telecommunication

telecommunication, science and practice of transmitting information by electromagnetic means. Modern telecommunication centres on the problems involved in transmitting large volumes of information over long distances without damaging loss due to noise and interference. The basic components of a modern digital telecommunications system must be capable of transmitting voice, data, radio, and television signals. Digital transmission is employed in order to achieve high reliability and because the cost of digital switching systems is much lower than the cost of analog systems. In order to use digital transmission, however, the analog signals that make up most voice, radio, and television communication must be subjected to a process of analog-to-digital conversion. (In data transmission this step is bypassed because the signals are already in digital form; most television, radio, and voice communication, however, use the analog system and must be digitized.) In many cases, the digitized signal is passed through a source encoder, which employs a number of formulas to reduce redundant binary information. After source encoding, the digitized signal is processed in a channel encoder, which introduces redundant information that allows errors to be detected and corrected. The encoded signal is made suitable for transmission by modulation onto a carrier wave and may be made part of a larger signal in a process known as multiplexing. The multiplexed signal is then sent into a multiple-access transmission channel. After transmission, the above process is reversed at the receiving end, and the information is extracted.

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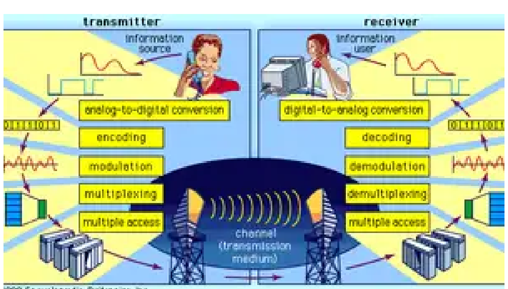

Block diagram of a digital telecommunications system.

This article describes the components of a digital telecommunications system as outlined above. For details on specific applications that utilize telecommunications systems, see the articles telephone, telegraph, fax, radio, and television. Transmission over electric wire, radio wave, and optical fibre is discussed in telecommunications media. For an overview of the types

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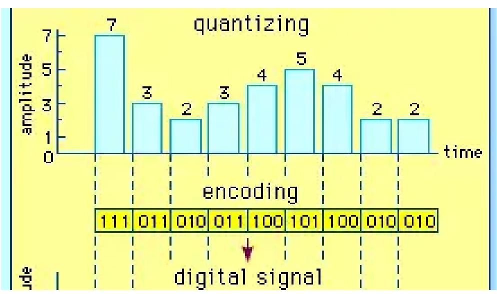
of networks used in information transmission, see telecommunications network.

Analog-to-digital conversion

In transmission of speech, audio, or video information, the object is high fidelity—that is, the best possible reproduction of the original message without the degradations imposed by signal distortion and noise. The basis of relatively noise-free and distortion-free telecommunication is the binary signal. The simplest possible signal of any kind that can be employed to transmit messages, the binary signal consists of only two possible values. These values are represented by the binary digits, or bits, 1 and 0. Unless the noise and distortion picked up during transmission are great enough to change the binary signal from one value to another, the correct value can be determined by the receiver so that perfect reception can occur.



Basic steps in analog-to-digital conversionAn analog signal is sampled at regular intervals. The amplitude at each interval is quantized, or assigned a value, and the values are mapped into a series of binary digits, or bits. The information is transmitted as a digital signal to the receiver, where it is decoded and the analog signal reconstituted.
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If the information to be transmitted is already in binary form (as in data communication), there is no need for the signal to be digitally encoded. But ordinary voice communications taking place by way of a telephone are not in binary form; neither is much of the information gathered for transmission from a space probe, nor are the television or radio signals gathered for transmission through a satellite link. Such signals, which continually vary among a range of values, are said to be analog, and in digital communications systems analog signals must be converted to digital form. The process of making this signal conversion is called analog-to-digital (A/D) conversion.

