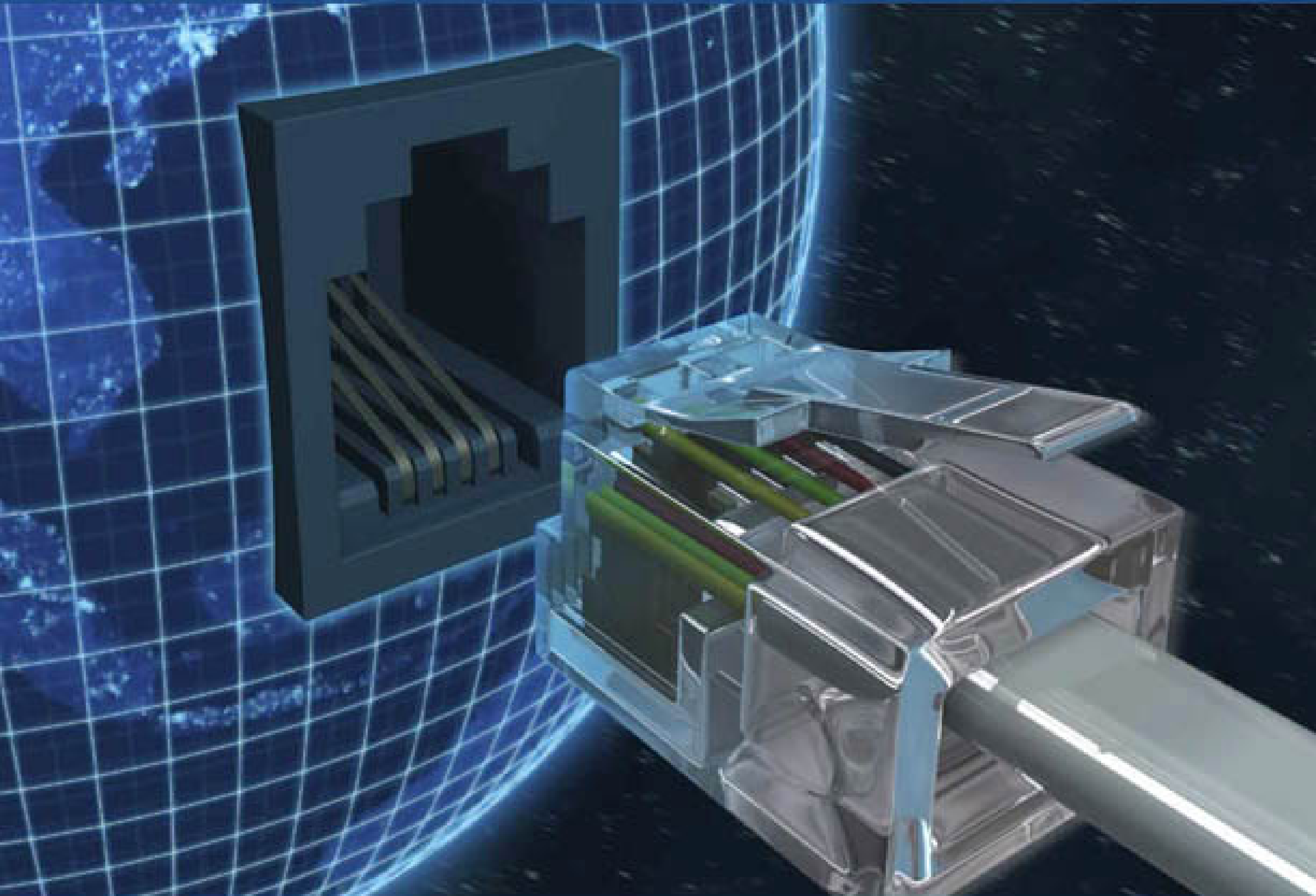
Elements and Applications of Telecommunications Engineering



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First Edition, 2012

ISBN 978-81-323-3543-6

Published by:
University Publications
4735/22 Prakashdeep Bldg,
Ansari Road, Darya Ganj,
Delhi - 110002
Email: info@wtbooks.com

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Telecommunication

A Gower telephone, at the *Musée des Arts et Métiers* in Paris

Telecommunication is the transmission of information, over significant distances, for the purpose of communication. In earlier times, telecommunications involved the use of visual signals, such as beacons, smoke, semaphore telegraphs, signal flags, and optical heliographs, or audio messages via coded drumbeats, lung-blown horns, or sent by loud whistles, for example. In the modern age of electricity and electronics, telecommunications now also includes the use of electrical devices such as telegraphs, telephones, and teletypes, the use of radio and microwave communications, as well as fiber optics and their associated electronics, plus the use of the orbiting satellites and the Internet.

The first breakthrough into modern electrical telecommunications came with the push to fully develop the telegraph starting in the 1830s. The use of these electrical means of communications exploded into use on all of the continents of the world during the 19th century, and these also connected the continents via cables on the floors of the ocean. The use of the first three popular systems of electrical telecommunications, the telegraph, telephone and teletype, all required the use of conducting metal wires.

A revolution in wireless telecommunications began in the first decade of the 20th century, with Guglielmo Marconi winning the Nobel Prize in Physics in 1909 for his pioneering developments in wireless radio communications. Other highly notable pioneering inventors and developers in the field of electrical and electronic telecommunications include Charles Wheatstone and Samuel Morse (telegraph), Alexander Graham Bell (telephone), Nikola Tesla, Edwin Armstrong, and Lee de Forest (radio), as well as John Logie Baird and Philo Farnsworth (television).

Telecommunications play an important role in the world economy and the worldwide telecommunication industry's revenue was estimated to be $3.85 trillion in 2008. The service revenue of the global telecommunications industry was estimated to be $1.7 trillion in 2008, and is expected to touch $2.7 trillion by 2013.

History

Ancient systems

Greek hydraulic semaphore systems were used as early as the 4th century BCE. The hydraulic semaphores, which worked with water filled vessels and visual signals, functioned as optical telegraphs. However, they could only utilize a very limited range of pre-determined messages, and as with all such optical telegraphs could only be deployed during good visibility conditions.

During the Middle Ages, chains of beacons were commonly used on hilltops as a means of relaying a signal. Beacon chains suffered the drawback that they could only pass a single bit of information, so the meaning of the message such as "the enemy has been sighted" had to be agreed upon in advance. One notable instance of their use was during the Spanish Armada, when a beacon chain relayed a signal from Plymouth to London that signaled the arrival of the Spanish warships.

Systems since the Middle Ages

A replica of one of Chappe's semaphore towers in Nalbach

In 1792, Claude Chappe, a French engineer, built the first fixed visual telegraphy system (or semaphore line) between Lille and Paris. However semaphore systems suffered from the need for skilled operators and the expensive towers at intervals of ten to thirty kilometers (six to twenty miles). As a result of competition from the electrical telegraph, Europe's last commercial semaphore line in Sweden was abandoned in 1880.

The telegraph and telephone

The first commercial electrical telegraph was constructed by Sir Charles Wheatstone and Sir William Fothergill Cooke, and its use began on April 9, 1839. Both Wheatstone and

Cooke viewed their device as "an improvement to the [already-existing, so-called] electromagnetic telegraph" not as a new device.

The businessman Samuel F.B. Morse and the physicist Joseph Henry of the United States developed their own, simpler version of the electrical telegraph, independently. Morse successfully demonstrated this system on September 2, 1837. Morse's most important technical contribution to this telegraph was the rather simple and highly efficient Morse Code, which was an important advance over Wheatstone's complicated and significantly more expensive telegraph system. The communications efficiency of the Morse Code anticipated that of the Huffman code in digital communications by over 100 years, but Morse and his associate Alfred Vail developed the code purely empirically, unlike Huffman, who gave a detailed theoretical explanation of how his method worked.

The first permanent transatlantic telegraph cable was successfully completed on 27 July 1866, allowing transatlantic electrical communication for the first time. An earlier transatlantic cable had operated for a few months in 1859, and among other things, it carried messages of greeting back and forth between President James Buchanan of the United States and Queen Victoria of the United Kingdom.

However, that transatlantic cable failed soon, and the project to lay a replacement line was delayed for five years by the American Civil War. Also, these transatlantic cables would have been completely incapable of carrying telephone calls even had the telephone already been invented. The first transatlantic telephone cable (which incorporated hundreds of electronic amplifiers) was not operational until 1956.

The conventional telephone now in use worldwide was first patented by Alexander Graham Bell in March 1876. That first patent by Bell was the *master patent* of the telephone, from which all other patents for electric telephone devices and features flowed. Credit for the invention of the electric telephone has been frequently disputed, and new controversies over the issue have arisen from time-to-time. As with other great inventions such as radio, television, the light bulb, and the digital computer, there were several inventors who did pioneering experimental work on *voice transmission over a wire*, and then they improved on each other's ideas. However, the key innovators were Alexander Graham Bell and Gardiner Greene Hubbard, who created the first telephone company, the Bell Telephone Company in the United States, which later evolved into American Telephone & Telegraph (AT&T).

The first commercial telephone services were set up in 1878 and 1879 on both sides of the Atlantic in the cities of New Haven, Connecticut, and London, England.

Radio and television

In 1832, James Lindsay gave a classroom demonstration of wireless telegraphy via conductive water to his students. By 1854, he was able to demonstrate a transmission across the Firth of Tay from Dundee, Scotland, to Woodhaven, a distance of about two miles (3 km), again using water as the transmission medium. In December 1901, Guglielmo Marconi established wireless communication between St. John's, Newfoundland and Poldhu, Cornwall (England), earning him the Nobel Prize in Physics for 1909, one which he shared with Karl Braun. However *small-scale* radio communication had already been demonstrated in 1893 by Nikola Tesla in a presentation before the National Electric Light Association.

On March 25, 1925, John Logie Baird of England was able to demonstrate the transmission of moving pictures at the Selfridge's department store in London, England. Baird's system relied upon the fast-rotating Nipkow disk, and thus it became known as the mechanical television. It formed the basis of experimental broadcasts done by the British Broadcasting Corporation beginning September 30, 1929. However, for most of the 20th century, television systems were designed around the cathode ray tube, invented by Karl Braun. The first version of such an electronic television to show promise was produced by Philo Farnsworth of the United States, and it was demonstrated to his family in Idaho on September 7, 1927.

Computer networks and the Internet

On 11 September 1940, George Stibitz was able to transmit problems using teletype to his Complex Number Calculator in New York and receive the computed results back at Dartmouth College in New Hampshire. This configuration of a centralized computer or mainframe computer with remote "dumb terminals" remained popular throughout the 1950s and into the 60's. However, it was not until the 1960s that researchers started to investigate packet switching — a technology that allows chunks of data to be sent between different computers without first passing through a centralized mainframe. A four-node network emerged on December 5, 1969. This network soon became the ARPANET, which by 1981 would consist of 213 nodes.

ARPANET's development centred around the Request for Comment process and on 7 April 1969, RFC 1 was published. This process is important because ARPANET would eventually merge with other networks to form the Internet, and many of the communication protocols that the Internet relies upon today were specified through the Request for Comment process. In September 1981, RFC 791 introduced the Internet Protocol version 4 (IPv4) and RFC 793 introduced the Transmission Control Protocol (TCP) — thus creating the TCP/IP protocol that much of the Internet relies upon today.

However, not all important developments were made through the Request for Comment process. Two popular link protocols for local area networks (LANs) also appeared in the 1970s. A patent for the token ring protocol was filed by Olof Soderblom on October 29, 1974, and a paper on the Ethernet protocol was published by Robert Metcalfe and David Boggs in the July 1976 issue of *Communications of the ACM*. The Ethernet protocol had been inspired by the ALOHAnet protocol which had been developed by electrical engineering researchers at the University of Hawaii.

Key concepts

Etymology

The word *telecommunication* was adapted from the French word *télécommunication*. It is a compound of the Greek prefix *tele-* (τηλε-), meaning "far off", and the Latin *communicare*, meaning "to share". The French word *télécommunication* was coined in 1904 by the French engineer and novelist Édouard Estaunié.

A number of key concepts reoccur throughout the literature on modern telecommunication systems. Some of these concepts are discussed below.

Basic elements

A basic telecommunication system consists of three primary units that are always present in some form:

- A transmitter that takes information and converts it to a signal.
- A transmission medium, also called the "physical channel" that carries the signal. An example of this is the "free space channel".
- A receiver that takes the signal from the channel and converts it back into usable information.

For example, in a radio broadcasting station the station's large power amplifier is the transmitter; and the broadcasting antenna is the interface between the power amplifier and the "free space channel". The free space channel is the transmission medium; and the receiver's antenna is the interface between the free space channel and the receiver. Next, the radio receiver is the destination of the radio signal, and this is where it is converted from electricity to sound for people to listen to.

Sometimes, telecommunication systems are "duplex" (two-way systems) with a single box of electronics working as both a transmitter and a receiver, or a *transceiver*. For example, a cellular telephone is a transceiver. The transmission electronics and the receiver electronics in a transceiver are actually quite independent of each other. This can be readily explained by the fact that radio transmitters contain power amplifiers that operate with electrical powers measured in the watts or kilowatts, but radio receivers deal with radio powers that are measured in the microwatts or nanowatts. Hence, transceivers have to be carefully designed and built to isolate their high-power circuitry and their low-power circuitry from each other.

Telecommunication over telephone lines is called point-to-point communication because it is between one transmitter and one receiver. Telecommunication through radio broadcasts is called broadcast communication because it is between one powerful transmitter and numerous low-power but sensitive radio receivers.

Telecommunications in which multiple transmitters and multiple receivers have been designed to cooperate and to share the same physical channel are called multiplex systems.

Analog versus digital communications

Communications signals can be either by analog signals or digital signals. There are analog communication systems and digital communication systems. For an analog signal, the signal is varied continuously with respect to the information. In a digital signal, the information is encoded as a set of discrete values (for example, a set of ones and zeros). During the propagation and reception, the information contained in analog signals will inevitably be degraded by undesirable physical noise. (The output of a transmitter is noise-free for all practical purposes.) Commonly, the noise in a communication system can be expressed as adding or subtracting from the desirable signal in a completely random way. This form of noise is called ***"additive noise"***, with the understanding that the noise can be negative or positive at different instants of time. Noise that is not additive noise is a much more difficult situation to describe or analyze, and these other kinds of noise will be omitted here.

On the other hand, unless the *additive noise* disturbance exceeds a certain threshold, the information contained in digital signals will remain intact. Their resistance to noise represents a key advantage of digital signals over analog signals.

Telecommunication networks

A communications network is a collection of transmitters, receivers, and communications channels that send messages to one another. Some digital communications networks contain one or more routers that work together to transmit information to the correct user. An analog communications network consists of one or more switches that establish a connection between two or more users. For both types of network, repeaters may be necessary to amplify or recreate the signal when it is being transmitted over long distances. This is to combat attenuation that can render the signal indistinguishable from the noise.

Communication channels

The term "channel" has two different meanings. In one meaning, a channel is the physical medium that carries a signal between the transmitter and the receiver. Examples of this include the atmosphere for sound communications, glass optical fibers for some kinds of optical communications, coaxial cables for communications by way of the voltages and electric currents in them, and free space for communications using visible light, infrared waves, ultraviolet light, and radio waves. This last channel is called the "free space channel". The sending of radio waves from one place to another has nothing to do with the presence or absence of an atmosphere between the two. Radio waves travel through a perfect vacuum just as easily as they travel through air, fog, clouds, or any other kind of gas besides air.

The other meaning of the term "channel" in telecommunications is seen in the phrase communications channel, which is a subdivision of a transmission medium so that it can be used to send multiple streams of information simultaneously. For example, one radio station can broadcast radio waves into free space at frequencies in the neighborhood of 94.5 MHz (megahertz) while another radio station can simultaneously broadcast radio waves at frequencies in the neighborhood of 96.1 MHz. Each radio station would transmit radio waves over a frequency bandwidth of about 180 kHz (kilohertz), centered at frequencies such as the above, which are called the "carrier frequencies". Each station in this example is separated from its adjacent stations by 200 kHz, and the difference between 200 kHz and 180 kHz (20 kHz) is an engineering allowance for the imperfections in the communication system.

In the example above, the "free space channel" has been divided into communications channels according to frequencies, and each channel is assigned a separate frequency bandwidth in which to broadcast radio waves. This system of dividing the medium into channels according to frequency is called "frequency-division multiplexing" (**FDM**).

Another way of dividing a communications medium into channels is to allocate each sender a recurring segment of time (a "time slot", for example, 20 milliseconds out of each second), and to allow each sender to send messages only within its own time slot. This method of dividing the medium into communication channels is called "time-division multiplexing" (**TDM**), and is used in optical fiber communication. Some radio communication systems use TDM within an allocated FDM channel. Hence, these systems use a hybrid of TDM and FDM.

Modulation

The shaping of a signal to convey information is known as modulation. Modulation can be used to represent a digital message as an analog waveform. This is commonly called "keying" - a term derived from the older use of Morse Code in telecommunications - and several keying techniques exist (these include phase-shift keying, frequency-shift keying, and amplitude-shift keying). The "Bluetooth" system, for example, uses phase-shift keying to exchange information between various devices. In addition, there are combinations of phase-shift keying and amplitude-shift keying which is called (in the jargon of the field) "quadrature amplitude modulation" (QAM) that are used in high-capacity digital radio communication systems.

Modulation can also be used to transmit the information of low-frequency analog signals at higher frequencies. This is helpful because low-frequency analog signals cannot be effectively transmitted over free space. Hence the information from a low-frequency analog signal must be impressed into a higher-frequency signal (known as the "carrier wave") before transmission. There are several different modulation schemes available to achieve this [two of the most basic being amplitude modulation (AM) and frequency modulation (FM)]. An example of this process is a disc jockey's voice being impressed into a 96 MHz carrier wave using frequency modulation (the voice would then be received on a radio as the channel "96 FM"). In addition, modulation has the advantage of being about to use frequency division multiplexing (FDM).

Society and telecommunication

Telecommunication has a significant social, cultural. and economic impact on modern society. In 2008, estimates placed the telecommunication industry's revenue at $3.85 trillion (USD) or just under 3.0 percent of the gross world product (official exchange rate). Several following sections discuss the impact of telecommunication on society.

Economic impact

Microeconomics

On the microeconomic scale, companies have used telecommunications to help build global business empires. This is self-evident in the case of online retailer Amazon.com but, according to academic Edward Lenert, even the conventional retailer Wal-Mart has benefited from better telecommunication infrastructure compared to its competitors. In cities throughout the world, home owners use their telephones to organize many home services ranging from pizza deliveries to electricians. Even relatively-poor communities have been noted to use telecommunication to their advantage. In Bangladesh's Narshingdi district, isolated villagers use cellular phones to speak directly to wholesalers and arrange a better price for their goods. In Côte d'Ivoire, coffee growers share mobile phones to follow hourly variations in coffee prices and sell at the best price.

Macroeconomics

On the macroeconomic scale, Lars-Hendrik Röller and Leonard Waverman suggested a causal link between good telecommunication infrastructure and economic growth. Few dispute the existence of a correlation although some argue it is wrong to view the relationship as causal.

Because of the economic benefits of good telecommunication infrastructure, there is increasing worry about the inequitable access to telecommunication services amongst various countries of the world—this is known as the digital divide. A 2003 survey by the International Telecommunication Union (ITU) revealed that roughly one-third of countries have fewer than one mobile subscription for every 20 people and one-third of countries have fewer than one land-line telephone subscription for every 20 people. In terms of Internet access, roughly half of all countries have fewer than one out of 20 people with Internet access. From this information, as well as educational data, the ITU was able to compile an index that measures the overall ability of citizens to access and use information and communication technologies. Using this measure, Sweden, Denmark and Iceland received the highest ranking while the African countries Nigeria, Burkina Faso and Mali received the lowest.

Social impact

Telecommunication has played a significant role in social relationships. Nevertheless devices like the telephone system were originally advertised with an emphasis on the practical dimensions of the device (such as the ability to conduct business or order home services) as opposed to the social dimensions. It was not until the late 1920s and 1930s that the social dimensions of the device became a prominent theme in telephone advertisements. New promotions started appealing to consumers' emotions, stressing the importance of social conversations and staying connected to family and friends.

Since then the role that telecommunications has played in social relations has become increasingly important. In recent years, the popularity of social networking sites has increased dramatically. These sites allow users to communicate with each other as well as post photographs, events and profiles for others to see. The profiles can list a person's age, interests, sexual preference and relationship status. In this way, these sites can play important role in everything from organising social engagements to courtship.

Prior to social networking sites, technologies like short message service(SMS) and the telephone also had a significant impact on social interactions. In 2000, market research group Ipsos MORI reported that 81% of 15 to 24 year-old SMS users in the United Kingdom had used the service to coordinate social arrangements and 42% to flirt.

Other impacts

In cultural terms, telecommunication has increased the public's ability to access to music and film. With television, people can watch films they have not seen before in their own home without having to travel to the video store or cinema. With radio and the Internet, people can listen to music they have not heard before without having to travel to the music store.

Telecommunication has also transformed the way people receive their news. A survey by the non-profit Pew Internet and American Life Project found that when just over 3,000 people living in the United States were asked where they got their news "yesterday", more people said television or radio than newspapers. The results are summarised in the following table (the percentages add up to more than 100% because people were able to specify more than one source).

Local TV	National TV	Radio	Local paper	Internet	National paper
59%	47%	44%	38%	23%	12%

Telecommunication has had an equally significant impact on advertising. TNS Media Intelligence reported that in 2007, 58% of advertising expenditure in the United States was spent on mediums that depend upon telecommunication. The results are summarised in the following table.

	Internet	Radio	Cable TV	Syndicated TV	Spot TV	Network TV	Newspaper	Magazine	Outdoor	Total
Percent	7.6%	7.2%	12.1%	2.8%	11.3%	17.1%	18.9%	20.4%	2.7%	100%
Dollars	$11.31 billion	$10.69 billion	$18.02 billion	$4.17 billion	$16.82 billion	$25.42 billion	$28.22 billion	$30.33 billion	$4.02 billion	$149 billion

Telecommunication and government

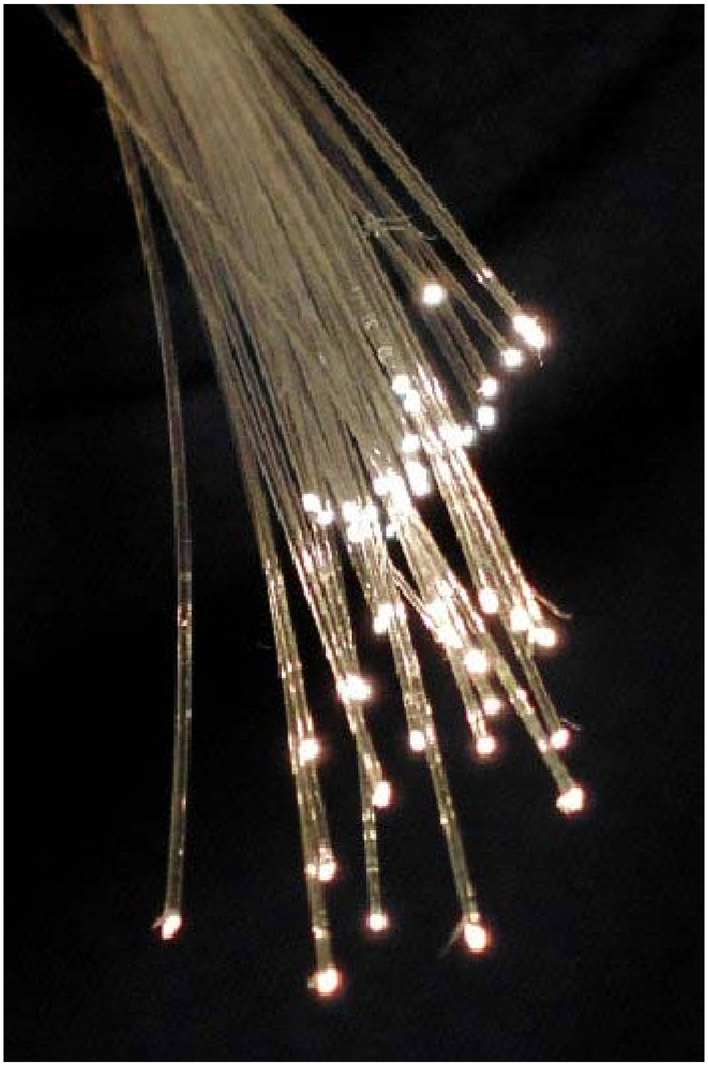
Many countries have enacted legislation which conform to the *International Telecommunication Regulations* establish by the International Telecommunication Union (ITU), which is the "leading United Nations agency for information and communication technology issues." In 1947, at the Atlantic City Conference, the ITU decided to "afford international protection to all frequencies registered in a new international frequency list and used in conformity with the Radio Regulation." According to the ITU's *Radio Regulations* adopted in Atlantic City, all frequencies referenced in the *International Frequency Registration Board*, examined by the board and registered on the *International Frequency List* "shall have the right to international protection from harmful interference."

