TELECOMMUNICATIONS SYSTEMS ENGINEERING FOR LIFE SUPPORT

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Contents

Summary

This chapter provides a discussion of the role of information in information and knowledge sharing and communications in support of sustainable development and other related subjects of international significance.

1. Life Support Telecommunications Issues

Telecommunications means conveying information at a distance. Virtually every aspect of modern civilization is dependent on the technology to do this that has emerged in the last century. Life support is no exception. Furthermore, life support has special requirements in that there is a particular need for the telecommunications used in life support systems to have these three attributes:

- *Reliability*: the system must function as designed in a very high fraction of all cases (greater than 99.99%).
- *Security*: the system must transfer information only to the intended recipient(s), and must ensure that information was sent by the apparent sender. It also must

remain available despite the efforts of adversarial parties.

- *Flexibility*: the system must be capable of transmitting information in multiple human-understandable and machine-readable media, and of rapid reconfiguration when the requirements for communication change.
- *Cost*: the telecommunications system must meet its requirements while consuming resources at the lowest practical level. This level must be one that is justified by its function within the larger system that employs it.

The first two attributes are associated with issues of trustworthiness, while the third and fourth are related to meeting evolving needs within acceptable, sustainable resources. This article will focus on telecommunications systems and technologies that are best suited to meet these requirements.

2. Principles of Telecommunications

Figure 1 shows a general model of the communication process, where a sender attempts to pass information to a receiver through a medium that is able to transmit information. A very important aspect of communication, shown in this model, is that no medium is ever totally free of interference or "noise." The noise corrupts the information or "signal" passed through the medium, although generally in practical cases enough information passes to be useful.

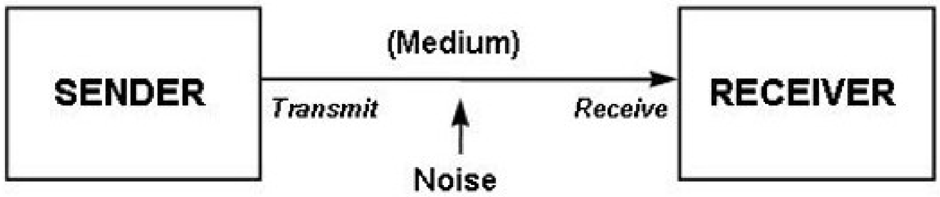

Figure 1: Model of Communication

The model helps us to understand several important forms of communication:

Telecommunication is any form of information transfer over a long distance. Normally it is accomplished by sending an electromagnetic signal through an appropriate medium. This could be electric current through wires, radio waves through free space, or light through optical fiber. In telecommunication, the signal, medium, and any associated connecting equipment often are referred to collectively as a *channel*.

Voice communication consists of transmitting human voice in a form that can be recognized by human hearing, as in the telephone. This was one of the two original forms of electromagnetic telecommunication. The other was telegraph, the earliest form of data communication.

Video communication consists of transmitting a moving image in a form that can be recognized by human vision as series of frames. For motion to appear smooth, the frame rate must be 24 frames per second or greater. Typically the frames are generated by a *raster*, which is a pattern that scans the input image in a sequence of horizontal lines from top to bottom. The required number of lines and resolution within the line varies with the size of the image produced at the receiver. United States standard broadcast video uses the "NTSC" standard with 525 visible lines of roughly 700 picture elements (*pixels*) each. The information rate present in such a signal is relatively large, equivalent to thousands of voice channels. Other systems such as the European "PAL" and "SECAM" standards and various proposed High Definition Television (HDTV) systems use even larger numbers of visual elements, while systems for video teleconferencing function at reduced frame rates and number of lines in order to reduce the demand on the channel (see the *bandwidth* concept below).