Sampling

Analog-to-digital conversion begins with sampling, or measuring the amplitude of the analog waveform at equally spaced discrete instants of time. The fact that samples of a continually varying wave may be used to represent that wave relies on the assumption that the wave is constrained in its rate of variation. Because a communications signal is actually a complex wave—essentially the sum of a number of component sine waves, all of which have their own precise amplitudes and phases—the rate of variation of the complex wave can be measured by the frequencies of oscillation of all its components. The difference between the maximum rate of oscillation (or highest frequency) and the minimum rate of oscillation (or lowest frequency) of the sine waves making up the signal is known as the bandwidth (*B*) of the signal. Bandwidth thus represents the maximum frequency range occupied by a signal. In the case of a voice signal having a minimum frequency of 300 hertz and a maximum frequency of 3,300 hertz, the bandwidth is 3,000 hertz, or 3 kilohertz. Audio signals generally occupy about 20 kilohertz of bandwidth, and standard video signals occupy approximately 6 million hertz, or 6 megahertz.

The concept of bandwidth is central to all telecommunication. In analog-to-digital conversion, there is a fundamental theorem that the analog signal may be uniquely represented by discrete samples spaced no more than one over twice the bandwidth ($1/2B$) apart. This theorem is commonly referred to as the sampling theorem, and the sampling interval ($1/2B$ seconds) is referred to as the Nyquist interval (after the Swedish-born American electrical engineer Harry Nyquist). As an example of the Nyquist interval, in past telephone practice the bandwidth, commonly fixed at 3,000 hertz, was sampled at least every 1/6,000 second. In current practice 8,000 samples are taken per second, in order to increase the frequency range and the fidelity of the speech representation.

Quantization

In order for a sampled signal to be stored or transmitted in digital form, each sampled amplitude must be converted to one of a finite number of possible values, or levels. For ease in conversion to binary form, the number of levels is usually a power of 2—that is, 8, 16, 32, 64, 128, 256, and so on, depending on the degree of precision required. In digital transmission of voice, 256 levels are commonly used because tests have shown that this provides adequate fidelity for the average telephone listener.

The input to the quantizer is a sequence of sampled amplitudes for which there are an infinite number of possible values. The output of the quantizer, on the other hand, must be restricted to a finite number of levels. Assigning infinitely variable amplitudes to a limited number of levels inevitably introduces inaccuracy, and inaccuracy results in a corresponding amount of signal distortion. (For this reason quantization is often called a “lossy” system.) The degree of inaccuracy depends on the number of output levels used by the quantizer. More quantization levels increase the accuracy of the representation, but they also increase the storage capacity or transmission speed required. Better performance with the same number of output levels can be achieved by judicious placement of the output levels and the amplitude thresholds needed for assigning those levels. This placement in turn depends on the nature of the waveform that is being quantized. Generally, an optimal quantizer places more levels in amplitude ranges where the signal is more likely to occur and fewer levels where the signal is less likely. This technique is known as nonlinear quantization. Nonlinear quantization can also be accomplished by passing the signal through a compressor circuit, which amplifies the signal's weak components and attenuates its strong components. The compressed signal, now occupying a narrower dynamic range, can be quantized with a uniform, or linear, spacing of thresholds and output levels. In the case of the telephone signal, the compressed signal is uniformly quantized at 256 levels, each level being represented by a sequence of eight bits. At the receiving end, the reconstituted signal is expanded to its original range of amplitudes. This sequence of compression and expansion, known as companding, can yield an effective dynamic range equivalent to 13 bits.

Bit mapping

In the next step in the digitization process, the output of the quantizer is mapped into a binary sequence. An encoding table that might be used to generate the binary sequence is shown below:

quantization level	binary code
0	000
1	001
2	010
3	011
4	100
5	101
6	110
7	111

It is apparent that 8 levels require three binary digits, or bits; 16 levels require four bits; and 256 levels require eight bits. In general 2^n levels require *n* bits.