From a global perspective, there have been political debates and legislation regarding the management of telecommunication and broadcasting. The history of broadcasting discusses some of debates in relation to balancing conventional communication such as printing and telecommunication such as radio broadcasting. The onset of World War II brought on the first explosion of international broadcasting propaganda. Countries, their governments, insurgents, terrorists, and militiamen have all used telecommunication and broadcasting techniques to promote propaganda. Patriotic propaganda for political movements and colonization started the mid 1930s. In 1936, the BBC did broadcast propaganda to the Arab World to partly counter similar broadcasts from Italy, which also had colonial interests in North Africa.

Modern insurgents, such as those in the latest Iraq war, often use intimidating telephone calls, SMSs and the distribution of sophisticated videos of an attack on coalition troops within hours of the operation. "The Sunni insurgents even have their own television station, Al-Zawraa, which while banned by the Iraqi government, still broadcasts from Erbil, Iraqi Kurdistan, even as coalition pressure has forced it to switch satellite hosts several times."

Modern telecommunication

Telephone



Optical fiber provides cheaper bandwidth for long distance communication

In an analog telephone network, the caller is connected to the person he wants to talk to by switches at various telephone exchanges. The switches form an electrical connection between the two users and the setting of these switches is determined electronically when the caller dials the number. Once the connection is made, the caller's voice is transformed to an electrical signal using a small microphone in the caller's handset. This electrical signal is then sent through the network to the user at the other end where it is transformed back into sound by a small speaker in that person's handset. There is a separate electrical connection that works in reverse, allowing the users to converse.

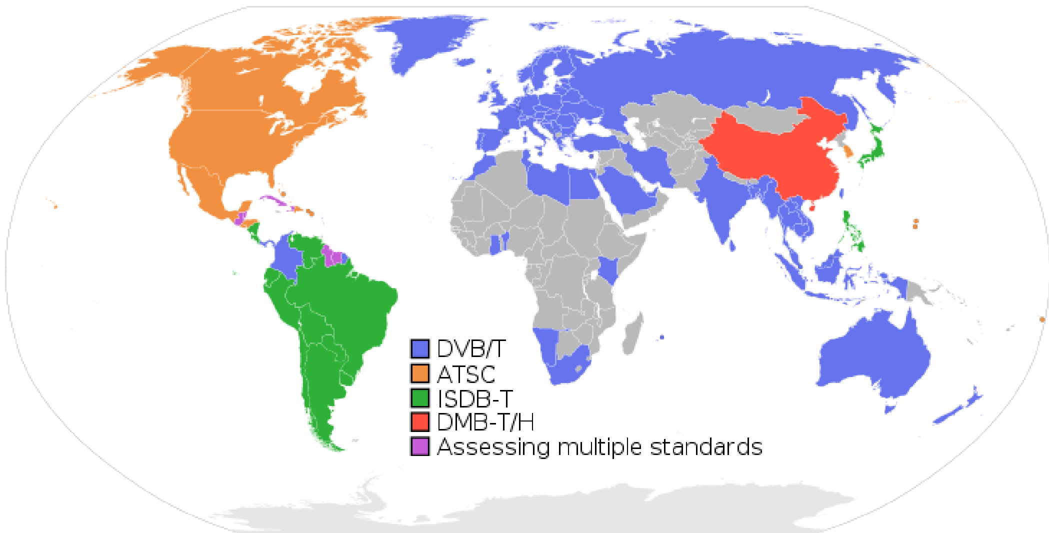
The fixed-line telephones in most residential homes are analog — that is, the speaker's voice directly determines the signal's voltage. Although short-distance calls may be handled from end-to-end as analog signals, increasingly telephone service providers are transparently converting the signals to digital for transmission before converting them back to analog for reception. The advantage of this is that digitized voice data can travel side-by-side with data from the Internet and can be perfectly reproduced in long distance communication (as opposed to analog signals that are inevitably impacted by noise).

Mobile phones have had a significant impact on telephone networks. Mobile phone subscriptions now outnumber fixed-line subscriptions in many markets. Sales of mobile phones in 2005 totalled 816.6 million with that figure being almost equally shared amongst the markets of Asia/Pacific (204 m), Western Europe (164 m), CEMEA (Central Europe, the Middle East and Africa) (153.5 m), North America (148 m) and Latin America (102 m). In terms of new subscriptions over the five years from 1999, Africa has outpaced other markets with 58.2% growth. Increasingly these phones are being serviced by systems where the voice content is transmitted digitally such as GSM or W-CDMA with many markets choosing to depreciate analog systems such as AMPS.

There have also been dramatic changes in telephone communication behind the scenes. Starting with the operation of TAT-8 in 1988, the 1990s saw the widespread adoption of systems based on optic fibres. The benefit of communicating with optic fibers is that they offer a drastic increase in data capacity. TAT-8 itself was able to carry 10 times as many telephone calls as the last copper cable laid at that time and today's optic fibre cables are able to carry 25 times as many telephone calls as TAT-8. This increase in data capacity is due to several factors: First, optic fibres are physically much smaller than competing technologies. Second, they do not suffer from crosstalk which means several hundred of them can be easily bundled together in a single cable. Lastly, improvements in multiplexing have led to an exponential growth in the data capacity of a single fibre.

Assisting communication across many modern optic fibre networks is a protocol known as Asynchronous Transfer Mode (ATM). The ATM protocol allows for the side-by-side data transmission mentioned in the second paragraph. It is suitable for public telephone networks because it establishes a pathway for data through the network and associates a traffic contract with that pathway. The traffic contract is essentially an agreement between the client and the network about how the network is to handle the data; if the network cannot meet the conditions of the traffic contract it does not accept the connection. This is important because telephone calls can negotiate a contract so as to guarantee themselves a constant bit rate, something that will ensure a caller's voice is not delayed in parts or cut-off completely. There are competitors to ATM, such as Multiprotocol Label Switching (MPLS), that perform a similar task and are expected to supplant ATM in the future.

Radio and television



Digital television standards and their adoption worldwide.

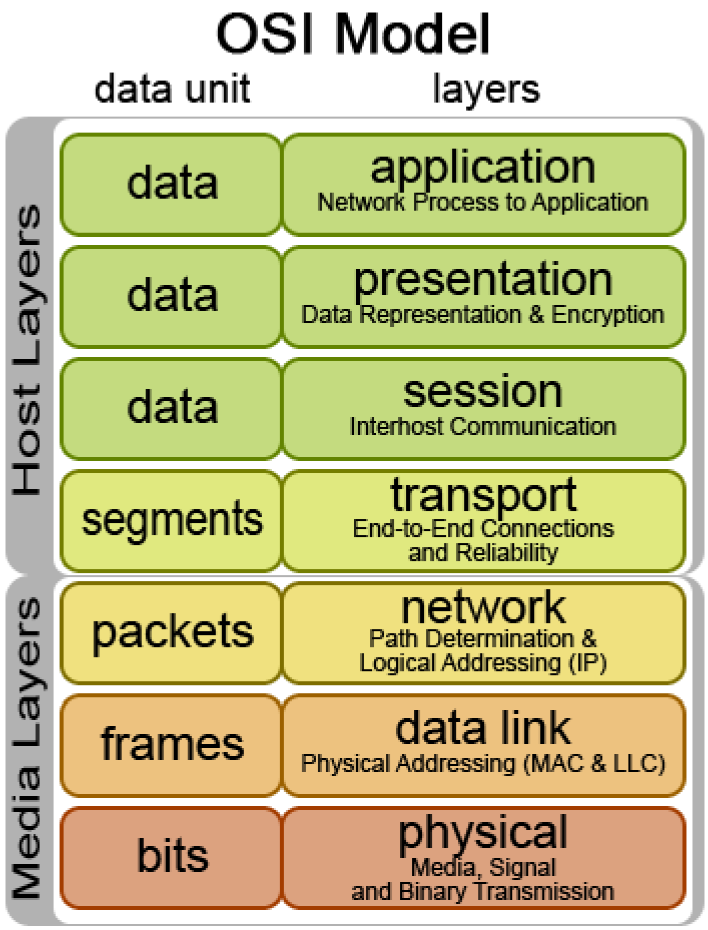
In a broadcast system, the central high-powered broadcast tower transmits a high-frequency electromagnetic wave to numerous low-powered receivers. The high-frequency wave sent by the tower is modulated with a signal containing visual or audio information. The receiver is then tuned so as to pick up the high-frequency wave and a demodulator is used to retrieve the signal containing the visual or audio information. The broadcast signal can be either analog (signal is varied continuously with respect to the information) or digital (information is encoded as a set of discrete values).

The broadcast media industry is at a critical turning point in its development, with many countries moving from analog to digital broadcasts. This move is made possible by the production of cheaper, faster and more capable integrated circuits. The chief advantage of digital broadcasts is that they prevent a number of complaints common to traditional analog broadcasts. For television, this includes the elimination of problems such as snowy pictures, ghosting and other distortion. These occur because of the nature of analog transmission, which means that perturbations due to noise will be evident in the final output. Digital transmission overcomes this problem because digital signals are reduced to discrete values upon reception and hence small perturbations do not affect the final output. In a simplified example, if a binary message 1011 was transmitted with signal amplitudes [1.0 0.0 1.0 1.0] and received with signal amplitudes [0.9 0.2 1.1 0.9] it would still decode to the binary message 1011 — a perfect reproduction of what was sent. From this example, a problem with digital transmissions can also be seen in that if the noise is great enough it can significantly alter the decoded message. Using forward error correction a receiver can correct a handful of bit errors in the resulting message but too much noise will lead to incomprehensible output and hence a breakdown of the transmission.

In digital television broadcasting, there are three competing standards that are likely to be adopted worldwide. These are the ATSC, DVB and ISDB standards; the adoption of these standards thus far is presented in the captioned map. All three standards use MPEG-2 for video compression. ATSC uses Dolby Digital AC-3 for audio compression, ISDB uses Advanced Audio Coding (MPEG-2 Part 7) and DVB has no standard for audio compression but typically uses MPEG-1 Part 3 Layer 2. The choice of modulation also varies between the schemes. In digital audio broadcasting, standards are much more unified with practically all countries choosing to adopt the Digital Audio Broadcasting standard (also known as the Eureka 147 standard). The exception being the United States which has chosen to adopt HD Radio. HD Radio, unlike Eureka 147, is based upon a transmission method known as in-band on-channel transmission that allows digital information to "piggyback" on normal AM or FM analog transmissions.

However, despite the pending switch to digital, analog television remains being transmitted in most countries. An exception is the United States that ended analog television transmission (by all but the very low-power TV stations) on 12 June 2009 after twice delaying the switchover deadline. For analog television, there are three standards in use for broadcasting color TV. These are known as PAL (British designed), NTSC (North American designed), and SECAM (French designed). (It is important to understand that these are the ways from sending color TV, and they do not have anything to do with the standards for black & white TV, which also vary from country to country.) For analog radio, the switch to digital radio is made more difficult by the fact that analog receivers are sold at a small fraction of the price of digital receivers. The choice of modulation for analog radio is typically between amplitude modulation (**AM**) or frequency modulation (**FM**). To achieve stereo playback, an amplitude modulated subcarrier is used for stereo FM.

The Internet



The OSI reference model

The Internet is a worldwide network of computers and computer networks that can communicate with each other using the Internet Protocol. Any computer on the Internet has a unique IP address that can be used by other computers to route information to it. Hence, any computer on the Internet can send a message to any other computer using its IP address. These messages carry with them the originating computer's IP address allowing for two-way communication. The Internet is thus an exchange of messages between computers.

As of 2008, an estimated 21.9% of the world population has access to the Internet with the highest access rates (measured as a percentage of the population) in North America (73.6%), Oceania/Australia (59.5%) and Europe (48.1%). In terms of broadband access, Iceland (26.7%), South Korea (25.4%) and the Netherlands (25.3%) led the world.

The Internet works in part because of protocols that govern how the computers and routers communicate with each other. The nature of computer network communication lends itself to a layered approach where individual protocols in the protocol stack run more-or-less independently of other protocols. This allows lower-level protocols to be customized for the network situation while not changing the way higher-level protocols operate. A practical example of why this is important is because it allows an Internet browser to run the same code regardless of whether the computer it is running on is connected to the Internet through an Ethernet or Wi-Fi connection. Protocols are often talked about in terms of their place in the OSI reference model (pictured on the right), which emerged in 1983 as the first step in an unsuccessful attempt to build a universally adopted networking protocol suite.

For the Internet, the physical medium and data link protocol can vary several times as packets traverse the globe. This is because the Internet places no constraints on what physical medium or data link protocol is used. This leads to the adoption of media and protocols that best suit the local network situation. In practice, most intercontinental communication will use the Asynchronous Transfer Mode (ATM) protocol (or a modern equivalent) on top of optic fibre. This is because for most intercontinental communication the Internet shares the same infrastructure as the public switched telephone network.

At the network layer, things become standardized with the Internet Protocol (IP) being adopted for logical addressing. For the World Wide Web, these "IP addresses" are derived from the human readable form using the Domain Name System (e.g. 72.14.207.99 is derived from www.google.com). At the moment, the most widely used version of the Internet Protocol is version four but a move to version six is imminent.

At the transport layer, most communication adopts either the Transmission Control Protocol (TCP) or the User Datagram Protocol (UDP). TCP is used when it is essential every message sent is received by the other computer where as UDP is used when it is merely desirable. With TCP, packets are retransmitted if they are lost and placed in order before they are presented to higher layers. With UDP, packets are not ordered or retransmitted if lost. Both TCP and UDP packets carry port numbers with them to specify what application or process the packet should be handled by. Because certain application-level protocols use certain ports, network administrators can manipulate traffic to suit particular requirements. Examples are to restrict Internet access by blocking the traffic destined for a particular port or to affect the performance of certain applications by assigning priority.

Above the transport layer, there are certain protocols that are sometimes used and loosely fit in the session and presentation layers, most notably the Secure Sockets Layer (SSL) and Transport Layer Security (TLS) protocols. These protocols ensure that the data transferred between two parties remains completely confidential and one or the other is in use when a padlock appears in the address bar of your web browser. Finally, at the application layer, are many of the protocols Internet users would be familiar with such as HTTP (web browsing), POP3 (e-mail), FTP (file transfer), IRC (Internet chat), BitTorrent (file sharing) and OSCAR (instant messaging).

Local Area Networks and Wide Area Networks

Despite the growth of the Internet, the characteristics of local area networks ("LANs" - computer networks that do not extend beyond a few kilometers in size) remain distinct. This is because networks on this scale do not require all the features associated with larger networks and are often more cost-effective and efficient without them. When they are not connected with the Internet, they also have the advantages of privacy and security. However, purposefully lacking a direct connection to the Internet will not provide 100% protection of the LAN from hackers, military forces, or economic powers. These threats exist if there are any methods for connecting remotely to the LAN.

There are also independent wide area networks ("WANs" - private computer networks that can and do extend for thousands of kilometers.) Once again, some of their advantages include their privacy, security, and complete ignoring of any potential hackers - who cannot "touch" them. Of course, prime users of private LANs and WANs include armed forces and intelligence agencies that *must* keep their information completely secure and secret.

In the mid-1980s, several sets of communication protocols emerged to fill the gaps between the data-link layer and the application layer of the OSI reference model. These included Appletalk, IPX, and NetBIOS with the dominant protocol set during the early 1990s being IPX due to its popularity with MS-DOS users. TCP/IP existed at this point, but it was typically only used by large government and research facilities.

As the Internet grew in popularity and a larger percentage of traffic became Internet-related, LANs and WANs gradually moved towards the TCP/IP protocols, and today networks mostly dedicated to TCP/IP traffic are common. The move to TCP/IP was helped by technologies such as DHCP that allowed TCP/IP clients to discover their own network address — a function that came standard with the AppleTalk/ IPX/ NetBIOS protocol sets.

It is at the data-link layer, though, that most modern LANs diverge from the Internet. Whereas Asynchronous Transfer Mode (ATM) or Multiprotocol Label Switching (MPLS) are typical data-link protocols for larger networks such as WANs; Ethernet and Token Ring are typical data-link protocols for LANs. These protocols differ from the former protocols in that they are simpler (e.g. they omit features such as Quality of Service guarantees) and offer collision prevention. Both of these differences allow for more economical systems. Despite the modest popularity of IBM token ring in the 1980s and 90's, virtually all LANs now use either wired or wireless Ethernets. At the physical layer, most wired Ethernet implementations use copper twisted-pair cables (including the common 10BASE-T networks). However, some early implementations used heavier coaxial cables and some recent implementations (especially high-speed ones) use optical fibers. When optic fibers are used, the distinction must be made between multimode fibers and single-mode fiberes. Multimode fibers can be thought of as thicker optical fibers that are cheaper to manufacture devices for but that suffers from less usable bandwidth and worse attenuation - implying poorer long-distance performance.

Chapter- 2

Wireless Network

Wireless network refers to any type of computer network that is not connected by cables of any kind. It is a method by which telecommunications networks and enterpise (business), installations avoid the costly process of introducing cables into to a building, or as a connection between various equipment locations. Wireless telecommunications networks are generally implemented and administered using a transmission system called radio waves. This implementation takes place at the physical level, (layer), of the network structure.

Types of wireless connections

Wireless PAN

Wireless Personal Area Networks (WPANs) interconnect devices within a relatively small area, generally within a persons reach. For example, both Bluetooth radio and invisible Infrared light provides a WPAN for interconnecting a headset to a laptop. ZigBee also supports WPAN applications. Wi-Fi PANs are becoming commonplace (2010) as equipment designers start to integrate Wi-Fi into a variety of consumer electronic devices. Intel "My WiFi" and Windows 7 "virtual Wi-Fi" capabilities have made Wi-Fi PANs simpler and easier to set up and configure.

Wireless LAN

A wireless local area network (WLAN) links two or more devices using a wireless distribution method, providing a connection through an access point to the wider internet. The use of spread-spectrum or OFDM technologies also gives users the mobility to move around within a local coverage area, and still remain connected to the network.

- Wi-Fi: "Wi-Fi" is a term used to describe 802.11 WLANs, although it is technically a declared standard of interoperability between 802.11 devices.
- Fixed Wireless Data: This implements point to point links between computers or networks at two distant locations, often using dedicated microwave or modulated laser light beams over line of sight paths. It is often used in cities to connect networks in two or more buildings without installing a wired link.

Wireless MAN

Wireless Metropolitan Area Networks are a type of wireless network that connects several Wireless LANs.

- WiMAX is a type of Wireless MAN and is described by the IEEE 802.16 standard.

Wireless WAN

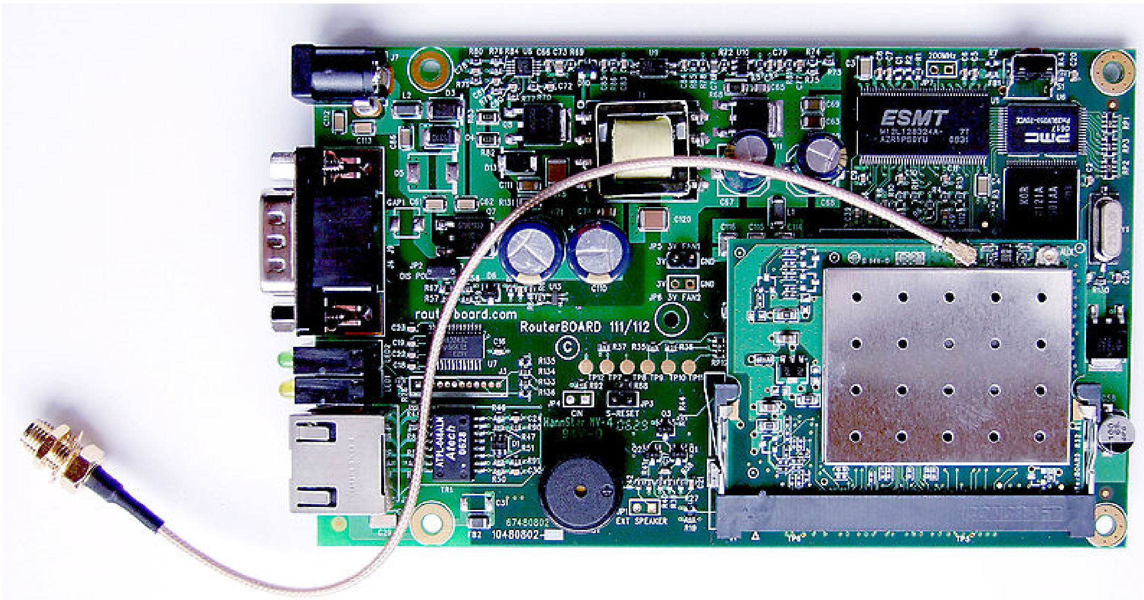
wireless wide area networks are wireless networks that typically cover large outdoor areas. These networks can be used to connect branch offices of business or as a public internet access system. They are usually deployed on the 2.4 GHz band. A typical system contains base station gateways, access points and wireless bridging relays. Other configurations are mesh systems where each access point acts as a relay also. When combined with renewable energy systems such as photo-voltaic solar panels or wind systems they can be stand alone systems.

Mobile devices networks

With the development of smart phones, cellular telephone networks routinely carry data in addition to telephone conversations:

- Global System for Mobile Communications (GSM): The GSM network is divided into three major systems: the switching system, the base station system, and the operation and support system. The cell phone connects to the base system station which then connects to the operation and support station; it then connects to the switching station where the call is transferred to where it needs to go. GSM is the most common standard and is used for a majority of cell phones.
- Personal Communications Service (PCS): PCS is a radio band that can be used by mobile phones in North America and South Asia. Sprint happened to be the first service to set up a PCS.
- D-AMPS: Digital Advanced Mobile Phone Service, an upgraded version of AMPS, is being phased out due to advancement in technology. The newer GSM networks are replacing the older system.

Uses



An embedded RouterBoard 112 with U.FL-RSMA pigtail and R52 mini PCI Wi-Fi card widely used by wireless Internet service providers (WISPs) in the Czech Republic.

Wireless networks continue to develop, usage has grown in 2010. Cellular phones are part of everyday wireless networks, allowing easy personal communications. Inter-continental network systems use radio satellites to communicate across the world. Emergency services such as the police utilize wireless networks to communicate effectively. Individuals and businesses use wireless networks to send and share data rapidly, whether it be in a small office building or across the world.

Another use for wireless networks is a cost effective means to connect to the Internet, in regions where the telecommunications infrastructure is both poor and lacking in resources, typically in rural areas and developing countries.

Compatibility issues also arise when dealing with wireless networks. Different devices may have compatibility issues, or might require modifications to solve these issues. Wireless networks are often typically slower than those found in modern versions of Ethernet cable connected installations.

A wireless network is more vulnerable, because anyone can intercept and sometimes divert a network broadcasting signal when point to point connections are used. Many wireless networks use WEP - Wired Equivalent Privacy - security systems. These have been found to be still vulnerable to intrusion. Though WEP does block some intruders, the security problems have caused some businesses to continue using wired networks until a more suitable security system can be introduced. The use of suitable firewalls overcome some security problems in wireless networks that are vulnerable to attempted unauthorised access.

Environmental concerns and health hazard

Starting around 2009, there have been increased concerns about the safety of wireless communications, despite little evidence of health risks so far. The president of Lakehead University refused to agree to installation of a wireless network citing a California Public Utilities Commission study which said that the possible risk of tumors and other diseases due to exposure to electromagnetic fields (EMFs) needs to be further investigated.

