TELECOMMUNICATION SYSTEMS AND DEVICES

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ANNOTATION

Telecommunication, science and practice of transmitting information by electromagnetic means. Modern telecommunication centers 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.

Block diagram of a digital telecommunications system. Encyclopædia Britannica, Inc.

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 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 conversion An 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.

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.

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 quantize is a sequence of sampled amplitudes for which there are an infinite number of possible values. The output of the quantize, 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 "loss" system.) The degree of inaccuracy depends on the number of output levels used by the quantize. 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 quantize 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.

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