In the case of 256-level voice quantization, where each level is represented by a sequence of 8 bits, the overall rate of transmission is 8,000 samples per second times 8 bits per sample, or 64,000 bits per second. All 8 bits must be transmitted before the next sample appears. In order to use more levels, more binary samples would have to be squeezed into the allotted time slot between successive signal samples. The circuitry would become more costly, and the bandwidth of the system would become correspondingly greater. Some transmission channels (telephone wires are one example) may not have the bandwidth capability required for the increased number of binary samples and would distort the digital signals. Thus, although the accuracy required determines the number of quantization levels used, the resultant binary sequence must still be transmitted within the bandwidth tolerance allowed.

Source encoding

As is pointed out in analog-to-digital conversion, any available telecommunications medium has a limited capacity for data transmission. This capacity is commonly measured by the parameter called bandwidth. Since the bandwidth of a signal increases with the number of bits to be transmitted each second, an important function of a digital communications system is to represent the digitized signal by as few bits as possible—that is, to reduce redundancy. Redundancy reduction is accomplished by a source encoder, which often operates in conjunction with the analog-to-digital converter.

Huffman codes

In general, fewer bits on the average will be needed if the source encoder takes into account the probabilities at which different quantization levels are likely to occur. A simple example will illustrate this concept. Assume a quantizing scale of only four levels: 1, 2, 3, and 4. Following the usual standard of binary encoding, each of the four levels would be mapped by a two-bit code word. But also assume that level 1 occurs 50 percent of the time, that level 2 occurs 25 percent of the time, and that levels 3 and 4 each occur 12.5 percent of the time. Using variable-bit code words, such as those also shown in the table, might cause more efficient mapping of these levels to be achieved. The variable-bit encoding rule would use only one bit 50 percent of the time, two bits 25 percent of the time, and three bits 25 percent of the time. On average it would use 1.75 bits per sample rather than the 2 bits per sample used in the standard code.

Thus, for any given set of levels and associated probabilities, there is an optimal encoding rule that minimizes the number of bits needed to represent the source. This encoding rule is known as the Huffman code, after the American D.A. Huffman, who created it in 1952. Even more efficient encoding is possible by grouping sequences of levels together and applying the Huffman code to these sequences.

The Lempel-Ziv algorithm

The design and performance of the Huffman code depends on the designers' knowing the probabilities of different levels and sequences of levels. In many cases, however, it is desirable to have an encoding system that can adapt to the unknown probabilities of a source. A very efficient technique for encoding sources without needing to know their probable occurrence was developed in the 1970s by the Israelis Abraham Lempel and Jacob Ziv. The Lempel-Ziv algorithm works by constructing a codebook out of sequences encountered previously. For example, the codebook might begin with a set of four 12-bit code words representing four possible signal levels. If two of those levels arrived in sequence, the encoder, rather than transmitting two full code words (of length 24), would transmit the code word for the first level (12 bits) and then an extra two bits to indicate the second level. The encoder would then construct a new code word of 12 bits for the sequence of two levels, so that even fewer bits would be used thereafter to represent that particular combination of levels. The encoder would continue to read quantization levels until another sequence arrived for which there was no code word. In this case the sequence without the last level would be in the codebook, but not the whole sequence of levels. Again, the encoder would transmit the code word for the initial sequence of levels and then an extra two bits for the last level. The process would continue until all 4,096 possible 12-bit combinations had been assigned as code words.

In practice, standard algorithms for compressing binary files use code words of 12 bits and transmit 1 extra bit to indicate a new sequence. Using such a code, the Lempel-Ziv algorithm can compress transmissions of English text by about 55 percent, whereas the Huffman code compresses the transmission by only 43 percent.

Run-length codes

Certain signal sources are known to produce "runs," or long sequences of only 1s or 0s. In these cases it is more efficient to transmit a code for the length of the run rather than all the bits that represent the run itself. One source of long runs is the fax machine. A fax machine works by scanning a document and mapping very small areas of the document into either a black pixel (picture element) or a white pixel. The document is divided into a number of lines (approximately 100 per inch), with 1,728 pixels in each line (at standard resolution). If all black pixels were mapped into 1s and all white pixels into 0s, then the scanned document would be represented by 1,857,600 bits (for a standard American 11-inch page). At older modem transmission speeds of 4,800 bits per second, it would take 6 minutes 27 seconds to send a single page. If, however, the sequence of 0s and 1s were compressed using a run-length code, significant reductions in transmission time would be made.