Chapter- 3

Modulation

In electronics, **modulation** is the process of varying one or more properties of a high-frequency periodic waveform, called the *carrier signal*, with respect to a *modulating signal*. This is done in a similar fashion as a musician may modulate a tone (a periodic waveform) from a musical instrument by varying its volume, timing and pitch. The three key parameters of a periodic waveform are its amplitude ("volume"), its phase ("timing") and its frequency ("pitch"), all of which can be modified in accordance with a low frequency signal to obtain the modulated signal. Typically a high-frequency sinusoid waveform is used as carrier signal, but a square wave pulse train may also occur.

In telecommunications, modulation is the process of conveying a message signal, for example a digital bit stream or an analog audio signal, inside another signal that can be physically transmitted. Modulation of a sine waveform is used to transform a baseband message signal into a passband signal, for example low-frequency audio signal into a radio-frequency signal (RF signal). In radio communications, cable TV systems or the public switched telephone network for instance, electrical signals can only be transferred over a limited passband frequency spectrum, with specific (non-zero) lower and upper cutoff frequencies. Modulating a sine-wave carrier makes it possible to keep the frequency content of the transferred signal as close as possible to the centre frequency (typically the carrier frequency) of the passband.

A device that performs modulation is known as a modulator and a device that performs the inverse operation of modulation is known as a demodulator (sometimes *detector* or *demod*). A device that can do both operations is a modem (modulator–demodulator).

Aim

The aim of **digital modulation** is to transfer a digital bit stream over an analog bandpass channel, for example over the public switched telephone network (where a bandpass filter limits the frequency range to between 300 and 3400 Hz), or over a limited radio frequency band.

The aim of **analog modulation** is to transfer an analog baseband (or lowpass) signal, for example an audio signal or TV signal, over an analog bandpass channel, for example a limited radio frequency band or a cable TV network channel.

Analog and digital modulation facilitate frequency division multiplexing (FDM), where several low pass information signals are transferred simultaneously over the same shared physical medium, using separate passband channels.

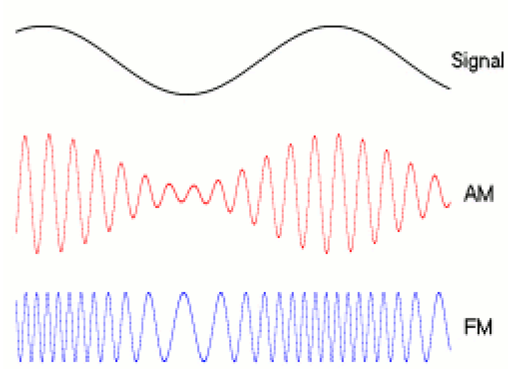
The aim of **digital baseband modulation** methods, also known as line coding, is to transfer a digital bit stream over a baseband channel, typically a non-filtered copper wire such as a serial bus or a wired local area network.

The aim of **pulse modulation** methods is to transfer a narrowband analog signal, for example a phone call over a wideband baseband channel or, in some of the schemes, as a bit stream over another digital transmission system.

In music synthesizers, modulation may be used to synthesise waveforms with a desired overtone spectrum. In this case the carrier frequency is typically in the same order or much lower than the modulating waveform.

Analog modulation methods

In analog modulation, the modulation is applied continuously in response to the analog information signal.



A low-frequency message signal (top) may be carried by an AM or FM radio wave.

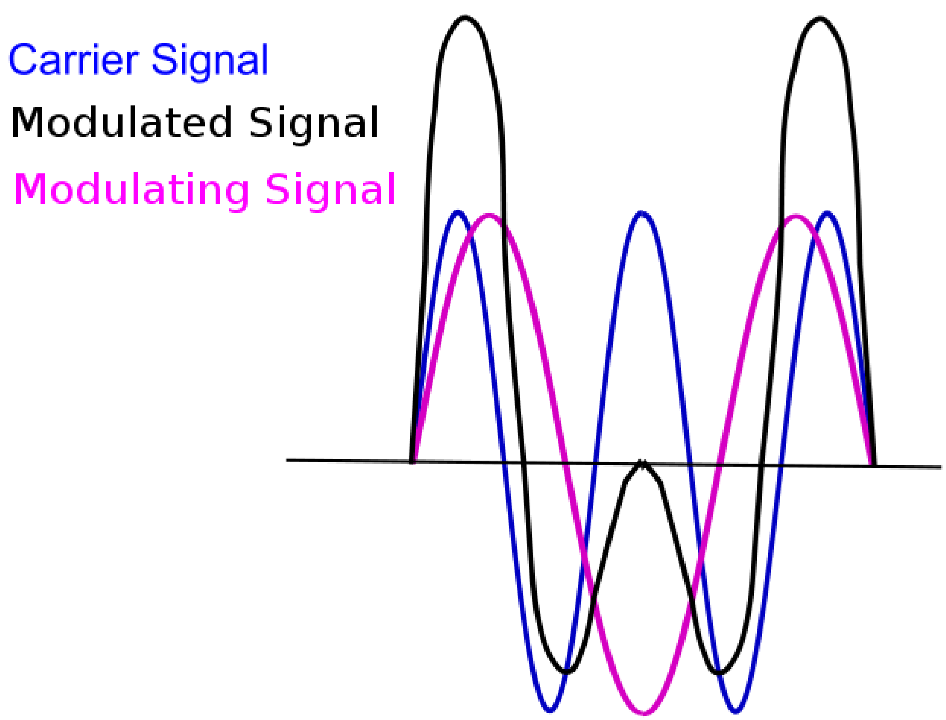
Common analog modulation techniques are:

- Amplitude modulation (AM) (here the amplitude of the carrier signal is varied in accordance to the instantaneous amplitude of the modulating signal)
    - Double-sideband modulation (DSB)

        - Double-sideband modulation with carrier (DSB-WC) (used on the AM radio broadcasting band)
        - Double-sideband suppressed-carrier transmission (DSB-SC)
        - Double-sideband reduced carrier transmission (DSB-RC)
    - Single-sideband modulation (SSB, or SSB-AM),
        - SSB with carrier (SSB-WC)
        - SSB suppressed carrier modulation (SSB-SC)
    - Vestigial sideband modulation (VSB, or VSB-AM)
    - Quadrature amplitude modulation (QAM)

- Angle modulation
    - Frequency modulation (FM) (here the frequency of the carrier signal is varied in accordance to the instantaneous amplitude of the modulating signal)
    - Phase modulation (PM) (here the phase shift of the carrier signal is varied in accordance to the instantaneous amplitude of the modulating signal)

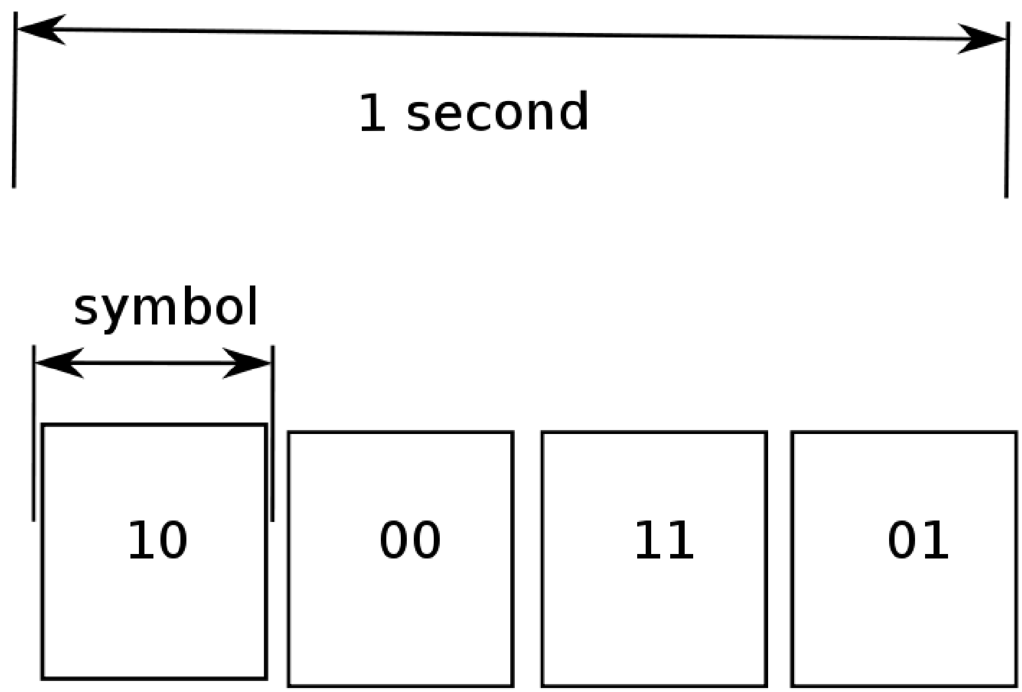
The accompanying figure shows the results of (amplitude-)modulating a signal onto a carrier (both of which are sine waves). At any point along the y-axis, the amplitude of the modulated signal is equal to the sum of the carrier signal and the modulating signal amplitudes.



Simple example of amplitude modulation.

Digital modulation methods

In digital modulation, an analog carrier signal is modulated by a digital bit stream. Digital modulation methods can be considered as digital-to-analog conversion, and the corresponding demodulation or detection as analog-to-digital conversion. The changes in the carrier signal are chosen from a finite number of M alternative symbols (the *modulation alphabet*).



Schematic of 4 baud (8 bps) data link.

A simple example: A telephone line is designed for transferring audible sounds, for example tones, and not digital bits (zeros and ones). Computers may however communicate over a telephone line by means of modems, which are representing the digital bits by tones, called symbols. If there are four alternative symbols (corresponding to a musical instrument that can generate four different tones, one at a time), the first symbol may represent the bit sequence 00, the second 01, the third 10 and the fourth 11. If the modem plays a melody consisting of 1000 tones per second, the symbol rate is 1000 symbols/second, or baud. Since each tone (i.e., symbol) represents a message consisting of two digital bits in this example, the bit rate is twice the symbol rate, i.e.

2000 bits per second. This is similar to the technique used by dialup modems as opposed to DSL modems.

According to one definition of digital signal, the modulated signal is a digital signal, and according to another definition, the modulation is a form of digital-to-analog conversion. Most textbooks would consider digital modulation schemes as a form of digital transmission, synonymous to data transmission; very few would consider it as analog transmission.

Fundamental digital modulation methods

The most fundamental digital modulation techniques are based on keying:

- In the case of PSK (phase-shift keying), a finite number of phases are used.
- In the case of FSK (frequency-shift keying), a finite number of frequencies are used.
- In the case of ASK (amplitude-shift keying), a finite number of amplitudes are used.
- In the case of QAM (quadrature amplitude modulation), a finite number of at least two phases, and at least two amplitudes are used.

In QAM, an inphase signal (the I signal, for example a cosine waveform) and a quadrature phase signal (the Q signal, for example a sine wave) are amplitude modulated with a finite number of amplitudes, and summed. It can be seen as a two-channel system, each channel using ASK. The resulting signal is equivalent to a combination of PSK and ASK.

In all of the above methods, each of these phases, frequencies or amplitudes are assigned a unique pattern of binary bits. Usually, each phase, frequency or amplitude encodes an equal number of bits. This number of bits comprises the *symbol* that is represented by the particular phase, frequency or amplitude.

If the alphabet consists of $M = 2^N$ alternative symbols, each symbol represents a message consisting of N bits. If the symbol rate (also known as the baud rate) is f_S symbols/second (or baud), the data rate is Nf_S bit/second.

For example, with an alphabet consisting of 16 alternative symbols, each symbol represents 4 bits. Thus, the data rate is four times the baud rate.

In the case of PSK, ASK or QAM, where the carrier frequency of the modulated signal is constant, the modulation alphabet is often conveniently represented on a constellation diagram, showing the amplitude of the I signal at the x-axis, and the amplitude of the Q signal at the y-axis, for each symbol.

Modulator and detector principles of operation

PSK and ASK, and sometimes also FSK, are often generated and detected using the principle of QAM. The I and Q signals can be combined into a complex-valued signal $I+jQ$ (where j is the imaginary unit). The resulting so called equivalent lowpass signal or equivalent baseband signal is a complex-valued representation of the real-valued modulated physical signal (the so called passband signal or RF signal).

These are the general steps used by the modulator to transmit data:

1. Group the incoming data bits into codewords, one for each symbol that will be transmitted.
2. Map the codewords to attributes, for example amplitudes of the I and Q signals (the equivalent low pass signal), or frequency or phase values.
3. Adapt pulse shaping or some other filtering to limit the bandwidth and form the spectrum of the equivalent low pass signal, typically using digital signal processing.
4. Perform digital-to-analog conversion (DAC) of the I and Q signals (since today all of the above is normally achieved using digital signal processing, DSP).
5. Generate a high-frequency sine wave carrier waveform, and perhaps also a cosine quadrature component. Carry out the modulation, for example by multiplying the sine and cosine wave form with the I and Q signals, resulting in that the equivalent low pass signal is frequency shifted into a modulated passband signal or RF signal. Sometimes this is achieved using DSP technology, for example direct digital synthesis using a waveform table, instead of analog signal processing. In that case the above DAC step should be done after this step.
6. Amplification and analog bandpass filtering to avoid harmonic distortion and periodic spectrum

At the receiver side, the demodulator typically performs:

1. Bandpass filtering.
2. Automatic gain control, AGC (to compensate for attenuation, for example fading).
3. Frequency shifting of the RF signal to the equivalent baseband I and Q signals, or to an intermediate frequency (IF) signal, by multiplying the RF signal with a local oscillator sinewave and cosine wave frequency.
4. Sampling and analog-to-digital conversion (ADC) (Sometimes before or instead of the above point, for example by means of undersampling).
5. Equalization filtering, for example a matched filter, compensation for multipath propagation, time spreading, phase distortion and frequency selective fading, to avoid intersymbol interference and symbol distortion.
6. Detection of the amplitudes of the I and Q signals, or the frequency or phase of the IF signal.
7. Quantization of the amplitudes, frequencies or phases to the nearest allowed symbol values.

8. Mapping of the quantized amplitudes, frequencies or phases to codewords (bit groups).
9. Parallel-to-serial conversion of the codewords into a bit stream.
10. Pass the resultant bit stream on for further processing such as removal of any error-correcting codes.

As is common to all digital communication systems, the design of both the modulator and demodulator must be done simultaneously. Digital modulation schemes are possible because the transmitter-receiver pair have prior knowledge of how data is encoded and represented in the communications system. In all digital communication systems, both the modulator at the transmitter and the demodulator at the receiver are structured so that they perform inverse operations.

Non-coherent modulation methods do not require a receiver reference clock signal that is phase synchronized with the sender carrier wave. In this case, modulation symbols (rather than bits, characters, or data packets) are asynchronously transferred. The opposite is coherent modulation.

List of common digital modulation techniques

The most common digital modulation techniques are:

- Phase-shift keying (PSK):
  - Binary PSK (BPSK), using M=2 symbols
  - Quadrature PSK (QPSK), using M=4 symbols
  - 8PSK, using M=8 symbols
  - 16PSK, using M=16 symbols
  - Differential PSK (DPSK)
  - Differential QPSK (DQPSK)
  - Offset QPSK (OQPSK)
  - π/4–QPSK
- Frequency-shift keying (FSK):
  - Audio frequency-shift keying (AFSK)
  - Multi-frequency shift keying (M-ary FSK or MFSK)
  - Dual-tone multi-frequency (DTMF)
  - Continuous-phase frequency-shift keying (CPFSK)
- Amplitude-shift keying (ASK)
- On-off keying (OOK), the most common ASK form
  - M-ary vestigial sideband modulation, for example 8VSB
- Quadrature amplitude modulation (QAM) - a combination of PSK and ASK:
  - Polar modulation like QAM a combination of PSK and ASK.
- Continuous phase modulation (CPM) methods:
  - Minimum-shift keying (MSK)
  - Gaussian minimum-shift keying (GMSK)
- Orthogonal frequency-division multiplexing (OFDM) modulation:
  - discrete multitone (DMT) - including adaptive modulation and bit-loading.

- Wavelet modulation
- Trellis coded modulation (TCM), also known as trellis modulation
- Spread-spectrum techniques:
    - Direct-sequence spread spectrum (DSSS)
    - Chirp spread spectrum (CSS) according to IEEE 802.15.4a CSS uses pseudo-stochastic coding
    - Frequency-hopping spread spectrum (FHSS) applies a special scheme for channel release

MSK and GMSK are particular cases of continuous phase modulation. Indeed, MSK is a particular case of the sub-family of CPM known as continuous-phase frequency-shift keying (CPFSK) which is defined by a rectangular frequency pulse (i.e. a linearly increasing phase pulse) of one symbol-time duration (total response signaling).

OFDM is based on the idea of frequency-division multiplexing (FDM), but is utilized as a digital modulation scheme. The bit stream is split into several parallel data streams, each transferred over its own sub-carrier using some conventional digital modulation scheme. The modulated sub-carriers are summed to form an OFDM signal. OFDM is considered as a modulation technique rather than a multiplex technique, since it transfers one bit stream over one communication channel using one sequence of so-called OFDM symbols. OFDM can be extended to multi-user channel access method in the orthogonal frequency-division multiple access (OFDMA) and multi-carrier code division multiple access (MC-CDMA) schemes, allowing several users to share the same physical medium by giving different sub-carriers or spreading codes to different users.

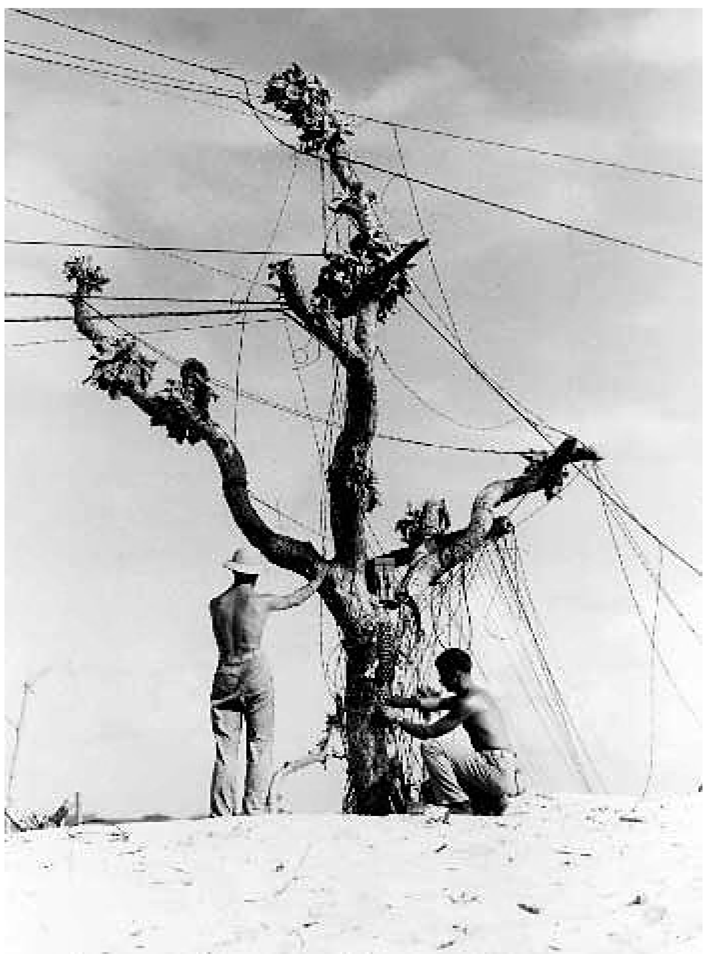
Of the two kinds of RF power amplifier, switching amplifiers (Class C amplifiers) cost less and use less battery power than linear amplifiers of the same output power. However, they only work with relatively constant-amplitude-modulation signals such as angle modulation (FSK or PSK) and CDMA, but not with QAM and OFDM. Nevertheless, even though switching amplifiers are completely unsuitable for normal QAM constellations, often the QAM modulation principle are used to drive switching amplifiers with these FM and other waveforms, and sometimes QAM demodulators are used to receive the signals put out by these switching amplifiers.

Digital baseband modulation or line coding

The term **digital baseband modulation** (or digital baseband transmission) is synonymous to line codes. These are methods to transfer a digital bit stream over an analog baseband channel (a.k.a. lowpass channel) using a pulse train, i.e. a discrete number of signal levels, by directly modulating the voltage or current on a cable. Common examples are unipolar, non-return-to-zero (NRZ), Manchester and alternate mark inversion (AMI) codings.

Chapter- 4

Channel



Old telephone wires are a challenging communications channel for modern digital communications.

In telecommunications and computer networking, a **communication channel**, or **channel**, refers either to a physical transmission medium such as a wire, or to a logical connection over a multiplexed medium such as a radio channel. A channel is used to convey an information signal, for example a digital bit stream, from one or several *senders* (or transmitters) to one or several *receivers*. A channel has a certain capacity for transmitting information, often measured by its bandwidth in Hz or its data rate in bits per second

In information theory, a channel refers to a theoretical *channel model* with certain error characteristics. In this more general view, a storage device is also a kind of channel, which can be sent to (written) and received from (read).

Examples

A channel can take many forms. Examples of communications channels include:

1. A connection between initiating and terminating nodes of a circuit.
2. A single path provided by a transmission medium via either
    - physical separation, such as by multipair cable or
    - electrical separation, such as by frequency-division or time-division multiplexing.
3. A path for conveying electrical or electromagnetic signals, usually distinguished from other parallel paths.
    - A storage which can communicate a message over time as well as space
    - The portion of a storage medium, such as a track or a band, that is accessible to a given reading or writing station or head.
    - A buffer from which messages can be 'put' and 'got'.
4. In a communications system, the physical or logical link that connects a data source to a data sink.
5. A specific radio frequency, pair or band of frequencies, usually named with a letter, number, or codeword, and often allocated by international agreement. Examples:
    - Marine VHF radio uses some 88 channels in the VHF band for two-way FM voice communication. Channel 16, for example, is 156.800 MHz. In the US, seven additional channels, WX1 - WX7, are allocated for weather broadcasts.
    - Television channels such as North American TV Channel 2 = 55.25 MHz, Channel 13 = 211.25 MHz. Each channel is 6 MHz wide. Besides these "physical channels", television also has "virtual channels".
    - Wi-Fi consists of unlicensed channels 1-13 from 2412 MHz to 2484 MHz in 5 MHz steps.
    - The radio channel between an amateur radio repeater and a ham uses two bands often 600 kHz (0.6 MHz) apart. For example, a repeater that transmits on 146.94 MHz typically listens for a ham transmitting on 146.34 MHz.

6. A room in the Internet Relay Chat (IRC) network, in which participants can communicate with each other.

All of these communications channels share the property that they transfer information. The information is carried through the channel by a signal.

Channel models

A channel can be modelled physically by trying to calculate the physical processes which modify the transmitted signal. For example in wireless communications the channel can be modelled by calculating the reflection off every object in the environment. A sequence of random numbers might also be added in to simulate external interference and/or electronic noise in the receiver.

Statistically a communication channel is usually modelled as a triple consisting of an input alphabet, an output alphabet, and for each pair *(i, o)* of input and output elements a transition probability *p(i, o)*. Semantically, the transition probability is the probability that the symbol *o* is received given that *i* was transmitted over the channel.

Statistical and physical modelling can be combined. For example in wireless communications the channel is often modelled by a random attenuation (known as fading) of the transmitted signal, followed by additive noise. The attenuation term is a simplification of the underlying physical processes and captures the change in signal power over the course of the transmission. The noise in the model captures external interference and/or electronic noise in the receiver. If the attenuation term is complex it also describes the relative time a signal takes to get through the channel. The statistics of the random attenuation are decided by previous measurements or physical simulations.

Channel models may be continuous channel models in that there is no limit to how precisely their values may be defined.

Communication channels are also studied in a discrete-alphabet setting. This corresponds to abstracting a real world communication system in which the analog->digital and digital->analog blocks are out of the control of the designer. The mathematical model consists of a transition probability that specifies an output distribution for each possible sequence of channel inputs. In information theory, it is common to start with memoryless channels in which the output probability distribution only depends on the current channel input.

A channel model may either be digital (quantified, e.g. binary) or analog.