Data communication is the process of sending digital information, coded to represent alphanumeric characters, binary numbers consisting of digits 0 or 1, or other data that can be represented by combinations of 0 and 1. In this case the medium must be a channel that can accept digital information: discrete, identifiable states. Often these are often binary values, but in some cases there is a larger set of *symbols* (possible values) that the channel can pass. In data communication, noise takes the form of *errors*: cases where the received symbol is not the same as the transmitted symbol.

Multimedia communication is the process of sending information represented in more than one human-understandable medium, for example video, audio, still images, and text data. In general, multimedia systems are inherently capable of data communication.

3. Telecommunications Terminology

As with any advanced technology, there are many terms to be understood in the telecommunications area. Indeed, the field abounds with acronyms. In this section we focus on the concepts central to the technology, and associated terminology, with an effort to touch on the most important acronyms.

A signal that can take on a continuous range of values is said to be of analog form, and is characterized by its power level or amplitude and its rate of change or frequency. Relative change or phase is another parameter of interest (see the discussion of modems below). A signal with only discrete symbol values is said to be of digital form, and is characterized by the number of symbols per unit time or alternately by the number of bits per unit time. In this article we focus on digital systems, as they are generally more capable of meeting the reliability, flexibility, and cost requirements of life support. However, human senses demand analog interface, so we also consider below how analog sensory information can be conveyed by digital communications systems.

The capacity of a channel can be described in one of two ways. One is its bandwidth B, a measure of the range of analog signal frequencies that can pass through it and measured in Hertz, the units of frequency. The other is its data rate C, a measure of the number of information symbols per second that can pass through it, expressed in baud (symbols per second) or bits per second (b/s). These two measures are related by Shannon's law:

$$C = B \log_2 [1 + S/N] \qquad (1)$$

Where S is the desired signal's power in the channel, N is the power of noise (unwanted signal) in the channel, and $\log_2$ is the base two logarithm function (thus for example $\log_2(256)=8$). The rate at which information symbols are sampled in a channel is its baud rate b. If the more than two symbols are used, the data rate is determined by the number of bits per sample, n:

$$C = b\,n \qquad (2)$$

When data are communicated through a digital channel, it is required that precise timing relationship or "clocking" be maintained between sender and receiver. In the least complex arrangement to do this, known as asynchronous transmission, the code for each character is synchronized individually. A more efficient arrangement, synchronous transmission, collects the data into blocks or "frames" and places a synchronizing pattern at the beginning of the frame. Typically each frame will use some form of error detection code that works by sending a small amount of redundant information which can be used to check the accuracy of received data. The cyclic redundancy check (CRC) is the most commonly used technique. It will detect all single-bit errors, and burst errors (successive error bits) up to the length of the redundant frame check sequence (FCS) sent with the frame.

A simplex channel is one in which information can flow only one direction. A half duplex channel can be used in either direction, but only one direction at a time. A full duplex channel is capable of information flow in both directions simultaneously.

A network is a collection of senders and receivers (nodes), interconnected by channels or "links" such that a communications path is possible among any two of them, either directly or by going through an intermediary node of the network, as shown in Figure 2.

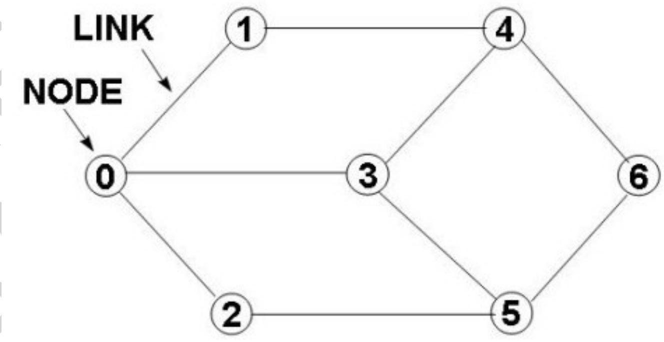

Figure 2: Network

A local area network (LAN) exists within a limited geographic area, usually a single building. Typically it is owned by the organization that uses it. Most LANs are capable of high data rates. Often they are constructed from very simple technology. For example, it is not uncommon for all nodes in a LAN to share a single length of wire as their communications medium.