The code for fax machines is actually a combination of a run-length code and a Huffman code; it can be explained as follows: A run-length code maps run lengths into code words, and the codebook is partitioned into two parts. The first part contains symbols for runs of lengths that are a multiple of 64; the second part is made up of runs from 0 to 63 pixels. Any run length would then be represented as a multiple of 64 plus some remainder. For example, a run of 205 pixels would be sent using the code word for a run of length 192 (3 × 64) plus the code word for a run of length 13. In this way the number of bits needed to represent the run is decreased significantly. In addition, certain runs that are known to have a higher probability of occurrence are encoded into code words of short length, further reducing the number of bits that need to be transmitted. Using this type of encoding, typical compressions for facsimile transmission range between 4 to 1 and 8 to 1. Coupled to higher modem speeds, these compressions reduce the transmission time of a single page to between 48 seconds and 1 minute 37 seconds.

Channel encoding

As described in Source encoding, one purpose of the source encoder is to eliminate redundant binary digits from the digitized signal. The strategy of the channel encoder, on the other hand, is to add redundancy to the transmitted signal—in this case so that errors caused by noise during transmission can be corrected at the receiver. The process of encoding for protection against channel errors is called error-control coding. Error-control codes are used in a variety of applications, including satellite communication, deep-space communication, mobile radio communication, and computer networking.

There are two commonly employed methods for protecting electronically transmitted information from errors. One method is called forward error control (FEC). In this method information bits are protected against errors by the transmitting of extra redundant bits, so that if errors occur during transmission the redundant bits can be used by the decoder to determine where the errors have occurred and how to correct them. The second method of error control is called automatic repeat request (ARQ). In this method redundant bits are added to the transmitted information and are used by the receiver to detect errors. The receiver then signals a request for a repeat transmission. Generally, the number of extra bits needed simply to detect an error, as in the ARQ system, is much smaller than the number of redundant bits needed both to detect and to correct an error, as in the FEC system.

Repetition codes

One simple, but not usually implemented, FEC method is to send each data bit three times. The receiver examines the three transmissions and decides by majority vote whether a 0 or 1 represents a sample of the original signal. In this coded system, called a repetition code of block-length three and rate one-third, three times as many bits per second are used to transmit the same signal as are used by an uncoded system; hence, for a fixed available bandwidth only one-third as many signals can be conveyed with the coded system as compared with the uncoded system. The gain is that now at least two of the three coded bits must be in error before a reception error occurs.

The Hamming code

Another simple example of an FEC code is known as the Hamming code. This code is able to protect a four-bit information signal from a single error on the channel by adding three redundant bits to the signal. Each sequence of seven bits (four information bits plus three redundant bits) is called a code word. The first redundant bit is chosen so that the sum of ones in the first three information bits plus the first redundant bit amounts to an even number. (This calculation is called a parity check, and the redundant bit is called a parity bit.) The second parity bit is chosen so that the sum of the ones in the last three information bits plus the second parity bit is even, and the third parity bit is chosen so that the sum of ones in the first, second, and fourth information bits and the last parity bit is even. This code can correct a single channel error by recomputing the parity checks. A parity check that fails indicates an error in one of the positions checked, and the two subsequent parity checks, by process of elimination, determine the precise location of the error. The Hamming code thus can correct any single error that occurs in any of the seven positions. If a double error occurs, however, the decoder will choose the wrong code word.