Digital channel models

In a digital channel model, the transmitted message is modelled as a digital signal at a certain protocol layer. Underlying protocol layers, such as the physical layer transmission technique, is replaced by a simplified model. The model may reflect channel performance measures such as bit rate, bit errors, latency/delay, delay jitter, etc. Examples of digital channel models are:

- Binary symmetric channel (BSC), a discrete memoryless channel with a certain bit error probability
- Binary bursty bit error channel model, a channel "with memory"
- Binary erasure channel (BEC), a discrete channel with a certain bit error detection (erasure) probability
- Packet erasure channel, where packets are lost with a certain packet loss probability or packet error rate
- Arbitrarily varying channel (AVC), where the behavior and state of the channel can change randomly

Analog channel models

In an analog channel model, the transmitted message is modelled as an analog signal. The model can be a linear or non-linear, time-continuous or time-discrete (sampled), memoryless or dynamic (resulting in burst errors), time-invariant or time-variant (also resulting in burst errors), baseband, passband (RF signal model), real-valued or complex-valued signal model. The model may reflect the following channel impairments:

- Noise model, for example
    - Additive white Gaussian noise (AWGN) channel, a linear continuous memoryless model
    - Phase noise model
- Interference model, for example cross-talk (co-channel interference) and intersymbol interference (ISI)
- Distortion model, for example a non-linear channel model causing intermodulation distortion (IMD)
- Frequency response model, including attenuation and phase-shift
- Group delay model
- Modelling of underlying physical layer transmission techniques, for example a complex-valued equivalent baseband model of modulation and frequency response
- Radio frequency propagation model, for example
    - Log-distance path loss model
    - Fading model, for example Rayleigh fading, Ricean fading, log-normal shadow fading and frequency selective (dispersive) fading
    - Doppler shift model, which combined with fading results in a time-variant system
    - Ray tracing models, which attempt to model the signal propagation and distortions for specified transmitter-receiver geometries, terrain types, and antennas
    - Mobility models, which also causes a time-variant system

Types of communications channels

- Digital (discrete) or analog (continuous) channel
- Baseband and passband channel
- Transmission medium, for example a fibre channel
- Multiplexed channel
- Computer network virtual channel
- Simplex communication, duplex communication or half duplex communication channel
- Return channel
- Uplink or downlink (upstream or downstream channel)
- Broadcast channel, unicast channel or multicast channel

Multi-terminal channels, with application to cellular systems

In networks, as opposed to point-to-point communication, the communication media is shared between multiple nodes (terminals). Depending on the type of communication, different terminals can cooperate or interfere on each other. In general, any complex multi-terminal network can be considered as a combination of simplified multi-terminal channels. The following channels are the principal multi-terminal channels which was first introduced in the field of information theory:

- A point-to-multipoint channel, also known as broadcasting medium (not to be confused with broadcasting channel): In this channel, a single sender transmits multiple messages to different destination nodes. All wireless channels except radio links can be considered as broadcasting media, but may not always provide broadcasting service. The downlink of a cellular system can be considered as a point-to-multipoint channel, if only one cell is considered and inter-cell co-channel interference is neglected. However, the communication service of a phone call is unicasting.
- Multiple access channel: In this channel, multiple senders transmit multiple possible different messages over a shared physical medium to one or several destination nodes. This requires a channel access scheme, including a media access control (MAC) protocol combiend with a multiplexing scheme. This channel model has applications in the uplink of the cellular networks.
- Relay channel: In this channel, one or several intermediate nodes (called relay, repeater or gap filler nodes) cooperate with a sender to send the message to an ultimate destination node. Relay nodes are considered as a possible add-on in the upcoming cellular standards like 3GPP Long Term Evolution (LTE).
- Interference channel: In this channel, two different senders transmit their data to different destination nodes. Hence, the different senders can have a possible cross-talk or co-channel interference on the signal of each other. The inter-cell interference in the cellular wireless communications is an example of the

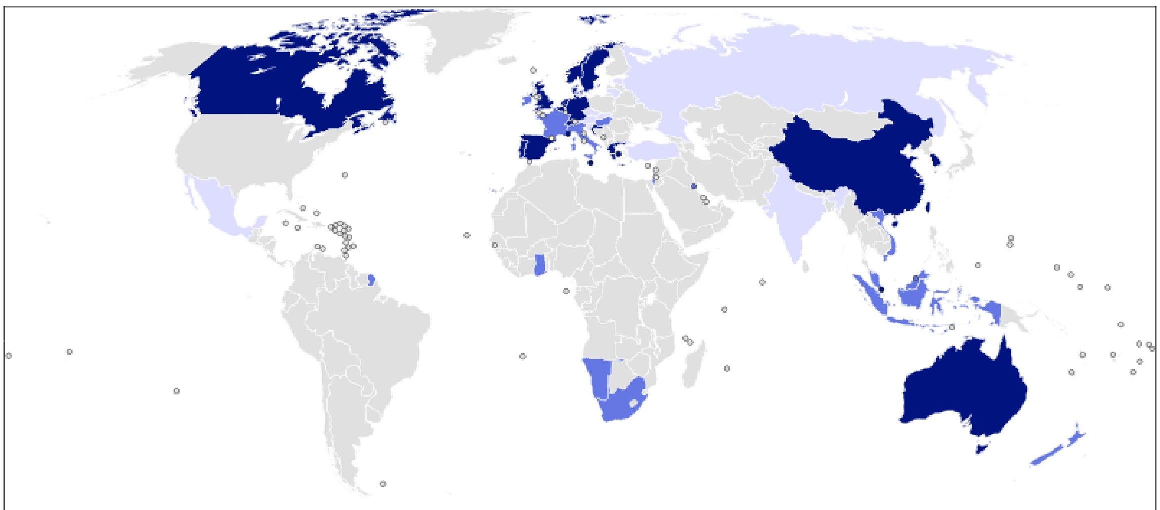
interference channel. In spread spectrum systems like 3G, interference also occur inside the cell if non-orthogonal codes are used.

- A unicasting channel is a channel that provides a unicasting service, i.e. that sends data addressed to one specific user. An established phone call is an example.
- A broadcasting channel is a channel that provides a broadcasting service, i.e. that sends data addressed to all users in the network. Cellular network examples are the paging service as well as the Multimedia Broadcast Multicast Service.
- A multicasting channel is a channel where data is addressed to a group of subscribing users. LTE exampels are the Physical Multicast Channel (PMCH) and MBSFN (Multicast Broadcast Single Frequency Network).

From the above 4 basic multi-terminal channels, multiple access channel is the only one whose capacity region is known. Even for the special case of the Gaussian scenario, the capacity region of the other 3 channels except the broadcast channel is unknown in general.

Chapter- 5

Digital Audio Broadcasting



Countries with regular services
Countries with trials / tests
Countries with interest
DAB no longer used

Digital Audio Broadcasting (**DAB**), is a digital radio technology for broadcasting radio stations, used in several countries, particularly in Europe. As of 2006, approximately 1,000 stations worldwide broadcast in the DAB format.

The DAB standard was initiated as a European research project in the 1980s, and the BBC launched the first DAB digital radio in 1995. DAB receivers have been available in many countries since the end of the nineties. DAB may offer more radio programmes over a specific spectrum than analogue FM radio. DAB is more robust with regard to noise and multipath fading for mobile listening, since DAB reception quality first degrades rapidly when the signal strength falls below a critical threshold, whereas FM reception quality degrades slowly with the decreasing signal.

An "informal listening test" by Professor Sverre Holm has shown that for stationary listening the audio quality on DAB is lower than FM stereo, due to most stations using a bit rate of 128 kbit/s or less, with the MP2 audio codec, which requires 160 kbit/s to achieve perceived FM quality. 128 kbit/s gives better dynamic range or signal-to-noise ratio than FM radio, but a more smeared stereo image, and an upper cutoff frequency of 14 kHz, corresponding to 15 kHz of FM radio. However, "CD sound quality" with MP2 is possible "with 256..192 kbps".

An upgraded version of the system was released in February 2007, which is called **DAB+**. DAB is not forward compatible with DAB+, which means that DAB-only receivers will not be able to receive DAB+ broadcasts. DAB+ is approximately twice as efficient as DAB due to the adoption of the AAC+ audio codec, and DAB+ can provide high quality audio with as low as 64kbit/s. Reception quality will also be more robust on DAB+ than on DAB due to the addition of Reed-Solomon error correction coding.

More than 20 countries provide DAB transmissions, and several countries, such as Australia, Italy, Malta and Switzerland, have started transmitting DAB+ stations.

History

DAB has been under development since 1981 at the Institut für Rundfunktechnik (IRT). In 1985 the first DAB demonstrations were held at the WARC-ORB in Geneva and in 1988 the first DAB transmissions were made in Germany. Later DAB was developed as a research project for the European Union (EUREKA), which started in 1987 on initiative by a consortium formed in 1986. The MPEG-1 Audio Layer II ("MP2") codec was created as part of the EU147 project. DAB was the first standard based on orthogonal frequency division multiplexing (OFDM) modulation technique, which since then has become one of the most popular transmission schemes for modern wideband digital communication systems.

A choice of audio codec, modulation and error-correction coding schemes and first trial broadcasts were made in 1990. Public demonstrations were made in 1993 in the United Kingdom. The protocol specification was finalized in 1993 and adopted by the ITU-R standardization body in 1994, the European community in 1995 and by ETSI in 1997. Pilot broadcasts were launched in several countries in 1995.

The UK was the first country to receive a wide range of radio stations via DAB. Commercial DAB receivers began to be sold in 1999 and over 50 commercial and BBC services were available in London by 2001.

By 2006, 500 million people worldwide were in the coverage area of DAB broadcasts, although by this time sales had only taken off in the UK and Denmark. In 2006 there were approximately 1,000 DAB stations in operation world wide.

The standard was coordinated by the European DAB forum, formed in 1995 and reconstituted to the World DAB Forum in 1997, which represents more than 30 countries.

In 2006 the World DAB Forum became the World DMB Forum which now presides over both the DAB and DMB standard.

In October 2005, the World DMB Forum instructed its Technical Committee to carry out the work needed to adopt the AAC+ audio codec and stronger error correction coding. This work led to the launch of the new DAB+ system.

DAB and FM/AM compared

Traditionally radio programmes were broadcast on different frequencies via FM and AM, and the radio had to be tuned into each frequency, as needed. This used up a comparatively large amount of spectrum for a relatively small number of stations, limiting listening choice. DAB is a digital radio broadcasting system that through the application of multiplexing and compression combines multiple audio streams onto a relatively narrow band centred on a single broadcast frequency called a DAB ensemble.

Within an overall target bit rate for the DAB ensemble, individual stations can be allocated different bit rates. The number of channels within a DAB ensemble can be increased by lowering average bit rates, but at the expense of the quality of streams. Error correction under the DAB standard makes the signal more robust but reduces the total bit rate available for streams.

Use of frequency spectrum and transmitter sites

DAB gives substantially higher spectral efficiency, measured in programmes per MHz and per transmitter site, than analogue communication. This has led to an increase in the number of stations available to listeners, especially outside of the major urban areas.

Numerical example: Analog FM requires 0.2 MHz per programme. The frequency reuse factor in most countries is approximately 15, meaning that only one out of 15 transmitter sites can use the same channel frequency without problems with co-channel interference, i.e. cross-talk. Assuming a total availability of 102 FM channels at a bandwidth of 0.2MHz over the Band II spectrum of 87.5 to 108.0 MHz, an average of 102/15 = 6.8 radio channels are possible on each transmitter site (plus lower-power local transmitters causing less interference). This results in a system spectral efficiency of 1 / 15 / (0.2 MHz) = 0.30 programmes/transmitter/MHz. DAB with 192 kbit/s codec requires 1.536 MHz * 192 kbit/s / 1136 kbit/s = 0.26 MHz per audio programme. The frequency reuse factor for local programmes and multi-frequency broadcasting networks (MFN) is typically 4 or 5, resulting in 1 / 4 / (0.26 MHz) = 0.96 programmes/transmitter/MHz. This is **3.2 times as efficient as analog FM for local stations**. For single frequency network (SFN) transmission, for example of national programmes, the channel re-use factor is 1, resulting in 1/1/0.25 MHz = 3.85 programmes/transmitter/MHz, which is **12.7 times as efficient as FM for national and regional networks**.

Note the above capacity improvement may not always be achieved at the L-band frequencies, since these are more sensitive to obstacles than the FM band frequencies, and may cause shadow fading for hilly terrain and for indoor communication. The number of transmitter sites or the transmission power required for full coverage of a country may be rather high at these frequencies, to avoid that the system becomes noise limited rather than limited by co-channel interference.

Sound quality

The original objectives of converting to digital transmission were to enable higher fidelity, more stations and more resistance to noise, co-channel interference and multipath than in analogue FM radio. However, the leading countries in implementing DAB on stereo radio stations use compression to such a degree that it produces lower sound quality than that received from non-mobile FM broadcasts. This is because of the bit rate levels being too low for the MPEG Layer 2 audio codec to provide high fidelity audio quality.

The BBC Research & Development department states that at least 192kbit/s is necessary for a high fidelity stereo broadcast :

> “A value of 256 kbit/s has been judged to provide a high quality stereo broadcast signal. However, a small reduction, to 224 kbit/s is often adequate, and in some cases it may be possible to accept a further reduction to 192 kbit/s, especially if redundancy in the stereo signal is exploited by a process of 'joint stereo' encoding (i.e. some sounds appearing at the centre of the stereo image need not be sent twice). At 192 kbit/s, it is relatively easy to hear imperfections in critical audio material.”
>
> — BBC R&D White Paper WHP 061 June 2003

When BBC in July 2006 reduced the bit-rate of transmission of Radio 3 from 192 kbit/s to 160 kbit/s, the resulting degradation of audio quality prompted a number of complaints to the Corporation. BBC later announced that following this testing of new equipment, it would resume the previous practice of transmitting Radio 3 at 192 kbit/s whenever there were no other demands on bandwidth.

Satisified listeners

Still, a survey of DAB listeners (including mobile) has shown most find DAB to have equal or better sound quality than FM.

Criticism

Broadcasters have been criticized for ‘squeezing in’ more stations per ensemble than recommended, by:

- Minimizing the bit-rate, to the lowest level of sound-quality that listeners are willing to tolerate, such as 128 kbit/s for stereo and even 64 kbit/s for mono speech radio.
- Having few digital channels broadcasting in stereo.

Benefits of DAB

Current AM and FM terrestrial broadcast technology is well established, compatible, and cheap to manufacture. Benefits of DAB over analogue systems are explained below.

Improved end-user features

DAB radios automatically tune to all the available stations, offering a list of all stations.

DAB can carry "radiotext" (in DAB terminology, *Dynamic Label Segment*, or DLS) from the station giving real-time information such as song titles, music type and news or traffic updates. Advance programme guides can also be transmitted. A similar feature also exists on FM in the form of the RDS. (However, not all FM receivers allow radio stations to be stored by name.)

Some radios offer a pause facility on live broadcasts, caching the broadcast stream on local flash memory, although this function is limited.

Lower cost

DAB broadcast many channels over one transmitter (multiplex) which lowers maintenance and transmission costs radically.

More stations

DAB is not more bandwidth efficient than analogue measured in programmes per MHz of a specific transmitter (the so called link spectral efficiency). However, it is more rubust to co-channel interference (cross talk), which makes it possible to reduce the reuse distance, i.e. use the same radio frequency channel more densely. The system spectral efficiency (the average number of radio programmes per MHz and transmitter) is a factor three more efficient than analog FM for local radio stations, as can be seen in the above numerical example. For national and regional radio networks, the efficiency is improved by more than an order of magnitude due to the use of SFNs. In that case, adjacent transmitters use the same frequency.

In certain areas — particularly rural areas — the introduction of DAB gives radio listeners a greater choice of radio stations. For instance, in South Norway, radio listeners experienced an increase in available stations from 6 to 21 when DAB was introduced in November 2006.

Reception quality

The DAB standard integrates features to reduce the negative consequences of multipath fading and signal noise, which afflict existing analogue systems.

Also, as DAB transmits digital audio, there is no hiss with a weak signal, which can happen on FM. However, radios in the fringe of a DAB signal, can experience a "bubbling mud" sound interrupting the audio and/or the audio cutting out altogether.

Due to sensitivity to doppler shift in combination with multipath propagation, DAB receivers can not operate in travelling speeds of more than 200 to 600 km/h depending on carrier frequency.

Less pirate interference

The specialised nature and cost of DAB broadcasting equipment provide barriers to pirate radio stations broadcasting on DAB. In cities such as London with large numbers of pirate radio stations broadcasting on FM, this means that some stations can be reliably received via DAB in areas where they are regularly difficult or impossible to receive on FM due to pirate radio interference.

Variable bandwidth

Mono talk radio, news and weather channels and other non-music programs need significantly less bandwidth than a typical music radio station, which allows DAB to carry these programmes at lower bit rates, leaving more bandwidth to be used for other programs.

Criticisms of DAB

Music radio stations broadcasting in mono

A number of music radio stations and stations that carry drama on DAB in the UK are being broadcast in mono. These stations are often available in stereo on other digital platforms, where capacity is not as constrained, and on FM where applicable.

Reception quality

The reception quality on DAB can be poor even for people that live well within the coverage area. The reason for this is that the old version of DAB uses weak error correction coding, so that when there are a lot of errors with the received data not enough of the errors can be corrected and a "bubbling mud" sound occurs. In some cases a complete loss of signal can happen. This situation will be improved upon in the new DAB standard (DAB+, discussed below) that uses stronger error correction coding and as additional transmitters are built.

Signal delay

The nature of a SFN is such that the transmitters in a network must broadcast the same signal at the same time. To achieve synchronization, the broadcaster must counter any differences in propagation time incurred by the different methods and distances involved in carrying the signal from the multiplexer to the different transmitters. This is done by applying a delay to the incoming signal at the transmitter based on a timestamp generated at the multiplexer, created taking into account the maximum likely propagation time, with a generous added margin for safety. Also delays in the receiver due to digital processing (e.g. deinterleaving) add to the overall delay to the listener. This delays the signal to the listener by about 2 seconds (depending on the decoding circuitry used). This has two disadvantages: (i) DAB radios are out of step with live events so time signals are not accurate and the experience of listening to live commentaries on events being watched is impaired, and (ii) listeners using a combination of FM and DAB radios (e.g. in different rooms of a house) will not hear an intelligible signal when both receivers are within earshot.

Coverage

As DAB is at a relatively early stage of deployment, DAB coverage is poor in nearly all countries in comparison to the high population coverage provided by FM.

Transmission costs

DAB has been criticized to be more expensive than FM. The truth of the matter is that FM networks and DAB networks bear virtually the same costs.

Facts - DAB uses higher frequencies than FM and must therefore compensate with more transmitters, higher radiated powers, or a combination, to achieve the same coverage. - The power efficiency is slightly better for today's FM-transmitters, meaning they will require less electricity than DAB-transmitter, for the same radiated power. However only the last couple of years has seen significant improvement in power efficiency for DAB-transmitters - All in all, a DAB network is slightly more expensive than a FM network - BUT, a DAB network can carry 6-10 channels (with MPEG audio codec) or 10-16 channels (with HE AAC codec) - Meaning that broadcasting a channel in DAB is far less expensive than broadcasting in FM.

This is backed by independent network studies from Teracom (Sweden) and SSR/SRG (Switzerland). Among other things they show that DAB is up to 6 times less expensive than FM.

Replacing FM-radios and FM-transmitters with new DAB-radios and DAB-transmitters will be no more costly than rebuilding the existing FM facilities.

Compatibility

In 2006 tests began using the much improved HE-AAC codec for DAB+. Virtually none of the receivers made before 2008 support the new codec, however, thus making them partially obsolete once DAB+ broadcasts begin and completely obsolete once the old MPEG-1 Layer 2 stations are switched off. However new receivers are both DAB and DAB+ compatible.

Power requirements

As DAB requires digital signal processing techniques to convert from the received digitally encoded signal to the analogue audio content, the complexity of the electronic circuitry required to do this is high. This translates into needing more power to effect this conversion than compared to an analogue FM to audio conversion, meaning that portable receiving equipment will tend to have a shorter battery life, or require higher power (and hence more bulk). This effectively means that they're less energy efficient than an analog Band II VHF receiver.

As an indicator of this increased power consumption, dual FM/DAB radios quote the length of time they can play on a single charge. For a commonly used FM/DAB-receiver from manufacturer PURE, this is stated as: DAB 10 hours, FM 22 hours.

Use of Licensed Codecs

The use of MPEG and latterly AAC has prompted criticism of the fact that a (large) public system is financially supporting a private company. In general, an open system will permit equipment to be bought from various sources in competition with each other but by selecting a single vendor of codec, with which all equipment must be compatible, this is not possible.

Technology

Bands and modes

DAB uses a wide-bandwidth broadcast technology and typically spectra have been allocated for it in Band III (174–240 MHz) and L band (1452–1492 MHz), although the scheme allows for operation almost anywhere above 30 MHz. The US military has reserved L-Band in the USA only, blocking its use for other purposes in America, and the United States has reached an agreement with Canada that the latter will restrict L-Band DAB to terrestrial broadcast to avoid interference.

DAB has a number of country specific transmission modes (I, II, III and IV). For worldwide operation a receiver must support all 4 modes:

- Mode I for Band III, Earth
- Mode II for L-Band, Earth and satellite

- Mode III for frequencies below 3 GHz, Earth and satellite
- Mode IV for L-Band, Earth and satellite

Protocol stack

From a OSI model protocol stack viewpoint, the technologies used on DAB inhabit the following layers: the audio codec inhabits the presentation layer. Below that is the data link layer, in charge of packet mode statistical multiplexing and frame synchronization. Finally, the physical layer contains the error-correction coding, OFDM modulation, and dealing with the over-the-air transmission and reception of data. Some aspects of these are described below.

Audio codec

The older version of DAB that is being used in the UK, Ireland, Denmark, Norway and Switzerland, uses the MPEG-1 Audio Layer 2 audio codec, which is also known as *MP2* due to computer files using those characters for their file extension.

The new DAB+ standard has adopted the HE-AAC version 2 audio codec, commonly known as *AAC+* or *aacPlus*. AAC+ is approximately three-times more efficient than MP2, which means that broadcasters using DAB+ will be able to provide far higher audio quality or far more stations than they can on DAB, or, as is most likely, a combination of both higher audio quality and more stations will be provided.

One of the most important decisions regarding the design of a digital radio system is the choice of which audio codec to use, because the efficiency of the audio codec determines how many radio stations can be carried on a multiplex at a given level of audio quality. The capacity of a DAB multiplex is fixed, so the more efficient the audio codec is, the more stations can be carried, and vice versa. Similarly, for a fixed bit-rate level, the more efficient the audio codec is the higher the audio quality will be.

Error-correction coding

Error-correction coding (ECC) is an important technology for a digital communication system because it determines how robust the reception will be for a given signal strength - stronger ECC will provide more robust reception than a weaker form.

The old version of DAB uses punctured convolutional coding for its ECC. The coding scheme uses unequal error protection (UEP), which means that parts of the audio bit-stream that are more susceptible to errors causing audible disturbances are provided with more protection (i.e. a lower code rate) and vice versa. However, the UEP scheme used on DAB results in there being a grey area in between the user experiencing good reception quality and no reception at all, as opposed to the situation with most other wireless digital communication systems that have a sharp "digital cliff", where the signal rapidly becomes unusable if the signal strength drops below a certain threshold. When

DAB listeners receive a signal in this intermediate strength area they experience a "burbling" sound which interrupts the playback of the audio.

The new DAB+ standard has incorporated Reed-Solomon ECC as an "inner layer" of coding that is placed around the byte interleaved audio frame but inside the "outer layer" of convolutional coding used by the older DAB system, although on DAB+ the convolutional coding uses equal error protection (EEP) rather than UEP since each bit is equally important in DAB+. This combination of Reed-Solomon coding as the inner layer of coding, followed by an outer layer of convolutional coding - so-called "concatenated coding" - became a popular ECC scheme in the 1990s, and NASA adopted it for its deep-space missions. One slight difference between the concatenated coding used by the DAB+ system and that used on most other systems is that it uses a rectangular byte interleaver rather than Forney interleaving in order to provide a greater interleaver depth, which increases the distance over which error bursts will be spread out in the bit-stream, which in turn will allow the Reed-Solomon error decoder to correct a higher proportion of errors.

The ECC used on DAB+ is far stronger than is used on DAB, which, with all else being equal (i.e. if the transmission powers remained the same), would translate into people who currently experience reception difficulties on DAB receiving a much more robust signal with DAB+ transmissions. It also has a far steeper "digital cliff", and listening tests have shown that people prefer this when the signal strength is low compared to the shallower digital cliff on DAB.

