time-division multiplex access (TDMA). From Digital Communications Corporation; used by permission.

a satellite channel is often limited by the received signal-to-noise ratio (power-limited) rather than by the bandwidth of the channel.

For applications other than high-capacity trunking, the cost of large antennas at the earth stations is often prohibitive, so lower capacity is accepted, and received power may be increased by dedicating more power to the transponder or by use of spot beams, as the economics of the application dictate. Smaller antennas are less directive, possibly causing interference to adjacent satellites, unless the station is receive-only. Therefore VSATs (very small aperture terminals), which may have antennas as small as 1 m, typically operate in the higher frequency bands where directivity of smaller antennas may be adequate. Some applications, including the proposals included in Table 17.1.12, use much lower frequencies, and accept, or even exploit, the lesser directivity.