Convolutional encoding

The Hamming code is called a block code because information is blocked into bit sequences of finite length to which a number of redundant bits are added. When k information bits are provided to a block encoder, $n - k$ redundancy bits are appended to the information bits to form a transmitted code word of *n* bits. The entire code word of length *n* is thus completely determined by one block of *k* information bits. In another channel-encoding scheme, known as convolutional encoding, the encoder output is not naturally segmented into blocks but is instead an unending stream of bits. In convolutional encoding, memory is incorporated into the encoding process, so that the preceding *M* blocks of *k* information bits, together with the current block of *k* information bits, determine the encoder output. The encoder accomplishes this by shifting among a finite number of "states," or "nodes." There are several variations of convolutional encoding, but the simplest example may be seen in what is known as the (*n*,1) encoder, in which the current block of *k* information bits consists of only one bit. At each given state of the (*n*,1) encoder, when the information bit (a 0 or a 1) is received, the encoder transmits a sequence of *n* bits assigned to represent that bit when the encoder is at that current state. At the same time, the encoder shifts to one of only two possible successor states, depending on whether the information bit was a 0 or a 1. At this successor state, in turn, the next information bit is represented by a specific sequence of *n* bits, and the encoder is again shifted to one of two possible successor states. In this way, the sequence of information bits stored in the encoder's memory determines both the state of the encoder and its output, which is modulated and transmitted across the channel. At the receiver, the demodulated bit sequence is compared to the possible bit sequences that can be produced by the encoder. The receiver determines the bit sequence that is most likely to have been transmitted, often by using an efficient decoding algorithm called Viterbi decoding (after its inventor, A.J. Viterbi). In general, the greater the memory (i.e., the more states) used by the encoder, the better the error-correcting performance of the code—but only at the cost of a more complex decoding algorithm. In addition, the larger the number of bits (*n*) used to transmit information, the better the performance—at the cost of a decreased data rate or larger bandwidth.

Coding and decoding processes similar to those described above are employed in trellis coding, a coding scheme used in high-speed modems. However, instead of the sequence of bits that is produced by a convolutional encoder, a trellis encoder produces a sequence of modulation symbols. At the transmitter, the channel-encoding process is coupled with the modulation process, producing a system known as trellis-coded modulation. At the receiver, decoding and demodulating are performed jointly in order to optimize the performance of the error-correcting algorithm.

Modulation

In many telecommunications systems, it is necessary to represent an information-bearing signal with a waveform that can pass accurately through a transmission medium. This assigning of a suitable waveform is accomplished by modulation, which is the process by which some characteristic of a carrier wave is varied in accordance with an information signal, or modulating wave. The modulated signal is then transmitted over a channel, after which the original information-bearing signal is recovered through a process of demodulation.

Modulation is applied to information signals for a number of reasons, some of which are outlined below.

1. Many transmission channels are characterized by limited passbands—that is, they will pass only certain ranges of frequencies without seriously attenuating them (reducing their amplitude). Modulation methods must therefore be applied to the information signals in order to "frequency translate" the signals into the range of frequencies that are permitted by the channel. Examples of channels that exhibit passband characteristics include alternating-current-coupled coaxial cables, which pass signals only in the range of 60 kilohertz to several hundred megahertz, and fibre-optic cables, which pass light signals only within a given wavelength range without significant attenuation. In these instances frequency translation is used to "fit" the information signal to the communications channel.
2. In many instances a communications channel is shared by multiple users. In order to prevent mutual interference, each user's information signal is modulated onto an assigned carrier of a specific frequency. When the frequency assignment and subsequent combining is done at a central point, the resulting combination is a frequency-division multiplexed signal, as is discussed in Multiplexing. Frequently there is no central combining point, and the communications channel itself acts as a distributed combine. An example of the latter situation is the broadcast radio bands (from 540 kilohertz to 600 megahertz), which permit simultaneous transmission of multiple AM radio, FM radio, and television signals without mutual interference as long as each signal is assigned to a different frequency band.
3. Even when the communications channel can support direct transmission of the information-bearing signal, there are often practical reasons why this is undesirable. A simple example is the transmission of a three-kilohertz (i.e., voiceband) signal via radio wave. In free space the wavelength of a three-kilohertz signal is 100 kilometres (60 miles). Since an effective radio antenna is typically as large as half the wavelength of the signal, a three-kilohertz radio wave might require an antenna up to 50 kilometres in length. In this case translation of the voice frequency to a higher frequency would allow the use of a much smaller antenna.