Modulation

Immunity to fading and inter-symbol interference (caused by multipath propagation) is achieved without equalization by means of the OFDM and DQPSK modulation techniques.

Using values for the most commonly used transmission mode on DAB, Transmission Mode I (TM I), the OFDM modulation consists of 1,536 subcarriers that are transmitted in parallel. The useful part of the OFDM symbol period is 1 millisecond, which results in the OFDM subcarriers each having a bandwidth of 1 kHz due to the inverse relationship between these two parameters, and the overall OFDM channel bandwidth is 1,537 kHz. The OFDM guard interval for TM I is 246 microseconds, which means that the overall OFDM symbol duration is 1.246 milliseconds. The guard interval duration also determines the maximum separation between transmitters that are part of the same single-frequency network (SFN), which is approximately 74 km for TM I.

Single-frequency networks

OFDM allows the use of single-frequency networks (SFN), which means that a network of transmitters can provide coverage to a large area - up to the size of a country - where all transmitters use the same transmission frequency. Transmitters that are part of an SFN need to be very accurately synchronised with other transmitters in the network, which requires the transmitters to use very accurate clocks.

When a receiver receives a signal that has been transmitted from the different transmitters that are part of an SFN, the signals from the different transmitters will typically have different delays, but to OFDM they will appear to simply be different multipaths of the same signal. Reception difficulties can arise, however, when the relative delay of multipaths exceeds the OFDM guard interval duration, and there are frequent reports of reception difficulties due to this issue when there is a *lift*, such as when there's high pressure, due to signals travelling farther than usual, and thus the signals are likely to arrive with a relative delay that is greater than the OFDM guard interval.

Low power *gap-filler* transmitters can be added to an SFN as and when desired in order to improve reception quality, although the way SFNs have been implemented in the UK up to now they have tended to consist of higher power transmitters being installed at main transmitter sites in order to keep costs down.

Bit rates

An ensemble has a maximum bit rate that can be carried, but this depends on which error protection level is used. However, all DAB multiplexes can carry a total of 864 "capacity units". The number of capacity units, or CU, that a certain bit-rate level requires depends on the amount of error correction added to the transmission, as described above. In the UK, most services transmit using 'protection level three', which provides an average ECC code rate of approximately ½, equating to a maximum bit rate per multiplex of 1184 kbit/s.

Services and ensembles

Various different services are embedded into one ensemble (which is also typically called a multiplex). These services can include:

- Primary services, like main radio stations
- Secondary services, like additional sports commentaries
- Data services
    - Electronic Programme Guide (EPG)
    - Collections of HTML pages and digital images (Known as 'Broadcast Web Sites')
    - Slideshows, which may be synchronised with audio broadcasts
    - Video
    - Java Platform Applications
    - IP tunneling
    - Other raw data

DAB$^+$ and DMB

The term DAB most commonly refers both to a specific DAB-standard using the MP2 audio codec, but can sometimes refer to a whole family of DAB related standards, such as DAB+, DMB and DAB-IP.

DAB$^+$

WorldDMB, the organisation in charge of the DAB standards, announced DAB+, a major upgrade to the DAB standard in 2006. When the HE-AAC v2 audio codec (also known as eAAC+) was adopted. The new standard, which is called DAB+, has also adopted the MPEG Surround audio format and stronger error correction coding in the form of Reed-Solomon coding. DAB+ has been standardised as ETSI TS 102 563.

As DAB is not forward compatible with DAB+, older DAB receivers can not receive DAB+ broadcasts. However, DAB receivers that will be able to receive the new DAB+ standard via a firmware upgrade went on sale in July 2007. If a receiver is DAB+, there will be a sign on the product packaging.

DAB+ broadcasts have launched in several countries like Switzerland, Malta, Italy and Australia. Several other countries are also expected to launch DAB+ broadcasts over the next few years, such as Hungary, Germany and Asian countries, such as China and Vietnam. If DAB+ stations launch in established DAB countries, they can transmit alongside existing DAB stations that use the older MPEG-1 Audio Layer II audio format, and most existing DAB stations are expected to continue broadcasting until the vast majority of receivers support DAB+.

DMB

Digital Multimedia Broadcasting (DMB) and DAB-IP are suitable for mobile radio and TV both because they support MPEG 4 AVC and WMV9 respectively as video codecs. However, a DMB video subchannel can easily be added to any DAB transmission—as DMB was designed from the outset to be carried on a DAB subchannel. DMB broadcasts in Korea carry conventional MPEG 1 Layer II DAB audio services alongside their DMB video services.

Norway, South Korea and France are countries currently broadcasting DMB.

Countries using DAB

More than 30 countries provide DAB, DAB+ and/or DMB broadcasts, either as a permanent technology or as test transmissions.

Chapter- 6

Satellite Radio

Satellite radio is an analogue or digital radio signal that is relayed through one or more satellites and thus can be received in a much wider geographical area than terrestrial FM radio stations. While in Europe many primarily-FM radio stations provide an additional unencrypted satellite feed, there are also subscription based digital packages of numerous channels that do not broadcast terrestrially, notably in the US. In Europe, FM radio is used by many suppliers that use a network of several local FM repeaters to broadcast a single programme to a large area, usually a whole nation. Many of those have an additional satellite signal that can be heard in many parts of the continent. In contrast, US terrestrial stations are always local and each of them has a unique programme, albeit they are sometimes interconnected for syndicated contents; but each local station still carries its own commercial and news breaks even then. This means that a national distribution of the contents of original terrestrial stations via satellite makes no real sense in the US, wherefore satellite radio is used in a different way there. History: Began broadcasting January 5, 2001 at 11:17AM Eastern, Tim McGraw was the first artist ever played on satellite radio. He gave a special welcome introduction which segued into his song "Things Change" on Sirius! Mobile services, such as Sirius, XM, and Worldspace, allow listeners to roam across an entire continent, listening to the same audio programming anywhere they go. Other services, such as Music Choice or Muzak's satellite-delivered content, require a fixed-location receiver and a dish antenna. In all cases, the antenna must have a clear view to the satellites. In areas where tall buildings, bridges, or even parking garages obscure the signal, repeaters can be placed to make the signal available to listeners.

Radio services are usually provided by commercial ventures and are subscription-based. The various services are proprietary signals, requiring specialized hardware for decoding and playback. Providers usually carry a variety of news, weather, sports, and music channels, with the music channels generally being commercial-free.

In areas with a relatively high population density, it is easier and less expensive to reach the bulk of the population with terrestrial broadcasts. Thus in the UK and some other countries, the contemporary evolution of radio services is focused on Digital Audio Broadcasting (DAB) services or HD Radio, rather than satellite radio.

Business applications

Satellite radio, particularly in the United States, has become a major provider of background music to businesses such as hotels, retail chains, and restaurants. Compared to old-line competitors such as Muzak, satellite radio's significantly lower price, commercial-free channel variety, and more reliable technology make it a very attractive option. Both North American satellite radio providers offer business subscriptions, though given the merger of XM Satellite Radio with Sirius, the future of XM for Business is uncertain. Sirius's commercial services are provided nationally by third-party partner Applied Media Technologies Corporation.

System design

Satellite radio uses the 2.3 GHz S band in North America and generally shares the 1.4 GHz L band with local Digital Audio Broadcasting (DAB) stations elsewhere. It is a type of direct broadcast satellite and is strong enough that it requires no satellite dish to receive. Curvature of the earth limits the reach of the signal, but due to the high orbit of the satellites, two or three are usually sufficient to provide coverage for an entire continent.

Local repeaters similar to broadcast translator boosters enable signals to be available even if the view of the satellite is blocked, for example, by skyscrapers in a large town. Major tunnels can also have repeaters. This method also allows local programming to be transmitted such as traffic and weather in most major metropolitan areas, as of March 2004.

Each receiver has an Electronic Serial Number (ESN) Radio ID to identify it. When a unit is activated with a subscription, an authorization code is sent in the digital stream telling the receiver to allow access to the blocked channels. Most services have at least one "free to air" or "in the clear" (ITC) channel as a test. For example, Sirius uses channel 184, Sirius Weather & Emergency.

Most (if not all) of the systems in use now are proprietary, using different codecs for audio data compression, different modulation techniques, and/or different methods for encryption and conditional access.

Like other radio services, satellite radio also transmits program-associated data (PAD or metadata), with the artist and title of each song or program and possibly the name of the channel.

Satellite radio vs. other formats

Satellite radio differs from AM or FM radio and digital television radio (or DTR) in the following ways. The table applies primarily to the United States.

Radio format	Satellite radio	AM/FM	Digital television radio (DTR)
Monthly fees	US$12.95 and up	None	Very low — DTR represents a small portion of the total monthly television fee.
Portability	Available	Prominent	None — a typical set consists of a stereo attached to a television set-top box (the primary function of the set top-box is normally designed for cable or satellite television viewing).
Listening availability	Very high — a satellite signal's footprint covers millions of square kilometres.	Low to moderate — implementation of FM service requires moderate to high population densities and is thus not practical in rural and/or remote locales; AM travels great distances at night.	Very high
Sound quality	Varies[2]	AM: Usually very low, but can be the highest FM: Usually Moderate, but can be very high	Varies[2]
Variety and depth of programming	Highest	Variable — highly dependent upon economic/demographic factors	Variable - dependent on the television provider and the various packages they provide and on the user's subscription.
Frequency of programming interruptions (by DJs or commercial advertising)[3]	None to high - mostly dependent on the channels, some of which have DJs; most channels are advertisement-free because of the paid subscription model	Highest[4]	None to low - dependent on the provider; however, it is common that some stations will have DJs. Usually no advertisements (DirecTV and Dish Network both claim

	of satellite radio.		to provide advertisement-free content).
Governmental regulation	Yes[5]	Yes — significant governmental regulations regarding content[6]	Low to none [5]

² The sound quality with both satellite radio providers and DTR providers varies with each channel. Some channels have near CD-quality audio, and others use low-bandwidth audio suitable only for speech. Since only a certain amount of bandwidth is available within the licenses available, adding more channels means that the quality on some channels must be reduced. Both the frequency response and the dynamic range of satellite channels can be superior to most, but not all AM or FM radio stations, as most AM and FM stations clip the audio peaks to sound louder; even the worst channels are still superior to most AM radios, but a very few AM tuners are equal to or better than the best FM or satellite broadcasts when tuned to a local station, even if not capable of stereo. AM does not suffer from multipath distortion or flutter in a moving vehicle like FM, nor does it become silent as you go behind a big hill like satellite radio.

³ Some satellite radio services and DTR services act as *in situ* repeaters for local AM/FM stations and thus feature a high frequency of interruption.

⁴ Nonprofit stations and public radio networks such as CBC/Radio-Canada, NPR, and PRI-affiliated stations and the BBC are commercial-free. In the US, all stations are required to have periodic station identifications and public service announcements.

⁵ In the United States, the FCC regulates technical broadcast spectrum only. Program content is unregulated. However, the FCC has tried in the past to expand its reach to regulate content to satellite radio and cable television, and its options are still open to attempt such in the future. The FCC does issue licenses to both satellite radio providers (XM and Sirius) and controls who holds these licenses to broadcast.

⁶ Degree of content regulation varies by country; however, the majority of industrialized nations have regulations regarding obscene and/or objectionable content.

Portable Satellite Radio

Portable satellite radios let you listen to satellite radio just about anywhere you go. They are very similar to standard portable music players, designed for music on the go. These however, feature built-in antennas that receive the satellite signal, and come with rechargeable batteries. In fact, all you have to do is plug in headphones, and you can easily listen to and carry them around easily. Reception can be tricky however, being blocked by buildings and tall trees, and sometimes by your own body depending you the way you are facing and how you are carrying it. However, the best reception will be received outdoors in the open.

United States

In the United States and Canada, one holding company, Sirius XM Radio, operates the two satellite radio services, after a merger (technically the acquisition of XM by Sirius) in July 2008. A monthly fee is charged for both services (as of 2005, Sirius also offers a one-time fee of nearly $500 valid for the lifetime of the equipment; however, there is a $70.00 USD fee for switching receiver, and this may be done only three times ever). Some XM music channels have commercials, while Sirius is commercial-free. Both services have commercial-free music stations, as well as talk and news stations, some of which include commercials. XM uses fixed-location geostationary satellites in two positions, and Sirius uses three geosynchronous satellites in highly elliptical orbits passing over North and South America, to transmit the digital streams. The net difference is that the Sirius signal comes from a higher elevation angle in the northern part of the U.S. and even more so in Canada. (This higher angle makes Sirius' signal less likely to drop out on cities, but more likely to drop out in parking garages, gas stations, tunnels, and other covered spaces.)

Both services are available mainly via portable receivers in automobiles, but both have many accessories so one can listen at home through a home stereo, with a portable boombox, or online through a personal computer. Both services now have some form of receiver that is completely portable.

Satellite radio's chief asset is the fact that it is not localized: drivers can receive the same programming anywhere in the footprint of the service. A stop at any truck stop will demonstrate the popularity of Sirius XM among long-haul drivers. In addition, both XM and Sirius carry programming that is simply not feasible on commercial radio stations. Specialty stations cover things such as family talk, radio drama, classical music, and live events.

The footprint of both Sirius and XM is only the United States (not including Alaska), Canada, and the upper third of Mexico; it does not cover Hawaii as satellite TV does.

Success so far

As of July 29, 2008 Sirius XM claims over 18.5 million subscribers. One critical factor for the success of satellite radio is the deployment of in-car receivers. Sirius XM has attempted to convince automakers to equip vehicles with their receiver. As of 2008, the following manufacturers offer satellite radio as original equipment:

Provider	BMW MINI Rolls- Royce	Chrysler Dodge Mercedes- Benz Jeep	Ford Lincoln Mercury Volvo Land Rover Jaguar	GM Cadillac Chevrolet Buick Pontiac GMC Saturn	Honda Acura	Hyundai Kia	Mitsubishi	Nissan Infiniti	Porsche	Toyota Lexus Scion	VW Audi Bentley	Suzuki

			Mazda	Saab								
Sirius	Yes	Yes	Yes	No	No	Yes	Yes	?	No	Yes	Yes	No
XM	No	No	No	Yes	Yes	Yes	No	Yes	Yes	Yes	No	Yes

Sirius has an exclusive contract for VW and Audi vehicles from 2007 through 2012. Those brands previously offered both services. GM, Honda and Suzuki are all major investors in XM; Sirius is not offered as options in their vehicles. Bentley and Rolls-Royce come not only with receivers but lifetime subscriptions for Sirius service as well. XM is featured in select Harley-Davidson motorcycle models, while Sirius can be heard in several brands of recreational vehicles and boats.

One of the challenges for satellite radio has been to move away from cars and into the homes of consumers. Several portable satellite radio receivers have been made for this purpose. XM satellite radio has developed the XM2go line of "Walkman-like" portable receivers, such as the Delphi MyFi, the Pioneer AirWare and Giant International's Tao. Polk Audio makes a component-style home XM Reference Tuner and a tabletop entertainment system, the I-Sonic, with XM capability. Sirius has developed the Kenwood Portable Satellite Radio Tuner, Sirius S50, Here2Anywhere and the Sirius Stiletto 100. The Pioneer Inno and Samsung Helix for XM were among the first portable receivers to offer the ability of recording live content for playback later. Thus allowing for satellite radio to compete more fully with MP3 players.

While key agreements with automobile manufacturers are still being made, both companies have made the leap away from satellite radio only in the car and into the homes of consumers. One bump in the road to becoming more widely used in the home was both Sirius and XM running into legal issues in early 2006 with the FCC about their internal FM Transmitters. This required Sirius and XM to pull several of their models off the shelf and fix the problem. The FCC was claiming that the emissions of the internal FM Transmitters were too powerful and needed to be lowered. With these changes any customer buying a new satellite radio receiver doesn't achieve nearly the broadcast distance as the old models. Since this is a key point in the ability to use a satellite radio in the home (i.e. by taking the signal received and then broadcasting it to multiple points throughout the home at the same time and avoid having to bring the satellite radio with them as they move around the home) it has led many subscribers to use an external Personal FM transmitter like the Whole House FM Transmitter, C. Crane Company, Griffin Technology, etc. to replace the lower powered internal FM Transmitter. Since these external FM Transmitters are Part 15 compliant they can broadcast the signal further than the new internal FM Transmitters now included in the satellite radios and still be legal. These external FM transmitters may prevent a slow down in the progress already made into the home consumer market for Sirius and XM satellite radio.

Satellite radio technology was inducted into the Space Foundation Space Technology Hall of Fame in 2002.

Canada

On November 1, 2004, the Canadian Radio-television and Telecommunications Commission (CRTC) began hearing applications for Canada's first satellite radio operations. Three applications were filed: one by Standard Broadcasting and the CBC in partnership with Sirius, one by Canadian Satellite Radio in partnership with XM, and one at the last minute by CHUM Limited and Astral Media.

The first two would use the same systems already set up for the U.S., while CHUM's application was for a subscription radio service delivered through existing terrestrial DAB transmitters rather than directly by satellite (although satellites would be used to deliver programming to the transmitters). The CHUM service is all-Canadian; the other two applications propose to offer a mix of Canadian-produced channels and existing channels from their American partner services.

A small "grey market" already exists for Sirius and XM receivers in Canada in which a Canadian would have an American order their receiver and setup.

On June 16, 2005, the CRTC approved all three services.

In its decision, the CRTC required the following conditions from the satellite radio licensees:

- A minimum of eight channels must be produced in Canada, and for each Canadian channel, nine foreign channels can be broadcast.
- At least 85% of the content on the Canadian-produced channels (whether musical or spoken word) must be Canadian.
- At least 25% of the Canadian channels must be French-language stations.
- At least 25% of the music aired on the Canadian channels must be new Canadian music.
- At least 25% of the music played on the Canadian channels must be from up-and-coming Canadian artists.

These conditions were an extension of the existing Canadian content rules applicable to all broadcasters in Canada. The applicants had until 13 November 2005, to notify the CRTC of their decision. Both companies managed to negotiate the standards a little to their favor, and in return, they would instead play 50% French content as opposed to 25%. Also, XM Canada succeeded in getting an extra five channels of National Hockey League Play-by-Play onto their platform, without an additional channel creation, by agreeing to cover every Canadian team's game during the season.

CHUM appealed the decision, claiming they would not survive if Sirius and XM both were allowed in the Canadian market, and that the licence conditions regarding Canadian content imposed on Canadian Satellite Radio and Sirius Canada were too lax. Canadian Satellite Radio and Sirius Canada countered that CHUM was simply trying to create a monopoly in the Canadian market.

In late August 2005, Heritage Minister Liza Frulla asked the Federal Cabinet to review the CRTC decision and possibly send it back to the CRTC for further review. Lobbyists complained that the CRTC decision did not require enough Canadian content from the broadcasters. The broadcasters responded by promising to add additional Canadian and French content.

After vigorous lobbying from both sides, the federal cabinet officially accepted the CRTC decision on September 10, 2005.

XM satellite radio was launched in Canada on November 29, 2005. Sirius followed later on December 1, 2005. Monthly subscription rates are $12.99 for XM (85 channels) with a one-time activation fee of $19.99 and $14.99 for Sirius with a one-time activation fee of $19.99 (100 channels). (All prices are in Canadian dollars.) The CHUM/Astral service never launched, and its license expired on June 16, 2007.

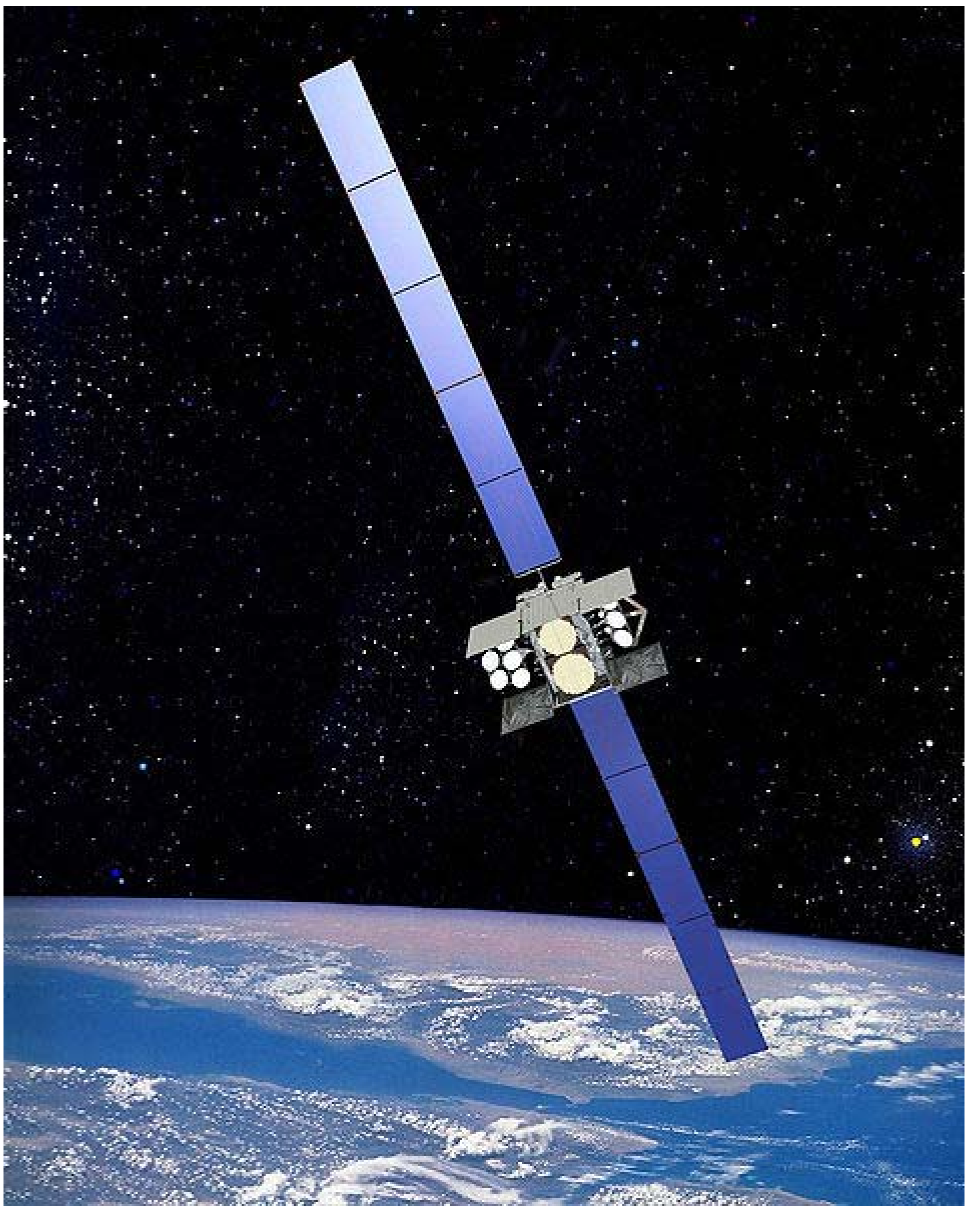
Europe

Eutelsat W2A satellite carrying a Solaris Mobile (an Eutelsat and SES Astra joint venture) DVB-SH S band payload was launched on 3 April 2009.

WorldSpace Europe ( ) and ONDAS Media ( ) will use ETSI SDR for their new networks covering Europe.

Chapter- 7

Communications Satellite



U.S. military WGSS communications satellite

A **communications satellite** (sometimes abbreviated to **COMSAT**) is an artificial satellite stationed in space for the purpose of telecommunications. Modern communications satellites use a variety of orbits including geostationary orbits, Molniya orbits, other elliptical orbits and low (polar and non-polar) Earth orbits.

For fixed (point-to-point) services, communications satellites provide a microwave radio relay technology complementary to that of submarine communication cables. They are also used for mobile applications such as communications to ships, vehicles, planes and hand-held terminals, and for TV and radio broadcasting, for which application of other technologies, such as cable, is impractical or impossible.