A wide area network (WAN) covers a larger geographic area than a LAN, from city blocks to worldwide. Its links are almost always leased from a commercial communications carrier. Its data rate may range from quite low to very high, with cost roughly proportional to area of coverage and data rate.

A network is described as “public” if it offers service to anyone who pays to subscribe, and “private” if it is restricted to a particular group of users.

A distributed system is a collection of computers and software interconnected by a network, where the computers are programmed to cooperate in solving problems. In the sense that the nodes in today's data networks always have a computer at each node, data networks also are distributed systems.

An internet is a group of networks interconnected by gateways such that all of the nodes in the various networks are able to communicate, as in Figure 3. The best-known example is “the” Internet (note capitalization), consisting of thousands of networks worldwide that work together or interoperate. The ability to build interoperable networks derives from use of standard protocols: rules for transmitting information. The computers connected by an internet are called “hosts.”

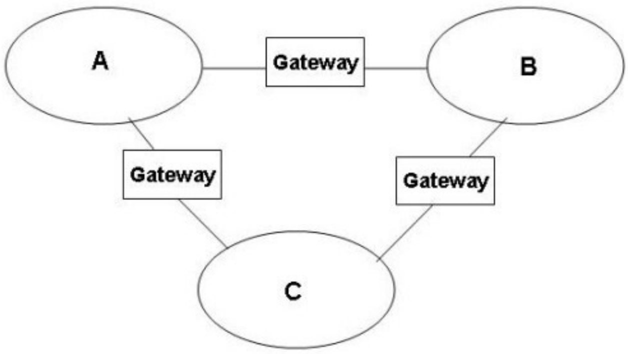

Figure 3: An Internet

An important aspect of a network's operation is how it passes information through a node. A network is said to be “switched” if its nodes can be programmed to automatically pass (“switch”) information between links so as to provide a path for the signal. One way to do this is to allocate a fixed fraction of each link's channel capacity, known as a “circuit” and interconnecting the circuits by circuit switching so that information flows through the network at a fixed rate as long as the switched path is in place. This is how most telephone systems work. The other common method is called packet switching. Here the information is divided into bundles or packets, each of which is passed around the network independently. Packet switching allows for more robust operation in that the packets can routed around a failed link or node automatically. Using packet switching it is possible to share the network’s facilities among arbitrary numbers of users, while also making maximum use of network resources by redirecting packets dynamically through underutilized portions of the network. The packet switches in an internet are called “routers.”

A broadcas*t* network is one in which all information goes to all nodes. Most LANs operate in a broadcast mode to avoid the expense of switching equipment. By contrast, unicast provides unique information flows as required between individual senders and receivers, and multicast provides for automatic distribution to specific groups of receivers from a sender.

4. Analog and Digital Communications

Analog and digital signals, suggested by Figure 4, are each used in telecommunications.