Analog modulation

As is noted in analog-to-digital conversion, voice signals, as well as audio and video signals, are inherently analog in form. In most modern systems these signals are digitized prior to transmission, but in some systems the analog signals are still transmitted directly without converting them to digital form. There are two commonly used methods of modulating analog signals. One technique, called amplitude modulation, varies the amplitude of a fixed-frequency carrier wave in proportion to the information signal. The other technique, called frequency modulation, varies the frequency of a fixed-amplitude carrier wave in proportion to the information signal.

Digital modulation

In order to transmit computer data and other digitized information over a communications channel, an analog carrier wave can be modulated to reflect the binary nature of the digital baseband signal. The parameters of the carrier that can be modified are the amplitude, the frequency, and the phase.

Three methods of digital signal modulationA digital signal, representing the binary digits 0 and 1 by a series of on and off amplitudes, is impressed onto an analog carrier wave of constant amplitude and frequency. In amplitude-shift keying (ASK), the modulated wave represents the series of bits by shifting abruptly between high and low amplitude. In frequency-shift keying (FSK), the bit stream is represented by shifts between two frequencies. In phase-shift keying (PSK), amplitude and frequency remain constant; the bit stream is represented by shifts in the phase of the modulated signal. *Encyclopædia Britannica, Inc.*

Amplitude-shift keying

If amplitude is the only parameter of the carrier wave to be altered by the information signal, the modulating method is called amplitude-shift keying (ASK). ASK can be considered a digital version of analog amplitude modulation. In its simplest form, a burst of radio frequency is transmitted only when a binary 1 appears and is stopped when a 0 appears. In another variation, the 0 and 1 are represented in the modulated signal by a shift between two preselected amplitudes.

Frequency-shift keying

If frequency is the parameter chosen to be a function of the information signal, the modulation method is called frequency-shift keying (FSK). In the simplest form of FSK signaling, digital data is transmitted using one of two frequencies, whereby one frequency is used to transmit a 1 and the other frequency to transmit a 0. Such a scheme was used in the Bell 103 voiceband modem, introduced in 1962, to transmit information at rates up to 300 bits per second over the public switched telephone network. In the Bell 103 modem, frequencies of 1,080 +/- 100 hertz and 1,750 +/- 100 hertz were used to send binary data in both directions.

Phase-shift keying

When phase is the parameter altered by the information signal, the method is called phase-shift keying (PSK). In the simplest form of PSK a single radio frequency carrier is sent with a fixed phase to represent a 0 and with a 180° phase shift—that is, with the opposite polarity—to represent a 1. PSK was employed in the Bell 212 modem, which was introduced about 1980 to transmit information at rates up to 1,200 bits per second over the public switched telephone network.

Advanced methods

In addition to the elementary forms of digital modulation described above, there exist more advanced methods that result from a superposition of multiple modulating signals. An example of the latter form of modulation is quadrature amplitude modulation (QAM). QAM signals actually transmit two amplitude-modulated signals in phase quadrature (i.e., 90° apart), so that four or more bits are represented by each shift of the combined signal. Communications systems that employ QAM include digital cellular systems in the United States and Japan as well as most voiceband modems transmitting above 2,400 bits per second.

A form of modulation that combines convolutional codes with QAM is known as trellis-coded modulation (TCM), which is described in Channel encoding. Trellis-coded modulation forms an essential part of most of the modern voiceband modems operating at data rates of 9,600 bits per second and above, including V.32 and V.34 modems.

Multiplexing

Because of the installation cost of a communications channel, such as a microwave link or a coaxial cable link, it is desirable to share the channel among multiple users. Provided that the channel's data capacity exceeds that required to support a single user, the channel may be shared through the use of multiplexing methods. Multiplexing is the sharing of a communications channel through local combining of signals at a common point. Two types of multiplexing are commonly employed: frequency-division multiplexing and time-division multiplexing.