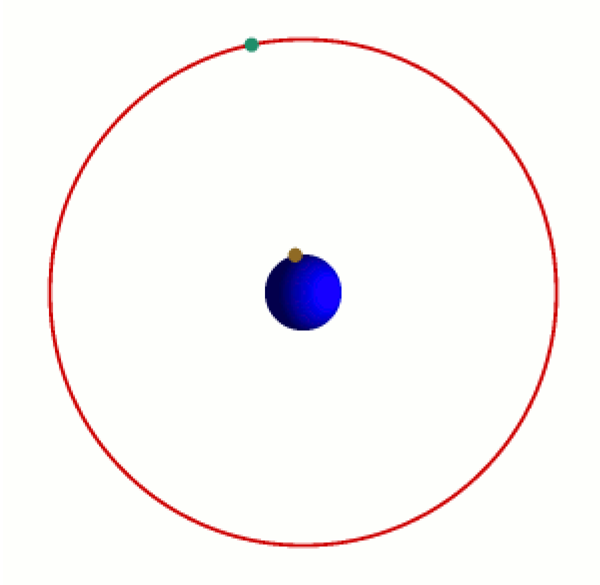
History

The first artificial satellite was the Soviet Sputnik 1, launched on October 4, 1957, and equipped with an on-board radio-transmitter that worked on two frequencies, 20.005 and 40.002 MHz. The first American satellite to relay communications was Project SCORE in 1958, which used a tape recorder to store and forward voice messages. It was used to send a Christmas greeting to the world from U.S. President Dwight D. Eisenhower. NASA launched an Echo satellite in 1960; the 100-foot (30 m) aluminized PET film balloon served as a passive reflector for radio communications. Courier 1B, built by Philco, also launched in 1960, was the world’s first active repeater satellite.

Telstar was the first active, direct relay communications satellite. Belonging to AT&T as part of a multi-national agreement between AT&T, Bell Telephone Laboratories, NASA, the British General Post Office, and the French National PTT (Post Office) to develop satellite communications, it was launched by NASA from Cape Canaveral on July 10, 1962, the first privately sponsored space launch. Telstar was placed in an elliptical orbit (completed once every 2 hours and 37 minutes), rotating at a 45° angle above the equator.

An immediate antecedent of the geostationary satellites was Hughes’ Syncom 2, launched on July 26, 1963. Syncom 2 revolved around the earth once per day at constant speed, but because it still had north-south motion, special equipment was needed to track it.

Geostationary orbits



Geostationary orbit

A satellite in a geostationary orbit appears to be in a fixed position to an earth-based observer. A geostationary satellite revolves around the earth at a constant speed once per day over the equator.

The geostationary orbit is useful for communications applications because ground based antennas, which must be directed toward the satellite, can operate effectively without the need for expensive equipment to track the satellite's motion. Especially for applications that require a large number of ground antennas (such as direct TV distribution), the savings in ground equipment can more than justify the extra cost and onboard complexity of lifting a satellite into the relatively high geostationary orbit.

The concept of the geostationary communications satellite was first proposed by Arthur C. Clarke, building on work by Konstantin Tsiolkovsky and on the 1929 work by Herman Potočnik (writing as Herman Noordung) *Das Problem der Befahrung des Weltraums - der Raketen-motor*. In October 1945 Clarke published an article titled "Extra-terrestrial Relays" in the British magazine *Wireless World*. The article described the fundamentals behind the deployment of artificial satellites in geostationary orbits for the purpose of relaying radio signals. Thus Arthur C. Clarke is often quoted as being the inventor of the communications satellite.

The first truly geostationary satellite launched in orbit was the Syncom 3, launched on August 19, 1964. It was placed in orbit at 180° east longitude, over the International Date Line. It was used that same year to relay experimental television coverage of the 1964 Summer Olympics in Tokyo, Japan to the United States, making these Olympic games the first to be broadcast internationally. Although Syncom 3 is some times credited with the first television transmission to cross the Pacific Ocean, the Relay 1 satellite first broadcast from the United States to Japan on November 22, 1963.

Shortly after Syncom 3, Intelsat I, aka *Early Bird*, was launched on April 6, 1965 and placed in orbit at 28° west longitude. It was the first geostationary satellite for telecommunications over the Atlantic Ocean.

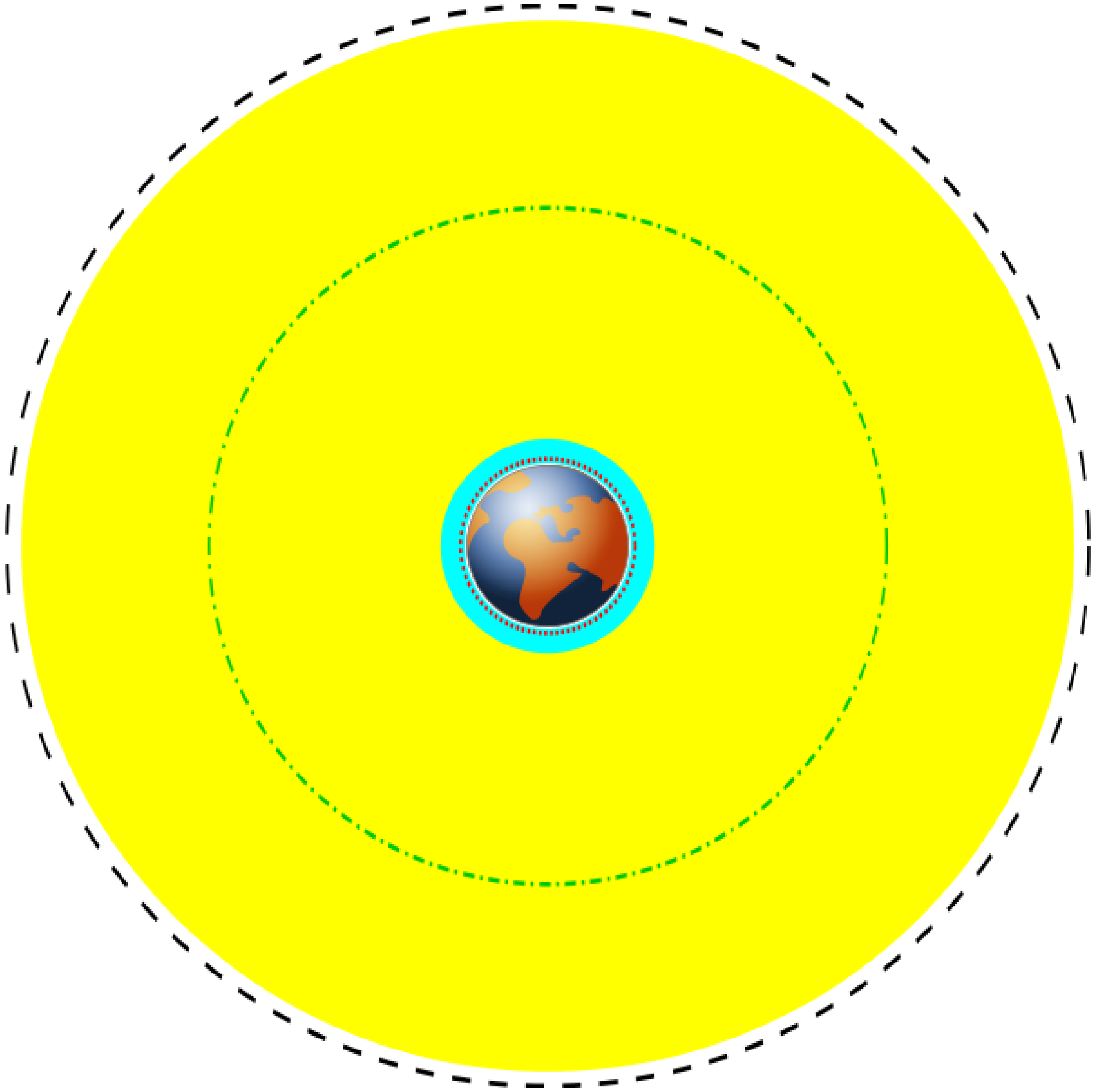
On November 9, 1972, Canada's first geostationary satellite serving the continent, Anik A1, was launched by Telesat Canada, with the United States following suit with the launch of Westar 1 by Western Union on April 13, 1974.

On May 30, 1974, the first geostationary communications satellite in the world to be three-axis stabilized was launched: the experimental satellite ATS-6 built for NASA

After the launches of Telstar, Syncom 3, Early Bird, Anik A1, and Westar 1, RCA Americom (later GE Americom, now SES Americom) launched Satcom 1 in 1975. It was Satcom 1 that was instrumental in helping early cable TV channels such as WTBS (now TBS Superstation), HBO, CBN (now ABC Family), and The Weather Channel become successful, because these channels distributed their programming to all of the local cable TV headends using the satellite. Additionally, it was the first satellite used by broadcast television networks in the United States, like ABC, NBC, and CBS, to distribute programming to their local affiliate stations. Satcom 1 was widely used because it had twice the communications capacity of the competing Westar 1 in America (24 transponders as opposed to the 12 of Westar 1), resulting in lower transponder-usage costs. Satellites in later decades tended to have even higher transponder numbers.

By 2000, Hughes Space and Communications (now Boeing Satellite Development Center) had built nearly 40 percent of the more than one hundred satellites in service worldwide. Other major satellite manufacturers include Space Systems/Loral, Orbital Sciences Corporation with the STAR Bus series, Indian Space Research Organization, Lockheed Martin (owns the former RCA Astro Electronics/GE Astro Space business), Northrop Grumman, Alcatel Space, now Thales Alenia Space, with the Spacebus series, and Astrium.

Low-Earth-orbiting satellites



Low Earth orbit in Cyan

A Low Earth Orbit (LEO) typically is a circular orbit about 400 kilometres above the earth's surface and, correspondingly, a period (time to revolve around the earth) of about 90 minutes. Because of their low altitude, these satellites are only visible from within a radius of roughly 1000 kilometres from the sub-satellite point. In addition, satellites in low earth orbit change their position relative to the ground position quickly. So even for local applications, a large number of satellites are needed if the mission requires uninterrupted connectivity.

Low earth orbiting satellites are less expensive to launch into orbit than geostationary satellites and, due to proximity to the ground, do not require as high signal strength (Recall that signal strength falls off as the square of the distance from the source, so the effect is dramatic). Thus there is a trade off between the number of satellites and their cost. In addition, there are important differences in the onboard and ground equipment needed to support the two types of missions.

A group of satellites working in concert is known as a satellite constellation. Two such constellations, intended to provide satellite phone services, primarily to remote areas, are the Iridium and Globalstar systems. The Iridium system has 66 satellites. Another LEO satellite constellation known as Teledesic, with backing from Microsoft entrepreneur Paul Allen, was to have over 840 satellites. This was later scaled back to 288 and ultimately ended up only launching one test satellite.

It is also possible to offer discontinuous coverage using a low Earth orbit satellite capable of storing data received while passing over one part of Earth and transmitting it later while passing over another part. This will be the case with the CASCADE system of Canada's CASSIOPE communications satellite. Another system using this store and forward method is Orbcomm.

Molniya satellites

As mentioned, geostationary satellites are constrained to operate above the equator. As a consequence, they are not always suitable for providing services at high latitudes: at high latitudes, a geostationary satellite will appear low on the horizon, affecting connectivity and causing multipath (interference caused by signals reflecting off the ground and into the ground antenna). The first satellite of the Molniya series was launched on April 23, 1965 and was used for experimental transmission of TV signal from a Moscow uplink station to downlink stations located in Siberia and the Russian Far East, in Norilsk, Khabarovsk, Magadan and Vladivostok. In November 1967 Soviet engineers created a unique system of national TV network of satellite television, called Orbita, that was based on Molniya satellites.

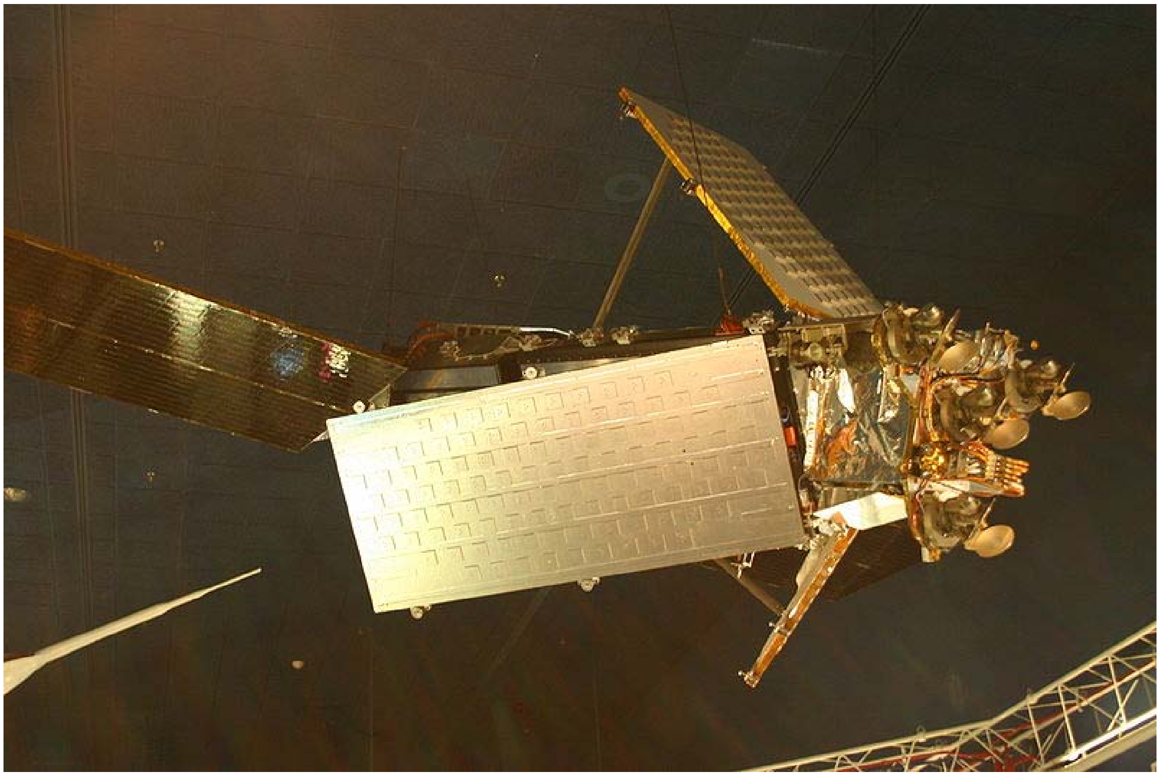
Molniya orbits can be an appealing alternative in such cases. The Molniya orbit is highly inclined, guaranteeing good elevation over selected positions during the northern portion of the orbit. (Elevation is the extent of the satellite's position above the horizon. Thus, a satellite at the horizon has zero elevation and a satellite directly overhead has elevation of 90 degrees).

Furthermore, the Molniya orbit is designed so that the satellite spends the great majority of its time over the far northern latitudes, during which its ground footprint moves only slightly. Its period is one half day, so that the satellite is available for operation over the targeted region for eight hours every second revolution. In this way a constellation of three Molniya satellites (plus in-orbit spares) can provide uninterrupted coverage.

Molniya satellites are typically used for telephony and TV services over Russia. Another application is to use them for mobile radio systems (even at lower latitudes) since cars travelling through urban areas need access to satellites at high elevation in order to secure good connectivity, e.g. in the presence of tall buildings.

Applications

Telephone


An Iridium satellite

The first and historically most important application for communication satellites was in intercontinental long distance telephony. The fixed Public Switched Telephone Network relays telephone calls from land line telephones to an earth station, where they are then transmitted to a geostationary satellite. The downlink follows an analogous path. Improvements in submarine communications cables, through the use of fiber-optics, caused some decline in the use of satellites for fixed telephony in the late 20th century, but they still serve remote islands such as Ascension Island, Saint Helena, Diego Garcia, and Easter Island, where no submarine cables are in service. There are also regions of some continents and countries where landline telecommunications are rare to nonexistent, for example large regions of South America, Africa, Canada, China, Russia, and Australia. Satellite communications also provide connection to the edges of Antarctica and Greenland.

Satellite phones connect directly to a constellation of either geostationary or low-earth-orbit satellites. Calls are then forwarded to a satellite teleport connected to the Public Switched Telephone Network

Satellite television

As television became the main market, its demand for simultaneous delivery of relatively few signals of large bandwidth to many receivers being a more precise match for the capabilities of geosynchronous comsats. Two satellite types are used for North American television and radio: Direct Broadcast Satellite (DBS), and Fixed Service Satellite (FSS)

The definitions of FSS and DBS satellites outside of North America, especially in Europe, are a bit more ambiguous. Most satellites used for direct-to-home television in Europe have the same high power output as DBS-class satellites in North America, but use the same linear polarization as FSS-class satellites. Examples of these are the Astra, Eutelsat, and Hotbird spacecraft in orbit over the European continent. Because of this, the terms FSS and DBS are more so used throughout the North American continent, and are uncommon in Europe.

Fixed Service Satellite

Fixed Service Satellites use the C band, and the lower portions of the K_u bands. They are normally used for broadcast feeds to and from television networks and local affiliate stations (such as program feeds for network and syndicated programming, live shots, and backhauls), as well as being used for distance learning by schools and universities, business television (BTV), Videoconferencing, and general commercial telecommunications. FSS satellites are also used to distribute national cable channels to cable television headends.

Free-to-air satellite TV channels are also usually distributed on FSS satellites in the K_u band. The Intelsat Americas 5, Galaxy 10R and AMC 3 satellites over North America provide a quite large amount of FTA channels on their K_u band transponders.

The American DISH Network DBS service has also recently utilized FSS technology as well for their programming packages requiring their SuperDish antenna, due to Dish Network needing more capacity to carry local television stations per the FCC's "must-carry" regulations, and for more bandwidth to carry HDTV channels.

Direct broadcast satellite

A **direct broadcast satellite** is a communications satellite that transmits to small DBS satellite dishes (usually 18 to 24 inches or 45 to 60 cm in diameter). Direct broadcast satellites generally operate in the upper portion of the microwave K_u band. DBS technology is used for DTH-oriented (Direct-To-Home) satellite TV services, such as DirecTV and DISH Network in the United States, Bell TV and Shaw Direct in Canada, Freesat and Sky Digital in the UK, the Republic of Ireland, and New Zealand.

Operating at lower frequency and lower power than DBS, FSS satellites require a much larger dish for reception (3 to 8 feet (1 to 2.5m) in diameter for K_u band, and 12 feet (3.6m) or larger for C band). They use linear polarization for each of the transponders' RF input and output (as opposed to circular polarization used by DBS satellites), but this is a minor technical difference that users do not notice. FSS satellite technology was also originally used for DTH satellite TV from the late 1970s to the early 1990s in the United States in the form of TVRO (TeleVision Receive Only) receivers and dishes. It was also used in its K_u band form for the now-defunct Primestar satellite TV service.

Satellites for communication have now been launched that have transponders in the K_a band, such as DirecTV's SPACEWAY-1 satellite, and Anik F2. NASA as well has launched experimental satellites using the K_a band recently.

Mobile satellite technologies

Initially available for broadcast to stationary TV receivers, by 2004 popular mobile direct broadcast applications made their appearance with that arrival of two satellite radio systems in the United States: Sirius and XM Satellite Radio Holdings. Some manufacturers have also introduced special antennas for mobile reception of DBS television. Using Global Positioning System (GPS) technology as a reference, these antennas automatically re-aim to the satellite no matter where or how the vehicle (on which the antenna is mounted) is situated. These mobile satellite antennas are popular with some recreational vehicle owners. Such mobile DBS antennas are also used by JetBlue Airways for DirecTV (supplied by LiveTV, a subsidiary of JetBlue), which passengers can view on-board on LCD screens mounted in the seats.

Satellite radio

Satellite radio offers audio services in some countries, notably the United States. Mobile services allow listeners to roam a continent, listening to the same audio programming anywhere.

A satellite radio or subscription radio (SR) is a digital radio signal that is broadcast by a communications satellite, which covers a much wider geographical range than terrestrial radio signals.

Satellite radio offers a meaningful alternative to ground-based radio services in some countries, notably the United States. Mobile services, such as Sirius, XM, and Worldspace, allow listeners to roam across an entire continent, listening to the same audio programming anywhere they go. Other services, such as Music Choice or Muzak's satellite-delivered content, require a fixed-location receiver and a dish antenna. In all cases, the antenna must have a clear view to the satellites. In areas where tall buildings, bridges, or even parking garages obscure the signal, repeaters can be placed to make the signal available to listeners.

Radio services are usually provided by commercial ventures and are subscription-based. The various services are proprietary signals, requiring specialized hardware for decoding and playback. Providers usually carry a variety of news, weather, sports, and music channels, with the music channels generally being commercial-free.

In areas with a relatively high population density, it is easier and less expensive to reach the bulk of the population with terrestrial broadcasts. Thus in the UK and some other countries, the contemporary evolution of radio services is focused on Digital Audio Broadcasting (DAB) services or HD Radio, rather than satellite radio.

Amateur radio

Amateur radio operators have access to the OSCAR satellites that have been designed specifically to carry amateur radio traffic. Most such satellites operate as spaceborne repeaters, and are generally accessed by amateurs equipped with UHF or VHF radio equipment and highly directional antennas such as Yagis or dish antennas. Due to launch costs, most current amateur satellites are launched into fairly low Earth orbits, and are designed to deal with only a limited number of brief contacts at any given time. Some satellites also provide data-forwarding services using the AX.25 or similar protocols.

Satellite Internet

After the 1990s, satellite communication technology has been used as a means to connect to the Internet via broadband data connections. This can be very useful for users who are located in very remote areas, and cannot access a broadband connection.

Military uses

Communications satellites are used for military communications applications, such as Global Command and Control Systems. Examples of military systems that use communication satellites are the MILSTAR, the DSCS, and the FLTSATCOM of the United States, NATO satellites, United Kingdom satellites, and satellites of the former Soviet Union. Many military satellites operate in the X-band, and some also use UHF radio links, while MILSTAR also utilizes Ka band.

Chapter- 8

Code Division Multiple Access

Code division multiple access (**CDMA**) is a channel access method used by various radio communication technologies. It should not be confused with the mobile phone standards called cdmaOne and CDMA2000 (which are often referred to as simply *CDMA*), which use CDMA as an underlying channel access method.

One of the basic concepts in data communication is the idea of allowing several transmitters to send information simultaneously over a single communication channel. This allows several users to share a band of frequencies. This concept is called Multiple Access. CDMA employs spread-spectrum technology and a special coding scheme (where each transmitter is assigned a code) to allow multiple users to be multiplexed over the same physical channel. By contrast, time division multiple access (*TDMA*) divides access by time, while frequency-division multiple access (*FDMA*) divides it by frequency. CDMA is a form of spread-spectrum signalling, since the modulated coded signal has a much higher data bandwidth than the data being communicated.

An analogy to the problem of multiple access is a room (channel) in which people wish to talk to each other simultaneously. To avoid confusion, people could take turns speaking (time division), speak at different pitches (frequency division), or speak in different languages (code division). CDMA is analogous to the last example where people speaking the same language can understand each other, but other languages are perceived as noise and rejected. Similarly, in radio CDMA, each group of users is given a shared code. Many codes occupy the same channel, but only users associated with a particular code can communicate.

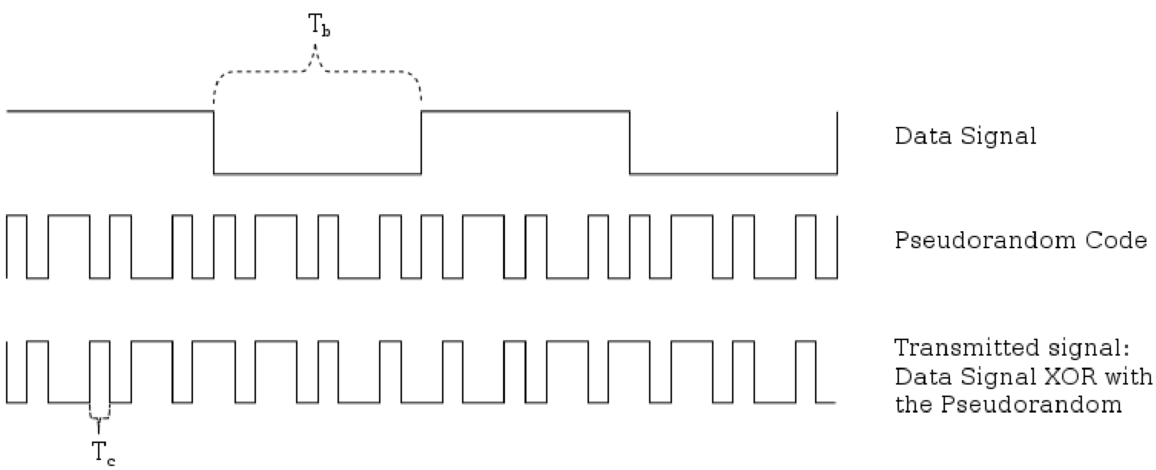
Uses

A CDMA2000 mobile phone

- One of the early applications for code division multiplexing is in GPS. This predates and is distinct from cdmaOne.
- The Qualcomm standard IS-95, marketed as cdmaOne.
- The Qualcomm standard IS-2000, known as CDMA2000. This standard is used by several mobile phone companies, including the Globalstar satellite phone network.
- CDMA has been used in the **OmniTRACS** satellite system for transportation logistics.