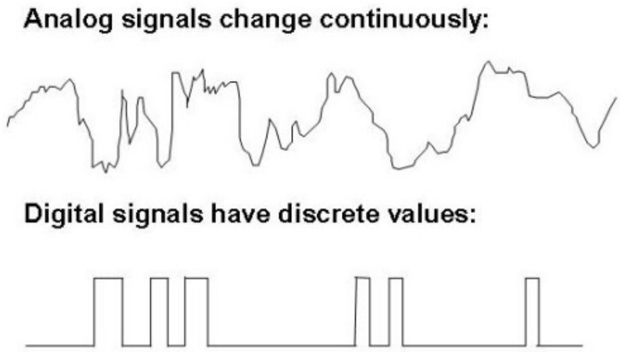

Figure 4: Analog and Digital Signals

The earliest voice communications systems represented the human voice in the communications medium by an analog signal where the variations of the signal were directly analogous to the variations in sound pressure created by the human vocal tract that, when they reach the human ear, are perceived as sound. Like analog signals, such sound pressures behave as continuous quantities. Channels that carry them are conditioned to respond in a linear fashion to the signal, in order to minimize distortion. The signal is corrupted by any nonlinearity and noise in the channel. Further, except for very simple media such as copper wires, an additional step of modulation is necessary. The purpose of this modulation is to impose the analog signal on a "carrier" signal that is suited to the channel, for example a radio frequency (RF) signal. The principal forms of modulation for analog signals are amplitude modulation (AM) in which the power level of the signal is varied in proportion to the analog signal, and frequency modulation (FM) in which the frequency of the carrier is varied in proportion to the analog signal. FM generally is superior because noise in the transmission medium is less likely to corrupt its delivered signal. Therefore FM is used widely in situations such as commercial broadcasting, where noise is a concern.

Modulation also is employed in cases where it is necessary to transmit data over analog channels. However in this case the analog channel itself is the medium through which the data must pass, so the carrier will be a signal suitable for a channel intended for human voice. Although the range of audible frequencies for a healthy, young person is in the vicinity of 20 to 20,000 Hertz, it is possible to convey the information in human speech adequately within the range 300 to 3,300 Hertz (bandwidth 3,000 Hertz). Telephone systems typically pass a bandwidth of 4,000 Hertz at most. Therefore data transmission systems use analog carrier frequencies within this range, modulated in ways that result in several bits coded within each signal sample (several bits per baud).

In addition to modulating the amplitude and frequency of the carrier (sometimes called amplitude shift keying - ASK and frequency shift keying - FSK), data systems modulate the “phase” or relative starting point of the carrier's waveform (phase shift keying - PSK). The device that transforms between the data signals produced by a computer and a form that will pass through the analog voice channel is called a modulator-demodulator or “modem.” Basic modems are available for data rates up to 19,200 bits per second. Recently, advanced modems have become available under the V.90 standard that include the ability to process the data for compression that is sending fewer information symbols by taking advantage of redundant patterns in the data. Such modems are rated at up to 56,000 bits per second. Basic modulation methods are suggested in Figure 5.

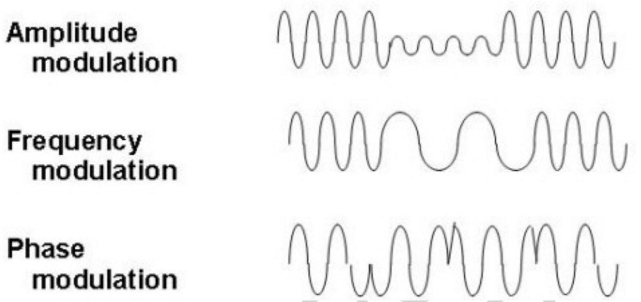

Figure 5: Modulation Methods

In the past two decades, a more sophisticated basis for communication has become widespread. Known as digital transmission, this system encodes information in the channel in discrete digital patterns and maintains a digital format between the sender and receiver. Where the signal has an analog form, it is transformed at the sending end to digital form by taking advantage of the sampling theorem which says that all information present in an analog signal can be captured by sampling it at a rate equal to twice its bandwidth (known as the Nyquist rate). The samples are then transformed into binary numbers by a process known as analog to digital conversion, resulting in a representation of the signal suitable for digital transmission. At the receiving end the digital signal undergoes digital to analog conversion. The resulting voice communication is much less susceptible to noise because the signal can be boosted along its path by regenerative repeaters that restore the digital form of the signal. By contrast, analog systems must use amplifiers that boost both signal and noise.