Frequency-division multiplexing

In frequency-division multiplexing (FDM), the available bandwidth of a communications channel is shared among multiple users by frequency translating, or modulating, each of the individual users onto a different carrier frequency. Assuming sufficient frequency separation of the carrier frequencies that the modulated signals do not overlap, recovery of each of the FDM signals is possible at the receiving end. In order to prevent overlap of the signals and to simplify filtering, each of the modulated signals is separated by a guard band, which consists of an unused portion of the available frequency spectrum. Each user is assigned a given frequency band for all time.

Analog multiplexing, as employed in the North American telephone systemIn frequency-division multiplexing (FDM), 12 separate voice signals, each of 4-kilohertz bandwidth, are modulated onto carrier waves in the 60–108-kilohertz range. These modulated signals are combined to form a single complex group signal. Groups are further combined to form a hierarchy of increasing bandwidth and voice-carrying capacity.
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While each user's information signal may be either analog or digital, the combined FDM signal is inherently an analog waveform. Therefore, an FDM signal must be transmitted over an analog channel. Examples of FDM are found in some of the old long-distance telephone transmission systems, including the American N- and L-carrier coaxial cable systems and analog point-to-point microwave systems. In the L-carrier system a hierarchical combining structure is employed in which 12 voiceband signals are frequency-division multiplexed to form a group signal in the frequency range of 60 to 108 kilohertz. Five group signals are multiplexed to form a supergroup signal in the frequency range of 312 to 552 kilohertz, corresponding to 60 voiceband signals, and 10 supergroup signals are multiplexed to form a master group signal. In the L1 carrier system, deployed in the 1940s, the master group was transmitted directly over coaxial cable. For microwave systems, it was frequency modulated onto a microwave carrier frequency for point-to-point transmission. In the L4 system, developed in the 1960s, six master groups were combined to form a jumbo group signal of 3,600 voiceband signals.

Time-division multiplexing

Multiplexing also may be conducted through the interleaving of time segments from different signals onto a single transmission path—a process known as time-division multiplexing (TDM). Time-division multiplexing of multiple signals is possible only when the available data rate of the channel exceeds the data rate of the total number of users. While TDM may be applied to either digital or analog signals, in practice it is applied almost always to digital signals. The resulting composite signal is thus also a digital signal.

Digital multiplexing, as employed in the North American telephone systemIn time-division multiplexing (TDM), 24 digitized voice signals, each at 64 kilobits per second, are assigned successive time slots in a 1.544-megabits-per-second signal. Combined signals are further combined to form data streams of increasing bit-rate and voice-carrying capacity.
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In a representative TDM system, data from multiple users are presented to a time-division multiplexer. A scanning switch then selects data from each of the users in sequence to form a composite TDM signal consisting of the interleaved data signals. Each user's data path is assumed to be time-aligned or synchronized to each of the other users' data paths and to the scanning mechanism. If only one bit were selected from each of the data sources, then the scanning mechanism would select the value of the arriving bit from each of the multiple data sources. In practice, however, the scanning mechanism usually selects a slot of data consisting of multiple bits of each user's data; the scanner switch is then advanced to the next user to select another slot, and so on. Each user is assigned a given time slot for all time.

Most modern telecommunications systems employ some form of TDM for transmission over long-distance routes. The multiplexed signal may be sent directly over cable systems, or it may be modulated onto a carrier signal for transmission via radio wave. Examples of such systems include the North American T carriers as well as digital point-to-point microwave systems. In T1 systems, introduced in 1962, 24 voiceband signals (or the digital equivalent) are time-division multiplexed together. The voiceband signal is a 64-kilobit-per-second data stream consisting of 8-bit symbols transmitted at a rate of 8,000 symbols per second. The TDM process interleaves 24 8-bit time slots together, along with a single frame-synchronization bit, to form a 193-bit frame. The 193-bit frames are formed at the rate of 8,000 frames per second, resulting in an overall data rate of 1.544 megabits per second. For transmission over more recent T-carrier systems, T1 signals are often further multiplexed to form higher-data-rate signals—again using a hierarchical scheme.