Steps in CDMA Modulation

CDMA is a spread spectrum multiple access technique. A spread spectrum technique spreads the bandwidth of the data uniformly for the same transmitted power. Spreading code is a pseudo-random code that has a narrow Ambiguity function, unlike other narrow pulse codes. In CDMA a locally generated code runs at a much higher rate than the data to be transmitted. Data for transmission is combined via bitwise XOR (exclusive OR) with the faster code. The figure shows how spread spectrum signal is generated. The data signal with pulse duration of T_b is XOR'ed with the code signal with pulse duration of T_c. (Note: bandwidth is proportional to $1 / T$ where T = bit time) Therefore, the bandwidth of the data signal is $1 / T_b$ and the bandwidth of the spread spectrum signal is $1 / T_c$. Since T_c is much smaller than T_b, the bandwidth of the spread spectrum signal is much larger than the bandwidth of the original signal. The ratio T_b / T_c is called spreading factor or processing gain and determines to a certain extent the upper limit of the total number of users supported simultaneously by a base station.



Each user in a CDMA system uses a different code to modulate their signal. Choosing the codes used to modulate the signal is very important in the performance of CDMA systems. The best performance will occur when there is good separation between the signal of a desired user and the signals of other users. The separation of the signals is made by correlating the received signal with the locally generated code of the desired user. If the signal matches the desired user's code then the correlation function will be high and the system can extract that signal. If the desired user's code has nothing in common with the signal the correlation should be as close to zero as possible (thus eliminating the signal); this is referred to as cross correlation. If the code is correlated with the signal at any time offset other than zero, the correlation should be as close to zero as possible. This is referred to as auto-correlation and is used to reject multi-path interference.

In general, CDMA belongs to two basic categories: synchronous (orthogonal codes) and asynchronous (pseudorandom codes).

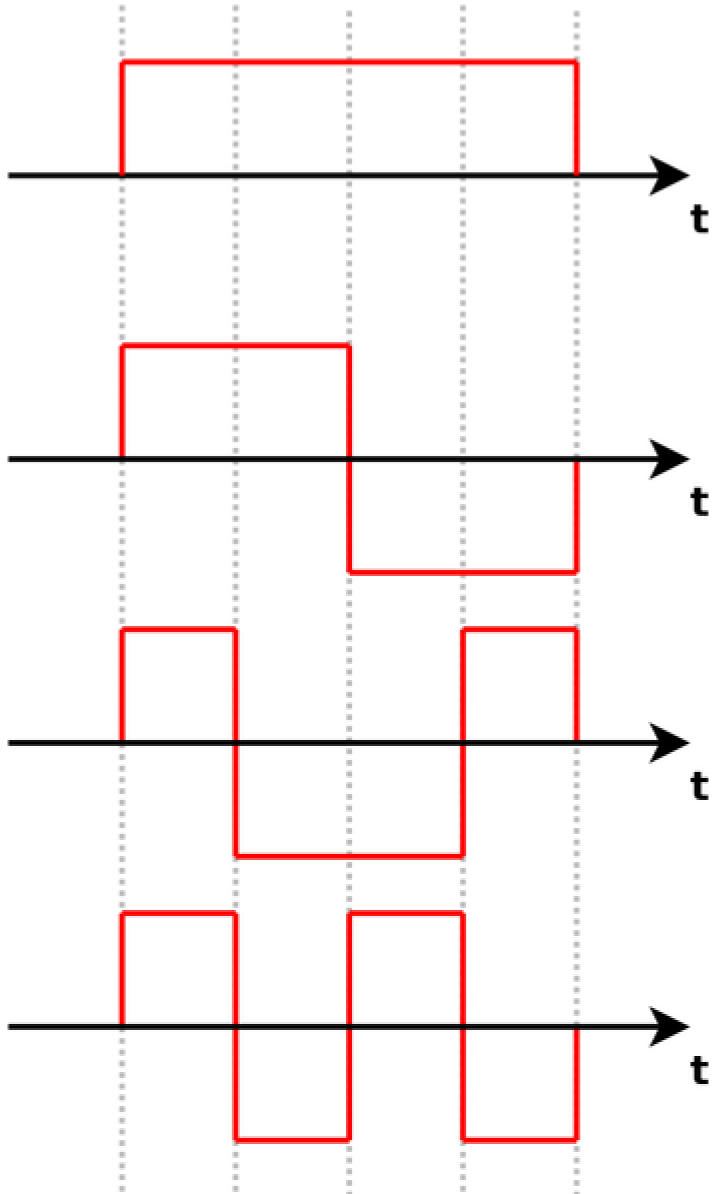
Code division multiplexing (Synchronous CDMA)

Synchronous CDMA exploits mathematical properties of orthogonality between vectors representing the data strings. For example, binary string *1011* is represented by the vector (1, 0, 1, 1). Vectors can be multiplied by taking their dot product, by summing the products of their respective components. If the dot product is zero, the two vectors are said to be *orthogonal* to each other (note: if u = (a, b) and v = (c, d), the dot product u·v = ac + bd). Some properties of the dot product aid understanding of how W-CDMA works. If vectors *a* and *b* are orthogonal, then $\mathbf{a}\cdot\mathbf{b}=0$and:

$$\begin{aligned}
\mathbf{a}\cdot(\mathbf{a}+\mathbf{b}) &= ||\mathbf{a}||^2 && \text{since} \quad \mathbf{a}\cdot\mathbf{a}+\mathbf{a}\cdot\mathbf{b} = ||a||^2+0 \\
\mathbf{a}\cdot(-\mathbf{a}+\mathbf{b}) &= -||\mathbf{a}||^2 && \text{since} \quad -\mathbf{a}\cdot\mathbf{a}+\mathbf{a}\cdot\mathbf{b} = -||a||^2+0 \\
\mathbf{b}\cdot(\mathbf{a}+\mathbf{b}) &= ||\mathbf{b}||^2 && \text{since} \quad \mathbf{b}\cdot\mathbf{a}+\mathbf{b}\cdot\mathbf{b} = 0+||b||^2 \\
\mathbf{b}\cdot(\mathbf{a}-\mathbf{b}) &= -||\mathbf{b}||^2 && \text{since} \quad \mathbf{b}\cdot\mathbf{a}-\mathbf{b}\cdot\mathbf{b} = 0-||b||^2
\end{aligned}$$

Each user in synchronous CDMA uses a code orthogonal to the others' codes to modulate their signal. An example of four mutually orthogonal digital signals is shown in the figure. Orthogonal codes have a cross-correlation equal to zero; in other words, they do not interfere with each other. In the case of IS-95 64 bit Walsh codes are used to encode the signal to separate different users. Since each of the 64 Walsh codes are orthogonal to one another, the signals are channelized into 64 orthogonal signals. The following example demonstrates how each user's signal can be encoded and decoded.

Example



An example of four mutually orthogonal digital signals.

Start with a set of vectors that are mutually orthogonal. (Although mutual orthogonality is the only condition, these vectors are usually constructed for ease of decoding, for example columns or rows from Walsh matrices.) An example of orthogonal functions is shown in the picture on the left. These vectors will be assigned to individual users and are called the *code*, *chip code*, or *chipping code*. In the interest of brevity, the rest of this example uses codes, **v**, with only 2 bits.

Each user is associated with a different code, say **v**. A 1 bit is represented by transmitting a positive code, **v**, and a 0 bit is represented by a negative code, –**v**. For example, if **v** = (1, –1) and the data that the user wishes to transmit is (1, 0, 1, 1), then the transmitted symbols would be $(1, -1, 1, 1) \otimes \mathbf{v} = (v_0, v_1, -v_0, -v_1, v_0, v_1, v_0, v_1) = (1, -1, -1, 1, 1, -1, 1, -1)$, where $\otimes$ is the Kronecker product.

Each sender has a different, unique vector **v** chosen from that set, but the construction method of the transmitted vector is identical.

Now, due to physical properties of interference, if two signals at a point are in phase, they add to give twice the amplitude of each signal, but if they are out of phase, they subtract and give a signal that is the difference of the amplitudes. Digitally, this behaviour can be modelled by the addition of the transmission vectors, component by component.

If sender0 has code (1, –1) and data (1, 0, 1, 1), and sender1 has code (1, 1) and data (0, 0, 1, 1), and both senders transmit simultaneously, then this table describes the coding steps:

Step	Encode sender0	Encode sender1
0	code0 = (1, –1), data0 = (1, 0, 1, 1)	code1 = (1, 1), data1 = (0, 0, 1, 1)
1	encode0 = 2(1, 0, 1, 1) – (1, 1, 1, 1) = (1, –1, 1, 1)	encode1 = 2(0, 0, 1, 1) – (1, 1, 1, 1) = (–1, –1, 1, 1)
2	signal0 = encode0 $\otimes$ code0 = (1, –1, 1, 1) $\otimes$ (1, –1) = (1, –1, –1, 1, 1, –1, 1, –1)	signal1 = encode1 $\otimes$ code1 = (–1, –1, 1, 1) $\otimes$ (1, 1) = (–1, –1, –1, –1, 1, 1, 1, 1)

Because signal0 and signal1 are transmitted at the same time into the air, they add to produce the raw signal:

$$(1, -1, -1, 1, 1, -1, 1, -1) + (-1, -1, -1, -1, 1, 1, 1, 1) = (0, -2, -2, 0, 2, 0, 2, 0)$$

This raw signal is called an interference pattern. The receiver then extracts an intelligible signal for any known sender by combining the sender's code with the interference pattern, the receiver combines it with the codes of the senders. The following table explains how this works and shows that the signals do not interfere with one another:

Step	Decode sender0	Decode sender1
0	code0 = (1, –1), signal = (0, –2, –2, 0, 2, 0, 2, 0)	code1 = (1, 1), signal = (0, –2, –2, 0, 2, 0, 2, 0)
1	decode0 = pattern.vector0	decode1 = pattern.vector1
2	decode0 = ((0, –2), (–2, 0), (2, 0), (2, 0)).(1, –1)	decode1 = ((0, –2), (–2, 0), (2, 0), (2, 0)).(1, 1)
3	decode0 = ((0 + 2), (–2 + 0), (2 + 0), (2 + 0))	decode1 = ((0 – 2), (–2 + 0), (2 + 0), (2 + 0))

4	data0=(2, –2, 2, 2), meaning (1, 0, 1, 1)	data1=(–2, –2, 2, 2), meaning (0, 0, 1, 1)

Further, after decoding, all values greater than 0 are interpreted as 1 while all values less than zero are interpreted as 0. For example, after decoding, data0 is (2, –2, 2, 2), but the receiver interprets this as (1, 0, 1, 1). Values of exactly 0 means that the sender did not transmit any data, as in the following example:

Assume signal0 = (1, –1, –1, 1, 1, –1, 1, –1) is transmitted alone. The following table shows the decode at the receiver:

Step	Decode sender0	Decode sender1
0	code0 = (1, –1), signal = (1, –1, –1, 1, 1, –1, 1, –1)	code1 = (1, 1), signal = (1, –1, –1, 1, 1, –1, 1, –1)
1	decode0 = pattern.vector0	decode1 = pattern.vector1
2	decode0 = ((1, –1), (–1, 1), (1, –1), (1, –1)).(1, –1)	decode1 = ((1, –1), (–1, 1), (1, –1), (1, –1)).(1, 1)
3	decode0 = ((1 + 1), (–1 – 1),(1 + 1), (1 + 1))	decode1 = ((1 – 1), (–1 + 1),(1 – 1), (1 – 1))
4	data0 = (2, –2, 2, 2), meaning (1, 0, 1, 1)	data1 = (0, 0, 0, 0), meaning no data

When the receiver attempts to decode the signal using sender1's code, the data is all zeros, therefore the cross correlation is equal to zero and it is clear that sender1 did not transmit any data.

Asynchronous CDMA

The previous example of orthogonal Walsh sequences describes how 2 users can be multiplexed together in a synchronous system, a technique that is commonly referred to as *code division multiplexing* (CDM). The set of 4 Walsh sequences shown in the figure will afford up to 4 users, and in general, an NxN Walsh matrix can be used to multiplex N users. Multiplexing requires all of the users to be coordinated so that each transmits their assigned sequence **v** (or the complement, –**v**) so that they arrive at the receiver at exactly the same time. Thus, this technique finds use in base-to-mobile links, where all of the transmissions originate from the same transmitter and can be perfectly coordinated.

On the other hand, the mobile-to-base links cannot be precisely coordinated, particularly due to the mobility of the handsets, and require a somewhat different approach. Since it is not mathematically possible to create signature sequences that are both orthogonal for arbitrarily random starting points and which make full use of the code space, unique "pseudo-random" or "pseudo-noise" (PN) sequences are used in *asynchronous* CDMA systems. A PN code is a binary sequence that appears random but can be reproduced in a deterministic manner by intended receivers. These PN codes are used to encode and decode a user's signal in Asynchronous CDMA in the same manner as the orthogonal codes in synchronous CDMA (shown in the example above). These PN sequences are statistically uncorrelated, and the sum of a large number of PN sequences results in *multiple access interference* (MAI) that is approximated by a Gaussian noise process (following the central limit theorem in statistics). Gold codes are an example of a PN suitable for this purpose, as there is low correlation between the codes. If all of the users are received with the same power level, then the variance (e.g., the noise power) of the MAI increases in direct proportion to the number of users. In other words, unlike synchronous CDMA, the signals of other users will appear as noise to the signal of interest and interfere slightly with the desired signal in proportion to number of users.

All forms of CDMA use spread spectrum process gain to allow receivers to partially discriminate against unwanted signals. Signals encoded with the specified PN sequence (code) are received, while signals with different codes (or the same code but a different timing offset) appear as wideband noise reduced by the process gain.

Since each user generates MAI, controlling the signal strength is an important issue with CDMA transmitters. A CDM (synchronous CDMA), TDMA, or FDMA receiver can in theory completely reject arbitrarily strong signals using different codes, time slots or frequency channels due to the orthogonality of these systems. This is not true for Asynchronous CDMA; rejection of unwanted signals is only partial. If any or all of the unwanted signals are much stronger than the desired signal, they will overwhelm it. This leads to a general requirement in any asynchronous CDMA system to approximately match the various signal power levels as seen at the receiver. In CDMA cellular, the base station uses a fast closed-loop power control scheme to tightly control each mobile's transmit power.

Advantages of asynchronous CDMA over other techniques

Efficient Practical utilization of Fixed Frequency Spectrum

In theory, CDMA, TDMA and FDMA have exactly the same spectral efficiency but practically, each has its own challenges – power control in the case of CDMA, timing in the case of TDMA, and frequency generation/filtering in the case of FDMA.

TDMA systems must carefully synchronize the transmission times of all the users to ensure that they are received in the correct timeslot and do not cause interference. Since this cannot be perfectly controlled in a mobile environment, each timeslot must have a guard-time, which reduces the probability that users will interfere, but decreases the spectral efficiency. Similarly, FDMA systems must use a guard-band between adjacent channels, due to the unpredictable doppler shift of the signal spectrum because of user mobility. The guard-bands will reduce the probability that adjacent channels will interfere, but decrease the utilization of the spectrum.

Flexible Allocation of Resources

Asynchronous CDMA offers a key advantage in the flexible allocation of resources i.e. allocation of a PN codes to active users. In the case of CDM, TDMA, and FDMA the number of simultaneous orthogonal codes, time slots and frequency slots respectively is fixed hence the capacity in terms of number of simultaneous users is limited. There are a fixed number of orthogonal codes, timeslots or frequency bands that can be allocated for CDM, TDMA, and FDMA systems, which remain underutilized due to the bursty nature of telephony and packetized data transmissions. There is no strict limit to the number of users that can be supported in an asynchronous CDMA system, only a practical limit governed by the desired bit error probability, since the SIR (Signal to Interference Ratio) varies inversely with the number of users. In a bursty traffic environment like mobile telephony, the advantage afforded by asynchronous CDMA is that the performance (bit error rate) is allowed to fluctuate randomly, with an average value determined by the number of users times the percentage of utilization. Suppose there are 2N users that only talk half of the time, then 2N users can be accommodated with the same *average* bit error probability as N users that talk all of the time. The key difference here is that the bit error probability for N users talking all of the time is constant, whereas it is a *random* quantity (with the same mean) for 2N users talking half of the time.

In other words, asynchronous CDMA is ideally suited to a mobile network where large numbers of transmitters each generate a relatively small amount of traffic at irregular intervals. CDM (synchronous CDMA), TDMA, and FDMA systems cannot recover the underutilized resources inherent to bursty traffic due to the fixed number of orthogonal codes, time slots or frequency channels that can be assigned to individual transmitters. For instance, if there are N time slots in a TDMA system and 2N users that talk half of the time, then half of the time there will be more than N users needing to use more than N timeslots. Furthermore, it would require significant overhead to continually allocate and deallocate the orthogonal code, time-slot or frequency channel resources. By comparison, asynchronous CDMA transmitters simply send when they have something to say, and go off the air when they don't, keeping the same PN signature sequence as long as they are connected to the system.

Spread-spectrum characteristics of CDMA

Most modulation schemes try to minimize the bandwidth of this signal since bandwidth is a limited resource. However, spread spectrum techniques use a transmission bandwidth that is several orders of magnitude greater than the minimum required signal bandwidth. One of the initial reasons for doing this was military applications including guidance and communication systems. These systems were designed using spread spectrum because of its security and resistance to jamming. Asynchronous CDMA has some level of privacy built in because the signal is spread using a pseudo-random code; this code makes the spread spectrum signals appear random or have noise-like properties. A receiver cannot demodulate this transmission without knowledge of the pseudo-random sequence used to encode the data. CDMA is also resistant to jamming. A jamming signal only has a finite amount of power available to jam the signal. The jammer can either spread its energy over the entire bandwidth of the signal or jam only part of the entire signal.

CDMA can also effectively reject narrowband interference. Since narrowband interference affects only a small portion of the spread spectrum signal, it can easily be removed through notch filtering without much loss of information. Convolution encoding and interleaving can be used to assist in recovering this lost data. CDMA signals are also resistant to multipath fading. Since the spread spectrum signal occupies a large bandwidth only a small portion of this will undergo fading due to multipath at any given time. Like the narrowband interference this will result in only a small loss of data and can be overcome.

Another reason CDMA is resistant to multipath interference is because the delayed versions of the transmitted pseudo-random codes will have poor correlation with the original pseudo-random code, and will thus appear as another user, which is ignored at the receiver. In other words, as long as the multipath channel induces at least one chip of delay, the multipath signals will arrive at the receiver such that they are shifted in time by at least one chip from the intended signal. The correlation properties of the pseudo-random codes are such that this slight delay causes the multipath to appear uncorrelated with the intended signal, and it is thus ignored.

Some CDMA devices use a rake receiver, which exploits multipath delay components to improve the performance of the system. A rake receiver combines the information from several correlators, each one tuned to a different path delay, producing a stronger version of the signal than a simple receiver with a single correlator tuned to the path delay of the strongest signal.

Frequency reuse is the ability to reuse the same radio channel frequency at other cell sites within a cellular system. In the FDMA and TDMA systems frequency planning is an important consideration. The frequencies used in different cells must be planned carefully to ensure signals from different cells do not interfere with each other. In a CDMA system, the same frequency can be used in every cell, because channelization is done using the pseudo-random codes. Reusing the same frequency in every cell eliminates the need for frequency planning in a CDMA system; however, planning of the different pseudo-random sequences must be done to ensure that the received signal from one cell does not correlate with the signal from a nearby cell.

Since adjacent cells use the same frequencies, CDMA systems have the ability to perform soft handoffs. Soft handoffs allow the mobile telephone to communicate simultaneously with two or more cells. The best signal quality is selected until the handoff is complete. This is different from hard handoffs utilized in other cellular systems. In a hard handoff situation, as the mobile telephone approaches a handoff, signal strength may vary abruptly. In contrast, CDMA systems use the soft handoff, which is undetectable and provides a more reliable and higher quality signal.

Chapter- 9

GSM

The GSM logo is used to identify compatible handsets and equipment

GSM (**Global System for Mobile Communications**: originally from ***Groupe Spécial Mobile***) is the world's most popular standard for mobile telephony systems. The GSM Association estimates that 80% of the global mobile market uses the standard. GSM is used by over 1.5 billion people across more than 212 countries and territories. This ubiquity means that subscribers can use their phones throughout the world, enabled by international roaming arrangements between mobile network operators. GSM differs from its predecessor technologies in that both signaling and speech channels are digital, and thus GSM is considered a *second generation* (2G) mobile phone system. This also facilitates the wide-spread implementation of data communication applications into the system.

The GSM standard has been an advantage to both consumers, who may benefit from the ability to roam and switch carriers without replacing phones, and also to network operators, who can choose equipment from many GSM equipment vendors. GSM also pioneered low-cost implementation of the short message service (SMS), also called text messaging, which has since been supported on other mobile phone standards as well. The standard includes a worldwide emergency telephone number feature (112).

Newer versions of the standard were backward-compatible with the original GSM system. For example, Release '97 of the standard added packet data capabilities by means of General Packet Radio Service (GPRS). Release '99 introduced higher speed data transmission using Enhanced Data Rates for GSM Evolution (EDGE).

History

In 1982, the European Conference of Postal and Telecommunications Administrations (CEPT) created the Groupe Spécial Mobile (GSM) to develop a standard for a mobile telephone system that could be used across Europe. In 1987, a memorandum of understanding was signed by 13 countries to develop a common cellular telephone system across Europe. In 1989, GSM responsibility was transferred to the European Telecommunications Standards Institute (ETSI) and phase I of the GSM specifications were published in 1990. The first GSM network was launched in 1991 by Radiolinja in Finland with joint technical infrastructure maintenance from Ericsson. By the end of 1993, over a million subscribers were using GSM phone networks being operated by 70 carriers across 48 countries.

Technical details

GSM cell site antennas in the Deutsches Museum, Munich, Germany

GSM is a cellular network, which means that mobile phones connect to it by searching for cells in the immediate vicinity. There are five different cell sizes in a GSM network—macro, micro, pico, femto and umbrella cells. The coverage area of each cell varies according to the implementation environment. Macro cells can be regarded as cells where the base station antenna is installed on a mast or a building above average roof top level. Micro cells are cells whose antenna height is under average roof top level; they are typically used in urban areas. Picocells are small cells whose coverage diameter is a few dozen metres; they are mainly used indoors. Femtocells are cells designed for use in residential or small business environments and connect to the service provider's network via a broadband internet connection. Umbrella cells are used to cover shadowed regions of smaller cells and fill in gaps in coverage between those cells.

Cell horizontal radius varies depending on antenna height, antenna gain and propagation conditions from a couple of hundred meters to several tens of kilometres. The longest distance the GSM specification supports in practical use is 35 kilometres (22 mi). There are also several implementations of the concept of an extended cell, where the cell radius could be double or even more, depending on the antenna system, the type of terrain and the timing advance.

Indoor coverage is also supported by GSM and may be achieved by using an indoor picocell base station, or an indoor repeater with distributed indoor antennas fed through power splitters, to deliver the radio signals from an antenna outdoors to the separate indoor distributed antenna system. These are typically deployed when a lot of call capacity is needed indoors; for example, in shopping centers or airports. However, this is not a prerequisite, since indoor coverage is also provided by in-building penetration of the radio signals from any nearby cell.

The modulation used in GSM is Gaussian minimum-shift keying (GMSK), a kind of continuous-phase frequency shift keying. In GMSK, the signal to be modulated onto the carrier is first smoothed with a Gaussian low-pass filter prior to being fed to a frequency modulator, which greatly reduces the interference to neighboring channels (adjacent-channel interference).