In addition to providing superior voice communication, digital transmission provides greatly improved options for data transmission. By making the digital voice transmission data path available end-to-end between computers, high data rates at multiples of 64,000 b/s are available. Digital telephony groups such digital voice channels into “carriers,” for example T1 consisting of 24 voice channels (data rate 1.536 megabits per second) and T3 consisting of 672 voice channels (data rate 44.736 megabits per second) in the North America and Japan. Similar European standards are E1 (2.048 Mb/s) and E3 (34.368 Mb/s). Customers willing to pay for such large amounts of communications resource can buy service from telephone carriers to interconnect computer systems. The equipment that combines multiple 64 kb/s channels in one carrier channel, and reverses the process at the other end of the channel, is called a multiplexer. The common standard for interoperable multiplexing has the name Synchronous Transfer Mode (STM).

Recently, digital channels have become available for lease in a new format called the Integrated Services Data Network (ISDN) that provides for "dial-up" connections for both digital voice and data. ISDN is available widely in Europe, Japan, and North America. Expected eventually to be available worldwide, ISDN provides a 56,000 or 64,000 b/s "B" channel to which a digital telephone system or data circuit adapter can be connected. ISDN currently is available in two forms. The basic rate interface (BRI) supports two B channels and a "D" (dialing) channel. The primary rate interface (PRI) supports 23 B channels plus a D channel in North America and Japan, and 30 B channels plus a D channel in Europe. The PRI is rapidly becoming the connection of choice in Europe for the private automatic branch exchange (PABX), a small telephone switch owned by the customer that provides on-site telephone switching and places outside calls through the ISDN trunks.

5. ATM Networks

A future option for very high capacity data channels is broadband ISDN (B-ISDN). This service is becoming available now for pilot tests and should become more widely available by the year 2000. It provides for direct connection to optical fiber, using an interface standard called Synchronous Optical Network (SONET) with data rates measured in optical connection (OC) increment that are multiples of 51.84 megabits per second, for example OC3 at 155 Mb/s. The B-ISDN system uses a form of packet switching called cell switching with a standardized cell 53 bytes in length. The mechanism for cell switching is an outgrowth of STM called Asynchronous Transfer Mode (ATM). Systems built around ATM switches can provide very high data rate connections in both LANs and WANs. Most often these high data rate systems are operated by communications carriers and used to provide high-capacity links to be used in packet-switched networks.

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Bibliography

Freeman, R. L. (1999). *Fundamentals of Telecommunications*. Wiley. [An excellent work that discusses the basic principles of telecommunications.]

Pullen, J. M. (2000). *Understanding Internet Protocols*. Wiley. [A work describing a wealth of material on Internet protocols.]

Stallings, W. (2000). *Data and Computer Communications*. 6th ed., Prentice-Hall,. [A classic work on communications.]

Tanenbaum, A. (1996). *Computer Networks*. 3rd Ed., Prentice-Hall. [This work provides a basic discussion of computer and communication networks.]

For further details see:

Roger L. Freeman, *Fundamentals of Telecommunications*, Wiley, 1999

J. Mark Pullen, *Understanding Internet Protocols*, Wiley, 2000

William Stallings, *Data and Computer Communications, 7th ed.*, Prentice-Hall, 2004

Andrew Tanenbaum, *Computer Networks, 4th Ed.*, Prentice-Hall, 1996

Biographical Sketch

J. Mark Pullen is Professor of Computer Science and Director of the C^4I Center at George Mason University, where he also heads the Networking and Simulation Laboratory. He holds BSEE and MSEE degrees from West Virginia University, and the Doctor of Science in Computer Science from the George Washington University. He is a licensed Professional Engineer, Fellow of the IEEE, and Fellow of the ACM. Prior to joining the GMU faculty he was an officer in the U.S. Army, in which capacity he served four years on the faculty of the U.S. Military Academy at West Point, New York and seven years at the Defense Advanced Research Projects Agency (DARPA). At DARPA he managed programs in high performance computing, networking, and simulation. Dr. Pullen teaches courses in computer networking and has active research projects in networking for distributed virtual simulation and networked multimedia tools for distributed collaboration and education. Dr. Pullen received the IEEE's Harry Diamond Memorial Award for his work in networking for distributed simulation.