Multiple access

Multiplexing is defined as the sharing of a communications channel through local combining at a common point. In many cases, however, the communications channel must be efficiently shared among many users that are geographically distributed and that sporadically attempt to communicate at random points in time. Three schemes have been devised for efficient sharing of a single channel under these conditions; they are called frequency-division multiple access (FDMA), time-division multiple access (TDMA), and code-division multiple access (CDMA). These techniques can be used alone or together in telephone systems, and they are well illustrated by the most advanced mobile cellular systems.

Frequency-division multiple access

In FDMA the goal is to divide the frequency spectrum into slots and then to separate the signals of different users by placing them in separate frequency slots. The difficulty is that the frequency spectrum is limited and that there are typically many more potential communicators than there are available frequency slots. In order to make efficient use of the communications channel, a system must be devised for managing the available slots. In the advanced mobile phone system (AMPS), the cellular system employed in the United States, different callers use separate frequency slots via FDMA. When one telephone call is completed, a network-managing computer at the cellular base station reassigns the released frequency slot to a new caller. A key goal of the AMPS system is to reuse frequency slots whenever possible in order to accommodate as many callers as possible. Locally within a cell, frequency slots can be reused when corresponding calls are terminated. In addition, frequency slots can be used simultaneously by multiple callers located in separate cells. The cells must be far enough apart geographically that the radio signals from one cell are sufficiently attenuated at the location of the other cell using the same frequency slot.

Time-division multiple access

In TDMA the goal is to divide time into slots and separate the signals of different users by placing the signals in separate time slots. The difficulty is that requests to use a single communications channel occur randomly, so that on occasion the number of requests for time slots is greater than the number of available slots. In this case information must be buffered, or stored in memory, until time slots become available for transmitting the data. The buffering introduces delay into the system. In the IS54 cellular system, three digital signals are interleaved using TDMA and then transmitted in a 30-kilohertz frequency slot that would be occupied by one analog signal in AMPS. Buffering digital signals and interleaving them in time causes some extra delay, but the delay is so brief that it is not ordinarily noticed during a call. The IS54 system uses aspects of both TDMA and FDMA.

Code-division multiple access

In CDMA, signals are sent at the same time in the same frequency band. Signals are either selected or rejected at the receiver by recognition of a user-specific signature waveform, which is constructed from an assigned spreading code. The IS95 cellular system employs the CDMA technique. In IS95 an analog speech signal that is to be sent to a cell site is first quantized and then organized into one of a number of digital frame structures. In one frame structure, a frame of 20 milliseconds' duration consists of 192 bits. Of these 192 bits, 172 represent the speech signal itself, 12 form a cyclic redundancy check that can be used for error detection, and 8 form an encoder "tail" that allows the decoder to work properly. These bits are formed into an encoded data stream. After interleaving of the encoded data stream, bits are organized into groups of six. Each group of six bits indicates which of 64 possible waveforms to transmit. Each of the waveforms to be transmitted has a particular pattern of alternating polarities and occupies a certain portion of the radio-frequency spectrum. Before one of the waveforms is transmitted, however, it is multiplied by a code sequence of polarities that alternate at a rate of 1.2288 megahertz, spreading the bandwidth occupied by the signal and causing it to occupy (after filtering at the transmitter) about 1.23 megahertz of the radio-frequency spectrum. At the cell site one user can be selected from multiple users of the same 1.23-megahertz bandwidth by its assigned code sequence.

CDMA is sometimes referred to as spread-spectrum multiple access (SSMA), because the process of multiplying the signal by the code sequence causes the power of the transmitted signal to be spread over a larger bandwidth. Frequency management, a necessary feature of FDMA, is eliminated in CDMA. When another user wishes to use the communications channel, it is assigned a code and immediately transmits instead of being stored until a frequency slot opens.

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Citation Information

Article Title: telecommunication

Website Name: Encyclopaedia Britannica

Publisher: Encyclopaedia Britannica, Inc.

Date Published: 01 March 2019

URL: https://www.britannica.com/technology/telecommunication

Access Date: December 20, 2021