GSM carrier frequencies

GSM networks operate in a number of different carrier frequency ranges (separated into GSM frequency ranges for 2G and UMTS frequency bands for 3G), with most 2G GSM networks operating in the 900 MHz or 1800 MHz bands. Where these bands were already allocated, the 850 MHz and 1900 MHz bands were used instead (for example in Canada and the United States). In rare cases the 400 and 450 MHz frequency bands are assigned in some countries because they were previously used for first-generation systems.

Most 3G networks in Europe operate in the 2100 MHz frequency band.

Regardless of the frequency selected by an operator, it is divided into timeslots for individual phones to use. This allows eight full-rate or sixteen half-rate speech channels per radio frequency. These eight radio timeslots (or eight burst periods) are grouped into a TDMA frame. Half rate channels use alternate frames in the same timeslot. The channel data rate for all 8 channels is 270.833 kbit/s, and the frame duration is 4.615 ms.

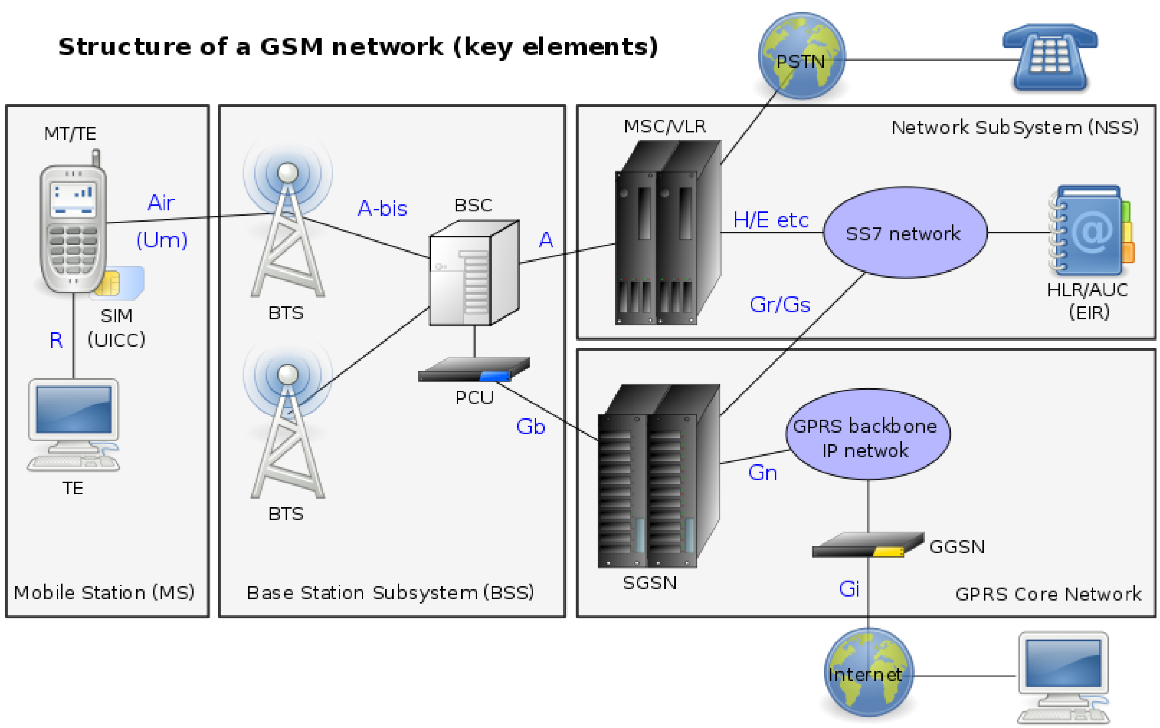
The transmission power in the handset is limited to a maximum of 2 watts in GSM850/900 and 1 watt in GSM1800/1900.

Voice codecs

GSM has used a variety of voice codecs to squeeze 3.1 kHz audio into between 6.5 and 13 kbit/s. Originally, two codecs, named after the types of data channel they were allocated, were used, called Half Rate (6.5 kbit/s) and Full Rate (13 kbit/s). These used a system based upon linear predictive coding (LPC). In addition to being efficient with bitrates, these codecs also made it easier to identify more important parts of the audio, allowing the air interface layer to prioritize and better protect these parts of the signal.

GSM was further enhanced in 1997 with the Enhanced Full Rate (EFR) codec, a 12.2 kbit/s codec that uses a full rate channel. Finally, with the development of UMTS, EFR was refactored into a variable-rate codec called AMR-Narrowband, which is high quality and robust against interference when used on full rate channels, and less robust but still relatively high quality when used in good radio conditions on half-rate channels.

Network structure



The structure of a GSM network

The network is structured into a number of discrete sections:

- The Base Station Subsystem (the base stations and their controllers).
- the Network and Switching Subsystem (the part of the network most similar to a fixed network). This is sometimes also just called the core network.
- The GPRS Core Network (the optional part which allows packet based Internet connections).
- The Operations support system (OSS) for maintenance of the network.

Subscriber Identity Module (SIM)

One of the key features of GSM is the Subscriber Identity Module, commonly known as a **SIM card**. The SIM is a detachable smart card containing the user's subscription information and phone book. This allows the user to retain his or her information after switching handsets. Alternatively, the user can also change operators while retaining the handset simply by changing the SIM. Some operators will block this by allowing the phone to use only a single SIM, or only a SIM issued by them; this practice is known as SIM locking.

Phone locking

Sometimes mobile network operators restrict handsets that they sell for use with their own network. This is called *locking* and is implemented by a software feature of the phone. Because the purchase price of the mobile phone to the consumer is typically subsidized with revenue from subscriptions, operators must recoup this investment before a subscriber terminates service. A subscriber may usually contact the provider to remove the lock for a fee, utilize private services to remove the lock, or make use of free or fee-based software and websites to unlock the handset themselves.

In some territories (e.g., Bangladesh, Hong Kong, India, Malaysia, Pakistan, Singapore) all phones are sold unlocked. In others (e.g., Finland, Singapore) it is unlawful for operators to offer any form of subsidy on a phone's price.

GSM service security

GSM was designed with a moderate level of service security. The system was designed to authenticate the subscriber using a pre-shared key and challenge-response. Communications between the subscriber and the base station can be encrypted. The development of UMTS introduces an optional Universal Subscriber Identity Module (USIM), that uses a longer authentication key to give greater security, as well as mutually authenticating the network and the user - whereas GSM only authenticates the user to the network (and not vice versa). The security model therefore offers confidentiality and authentication, but limited authorization capabilities, and no non-repudiation.

GSM uses several cryptographic algorithms for security. The A5/1 and A5/2 stream ciphers are used for ensuring over-the-air voice privacy. A5/1 was developed first and is a stronger algorithm used within Europe and the United States; A5/2 is weaker and used in other countries. Serious weaknesses have been found in both algorithms: it is possible to break A5/2 in real-time with a ciphertext-only attack, and in February 2008, Pico Computing, Inc revealed its ability and plans to commercialize FPGAs that allow A5/1 to be broken with a rainbow table attack. The system supports multiple algorithms so operators may replace that cipher with a stronger one.

On 28 December 2009 German computer engineer Karsten Nohl announced that he had cracked the A5/1 cipher. According to Nohl, he developed a number of rainbow tables (static values which reduce the time needed to carry out an attack) and have found new sources for known plaintext attacks. He also said that it is possible to build "a full GSM interceptor ... from open source components" but that they had not done so because of legal concerns.

In 2010, threatpost.com reported that "A group of cryptographers has developed a new attack that has broken Kasumi, the encryption algorithm used to secure traffic on 3G GSM wireless networks. The technique enables them to recover a full key by using a tactic known as a Related-key attack, but experts say it is not the end of the world for Kasumi."

Although security issues remain for GSM newer standards and algorithms may address this. New attacks are growing in the wild which take advantage of poor security implementations, architecture and development for smart phone applications. Some wiretapping and eavesdropping techniques hijack the audio input and output providing an opportunity for a 3rd party to listen in to the conversation. Although this threat is mitigated by the fact the attack has to come in the form of a Trojan, malware or a virus and might be detected by security software.

Standards information

The GSM systems and services are described in a set of standards governed by ETSI, where a full list is maintained.

GSM open-source software

Several open source software projects exist that provide certain GSM features:

- gsmd daemon by Openmoko
- OpenBTS develops a Base transceiver station
- OpenBSC is developing a minimalistic, self-contained GSM network
- *The GSM Software Project* aims to build a GSM analyzer for less than $1000
- *OsmocomBB* developers intend to replace the proprietary baseband GSM stack with a free software implementation

Issues with patents and open source

Patents remain a problem for any open source GSM implementation, because it is not possible for GNU or any other free software distributor to guarantee immunity from all lawsuits by the patent holders against the users. Furthermore new features are being added to the standard all the time which means they have patent protection for a number of years.

The original GSM implementations from 1991 are now entirely free of patent encumbrances and it is expected that OpenBTS will be able to implement features of that initial specification without limit and that as patents subsequently expire, those features can be added into the open source version. To date there have been no law suits against users of OpenBTS over GSM use.

Chapter- 10

Error Detection and Correction

In information theory and coding theory with applications in computer science and telecommunication, **error detection and correction** or **error control** are techniques that enable reliable delivery of digital data over unreliable communication channels. Many communication channels are subject to channel noise, and thus errors may be introduced during transmission from the source to a receiver. Error detection techniques allow detecting such errors, while error correction enables reconstruction of the original data.

The general definitions of the terms are as follows:

- *Error detection* is the detection of errors caused by noise or other impairments during transmission from the transmitter to the receiver.
- *Error correction* is the detection of errors and reconstruction of the original, error-free data.

Error correction may generally be realized in two different ways:

- *Automatic repeat request (ARQ)* (sometimes also referred to as *backward error correction*): This is an error control technique whereby an error detection scheme is combined with requests for retransmission of erroneous data. Every block of data received is checked using the error detection code used, and if the check fails, retransmission of the data is requested – this may be done repeatedly, until the data can be verified.
- *Forward error correction (FEC)*: The sender encodes the data using an *error-correcting code (ECC)* prior to transmission. The additional information (redundancy) added by the code is used by the receiver to recover the original data. In general, the reconstructed data is what is deemed the "most likely" original data.

ARQ and FEC may be combined, such that minor errors are corrected without retransmission, and major errors are corrected via a request for retransmission: this is called *hybrid automatic repeat-request (HARQ)*.

Introduction

The general idea for achieving error detection and correction is to add some redundancy (i.e., some extra data) to a message, which receivers can use to check consistency of the delivered message, and to recover data determined to be erroneous. Error-detection and correction schemes can be either systematic or non-systematic: In a systematic scheme, the transmitter sends the original data, and attaches a fixed number of *check bits* (or *parity data*), which are derived from the data bits by some deterministic algorithm. If only error detection is required, a receiver can simply apply the same algorithm to the received data bits and compare its output with the received check bits; if the values do not match, an error has occurred at some point during the transmission. In a system that uses a non-systematic code, the original message is transformed into an encoded message that has at least as many bits as the original message.

Good error control performance requires the scheme to be selected based on the characteristics of the communication channel. Common channel models include memory-less models where errors occur randomly and with a certain probability, and dynamic models where errors occur primarily in bursts. Consequently, error-detecting and correcting codes can be generally distinguished between *random-error-detecting/correcting* and *burst-error-detecting/correcting*. Some codes can also be suitable for a mixture of random errors and burst errors.

If the channel capacity cannot be determined, or is highly varying, an error-detection scheme may be combined with a system for retransmissions of erroneous data. This is known as automatic repeat request (ARQ), and is most notably used in the Internet. An alternate approach for error control is hybrid automatic repeat request (HARQ), which is a combination of ARQ and error-correction coding.

Error detection schemes

Error detection is most commonly realized using a suitable hash function (or checksum algorithm). A hash function adds a fixed-length *tag* to a message, which enables receivers to verify the delivered message by recomputing the tag and comparing it with the one provided.

There exists a vast variety of different hash function designs. However, some are of particularly widespread use because of either their simplicity or their suitability for detecting certain kinds of errors (e.g., the cyclic redundancy check's performance in detecting burst errors).

Random-error-correcting codes based on minimum distance coding can provide a suitable alternative to hash functions when a strict guarantee on the minimum number of errors to be detected is desired. Repetition codes, described below, are special cases of error-correcting codes: although rather inefficient, they find applications for both error correction and detection due to their simplicity.

Repetition codes

A **repetition code** is a coding scheme that repeats the bits across a channel to achieve error-free communication. Given a stream of data to be transmitted, the data is divided into blocks of bits. Each block is transmitted some predetermined number of times. For example, to send the bit pattern "1011", the four-bit block can be repeated three times, thus producing "1011 1011 1011". However, if this twelve-bit pattern was received as "1010 1011 1011" – where the first block is unlike the other two – it can be determined that an error has occurred.

Repetition codes are not very efficient, and can be susceptible to problems if the error occurs in exactly the same place for each group (e.g., "1010 1010 1010" in the previous example would be detected as correct). The advantage of repetition codes is that they are extremely simple, and are in fact used in some transmissions of numbers stations.

Parity bits

A **parity bit** is a bit that is added to a group of source bits to ensure that the number of set bits (i.e., bits with value 1) in the outcome is even or odd. It is a very simple scheme that can be used to detect single or any other odd number (i.e., three, five, etc.) of errors in the output. An even number of flipped bits will make the parity bit appear correct even though the data is erroneous.

Extensions and variations on the parity bit mechanism are horizontal redundancy checks, vertical redundancy checks, and "double," "dual," or "diagonal" parity (used in RAID-DP).

Checksums

A **checksum** of a message is a modular arithmetic sum of message code words of a fixed word length (e.g., byte values). The sum may be negated by means of a one's-complement prior to transmission to detect errors resulting in all-zero messages.

Checksum schemes include parity bits, check digits, and longitudinal redundancy checks. Some checksum schemes, such as the Luhn algorithm and the Verhoeff algorithm, are specifically designed to detect errors commonly introduced by humans in writing down or remembering identification numbers.

Cyclic redundancy checks (CRCs)

A **cyclic redundancy check (CRC)** is a single-burst-error-detecting cyclic code and non-secure hash function designed to detect accidental changes to digital data in computer networks. It is characterized by specification of a so-called *generator polynomial*, which is used as the divisor in a polynomial long division over a finite field, taking the input data as the dividend, and where the remainder becomes the result.

Cyclic codes have favorable properties in that they are well suited for detecting burst errors. CRCs are particularly easy to implement in hardware, and are therefore commonly used in digital networks and storage devices such as hard disk drives.

Even parity is a special case of a cyclic redundancy check, where the single-bit CRC is generated by the divisor $x+1$.

Cryptographic hash functions

A **cryptographic hash function** can provide strong assurances about data integrity, provided that changes of the data are only accidental (i.e., due to transmission errors). Any modification to the data will likely be detected through a mismatching hash value. Furthermore, given some hash value, it is infeasible to find some input data (other than the one given) that will yield the same hash value. Message authentication codes, also called *keyed* cryptographic hash functions, provide additional protection against intentional modification by an attacker.

Error-correcting codes

Any error-correcting code can be used for error detection. A code with *minimum Hamming distance*, d, can detect up to d-1 errors in a code word. Using minimum-distance-based error-correcting codes for error detection can be suitable if a strict limit on the minimum number of errors to be detected is desired.

Codes with minimum Hamming distance d=2 are degenerate cases of error-correcting codes, and can be used to detect single errors. The parity bit is an example of a single-error-detecting code.

The Berger code is an early example of a unidirectional error(-correcting) code that can detect any number of errors on an asymmetric channel, provided that only transitions of cleared bits to set bits *or* set bits to cleared bits can occur.

Error correction

Automatic repeat request

Automatic Repeat reQuest (ARQ) is an error control method for data transmission that makes use of error-detection codes, acknowledgment and/or negative acknowledgment messages, and timeouts to achieve reliable data transmission. An *acknowledgment* is a message sent by the receiver to indicate that it has correctly received a data frame.

Usually, when the transmitter does not receive the acknowledgment before the timeout occurs (i.e., within a reasonable amount of time after sending the data frame), it retransmits the frame until it is either correctly received or the error persists beyond a predetermined number of retransmissions.

Three types of ARQ protocols are Stop-and-wait ARQ, Go-Back-N ARQ, and Selective Repeat ARQ.

ARQ is appropriate if the communication channel has varying or unknown capacity, such as is the case on the Internet. However, ARQ requires the availability of a back channel, results in possibly increased latency due to retransmissions, and requires the maintenance of buffers and timers for retransmissions, which in the case of network congestion can put a strain on the server and overall network capacity.

Error-correcting code

An error-correcting code (ECC) or forward error correction (FEC) code is a system of adding redundant data, or *parity data*, to a message, such that it can be recovered by a receiver even when a number of errors (up to the capability of the code being used) were introduced, either during the process of transmission, or on storage. Since the receiver does not have to ask the sender for retransmission of the data, a back-channel is not required in forward error correction, and it is therefore suitable for simplex communication such as broadcasting. Error-correcting codes are frequently used in lower-layer communication, as well as for reliable storage in media such as CDs, DVDs, hard disks, and RAM.

Error-correcting codes are usually distinguished between convolutional codes and block codes:

- *Convolutional codes* are processed on a bit-by-bit basis. They are particularly suitable for implementation in hardware, and the Viterbi decoder allows optimal decoding.

- *Block codes* are processed on a block-by-block basis. Early examples of block codes are repetition codes, Hamming codes and multidimensional parity-check codes. They were followed by a number of efficient codes, Reed-Solomon codes being the most notable due to their current widespread use. Turbo codes and low-density parity-check codes (LDPC) are relatively new constructions that can provide almost optimal efficiency.

Shannon's theorem is an important theorem in forward error correction, and describes the maximum information rate at which reliable communication is possible over a channel that has a certain error probability or signal-to-noise ratio (SNR). This strict upper limit is expressed in terms of the channel capacity. More specifically, the theorem says that there exist codes such that with increasing encoding length the probability of error on a discrete memoryless channel can be made arbitrarily small, provided that the code rate is smaller than the channel capacity. The code rate is defined as the fraction k/n of k source symbols and n encoded symbols.

The actual maximum code rate allowed depends on the error-correcting code used, and may be lower. This is because Shannon's proof was only of existential nature, and did not show how to construct codes which are both optimal and have efficient encoding and decoding algorithms.

Hybrid schemes

Hybrid ARQ is a combination of ARQ and forward error correction. There are two basic approaches :

- Messages are always transmitted with FEC parity data (and error-detection redundancy). A receiver decodes a message using the parity information, and requests retransmission using ARQ only if the parity data was not sufficient for successful decoding (identified through a failed integrity check).
- Messages are transmitted without parity data (only with error-detection information). If a receiver detects an error, it requests FEC information from the transmitter using ARQ, and uses it to reconstruct the original message.

The latter approach is particularly attractive on an erasure channel when using a rateless erasure code.

Applications

Applications that require low latency (such as telephone conversations) cannot use Automatic Repeat reQuest (ARQ); they must use Forward Error Correction (FEC). By the time an ARQ system discovers an error and re-transmits it, the re-sent data will arrive too late to be any good.

Applications where the transmitter immediately forgets the information as soon as it is sent (such as most television cameras) cannot use ARQ; they must use FEC because when an error occurs, the original data is no longer available. (This is also why FEC is used in data storage systems such as RAID and distributed data store).

Applications that use ARQ must have a return channel. Applications that have no return channel cannot use ARQ.

Applications that require extremely low error rates (such as digital money transfers) must use ARQ.

The Internet

In a typical TCP/IP stack, error control is performed at multiple levels:

- Each Ethernet frame carries a CRC-32 checksum. Frames received with incorrect checksums are discarded by the receiver hardware.
- The IPv4 header contains a checksum protecting the contents of the header. Packets with mismatching checksums are dropped within the network or at the receiver.

- The checksum was omitted from the IPv6 header in order to minimize processing costs in network routing and because current link layer technology is assumed to provide sufficient error detection.
- UDP has an optional checksum covering the payload and addressing information from the UDP and IP headers. Packets with incorrect checksums are discarded by the operating system network stack. The checksum is optional under IPv4, only, because the IP layer checksum may already provide the desired level of error protection.
- TCP provides a checksum for protecting the payload and addressing information from the TCP and IP headers. Packets with incorrect checksums are discarded within the network stack, and eventually get retransmitted using ARQ, either explicitly (such as through triple-ack) or implicitly due to a timeout.

Deep-space telecommunications

Development of error-correction codes was tightly coupled with the history of deep-space missions due to the extreme dilution of signal power over interplanetary distances, and the limited power availability aboard space probes. Whereas early missions sent their data uncoded, starting from 1968 digital error correction was implemented in the form of (sub-optimally decoded) convolutional codes and Reed-Muller codes. The Reed-Muller code was well suited to the noise the spacecraft was subject to (approximately matching a bell curve), and was implemented at the Mariner spacecraft for missions between 1969 and 1977.

The Voyager 1 and Voyager 2 missions, which started in 1977, were designed to deliver color imaging amongst scientific information of Jupiter and Saturn. This resulted in increased coding requirements, and thus the spacecrafts were supported by (optimally Viterbi-decoded) convolutional codes that could be concatenated with an outer Golay (24,12,8) code. The Voyager 2 probe additionally supported an implementation of a Reed-Solomon code: the concatenated Reed-Solomon-Viterbi (RSV) code allowed for very powerful error correction, and enabled the spacecraft's extended journey to Uranus and Neptune.

The CCSDS currently recommends usage of error correction codes with performance similar to the Voyager 2 RSV code as a minimum. Concatenated codes are increasingly falling out of favor with space missions, and are replaced by more powerful codes such as Turbo codes or LDPC codes.

The different kinds of deep space and orbital missions that are conducted suggest that trying to find a "one size fits all" error correction system will be an ongoing problem for some time to come. For missions close to earth the nature of the channel noise is different from that of a spacecraft on an interplanetary mission experiences. Additionally, as a spacecraft increases its distance from earth, the problem of correcting for noise gets larger.

Satellite broadcasting (DVB)

The demand for satellite transponder bandwidth continues to grow, fueled by the desire to deliver television (including new channels and High Definition TV) and IP data. Transponder availability and bandwidth constraints have limited this growth, because transponder capacity is determined by the selected modulation scheme and Forward error correction (FEC) rate.

Overview

- QPSK coupled with traditional Reed Solomon and Viterbi codes have been used for nearly 20 years for the delivery of digital satellite TV.
- Higher order modulation schemes such as 8PSK, 16QAM and 32QAM have enabled the satellite industry to increase transponder efficiency by several orders of magnitude.
- This increase in the information rate in a transponder comes at the expense of an increase in the carrier power to meet the threshold requirement for existing antennas.
- Tests conducted using the latest chipsets demonstrate that the performance achieved by using Turbo Codes may be even lower than the 0.8 dB figure assumed in early designs.

Data storage

Error detection and correction codes are often used to improve the reliability of data storage media.

A "parity track" was present on the first magnetic tape data storage in 1951. The "Optimal Rectangular Code" used in group code recording tapes not only detects but also corrects single-bit errors.

Some file formats, particularly archive formats, include a checksum (most often CRC32) to detect corruption and truncation and can employ redundancy and/or parity files to recover portions of corrupted data.

Reed Solomon codes are used in compact discs to correct errors caused by scratches.

Modern hard drives use CRC codes to detect and Reed-Solomon codes to correct minor errors in sector reads, and to recover data from sectors that have "gone bad" and store that data in the spare sectors.

RAID systems use a variety of error correction techniques, to correct errors when a hard drive completely fails.

Error-correcting memory

DRAM memory may provide increased protection against soft errors by relying on error correcting codes. Such error-correcting memory, known as *ECC* or *EDAC-protected* memory, is particularly desirable for high fault-tolerant applications, such as servers, as well as deep-space applications due to increased radiation.

Error-correcting memory controllers traditionally use Hamming codes, although some use triple modular redundancy.

Interleaving allows distributing the effect of a single cosmic ray potentially upsetting multiple physically neighboring bits across multiple words by associating neighboring bits to different words. As long as a single event upset (SEU) does not exceed the error threshold (e.g., a single error) in any particular word between accesses, it can be corrected (e.g., by a single-bit error correcting code), and the illusion of an error-free memory system may be maintained.